IFERP
connecting engineers... developing research

ISSN (Online) 2394-2320

International Journal of Engineering Research in Computer Science and Engineering (IJERCSE)
Vol 5, Issue 5, May 2018

Alternative Optimization Algorithm for Channel and Clipping Amplitude Estimation in IoT-based Networks

[1] Thasni mol.K, [2] Ismayil siyad.C
[1][2] MEA Engineering College, Vengoor (P.O), Perinthalmanna, Malappuram (Dist), Kerala, India

***Abstract*: -- Internet of Things (IoT) is the idea to connect all devices to the internet. In IoT based devices the transmitter side should be low cost and low power. So the complexity of transmitter side should be transferred to the receiver side. Orthogonal Frequency Division Multiplexing (OFDM) is widely recognized as one of the key techniques for high data rate communications in wireless networks. The advantages of OFDM include high spectral efficiency, immunity against multipath fading and negligible inter-symbol interference. But it has a drawback of high PAPR. Among the PAPR reduction techniques clipping is selected because it is the simplest and cost-effective method. In this work, the joint estimation of clipping Amplitude and MMSE based channel estimation method has been proposed and evaluated. Here the channel is estimated by using MMSE technique and an iterative amplitude reconstruction method is used to estimate the clipping amplitude. An alternative optimization algorithm is used for the joint estimation of channel and clipping amplitude which is based on frequency domain block type training symbols. Here the channel is estimated by using LS and MMSE and made a comparison between them. The simulation results show that the MMSE has better performance than LS channel estimation in terms of SNR and MSE. The efficiency of these algorithms is evaluated by using CRLB calculation and achieves that lower bound at medium SNR. The simulation result shows a good performance without any prior information.**

***Index Terms* – Channel estimation, Clipping amplitude, CRLB, IAR, MMSE, LS, OFDM, PAPR.**

I. INTRODUCTION

Orthogonal Frequency Division Multiplexing (OFDM) is a popular technology in the field of high-speed wireless communication. It has many advantages such as efficient bandwidth utilization, strong ability to cancel inter -Symbol Interference (ISI) and frequency selective fading. The principle of OFDM is to divide the digital signal and is transmitted through a large number of sub-carriers. Each symbol has different amplitude. In order to amplify these signals, HPA is needed. Along with the advantages, OFDM has a disadvantage of high Peak-to-Average Power Ratio (PAPR). Due to this high PAPR, the HPA is forced to work in the saturation region and hence the transmitter output suffers some nonlinear distortions. In future wireless networks, for constructing efficient HPAs in IoT-based 5G, are more difficult and costly at millimeter wave spectrum [1]. In order to reduce the high PAPR of OFDM signals a number of techniques have been discussed in [2]. PAPR is the ratio of instantaneous power to the average power. In IoT-based devices the transmitter side should be low cost and low power, so we used clipping method in order to reduce PAPR, which is the simplest one. After passing through the channel the transmitted signals are received by the receiver. The transmitted signal travels through a channel, which may offer noises, by undergoing many damaging effects and then the corrupted signal reaches the receiver. This may adversely affect the performance of the system. The changes in the signals may due to clipping, channel effect etc. So in order to mitigate the effect of signals and to retrieve the transmitted signal, we have to estimate the channel and clipping level. The distortion introduced by the HPAs nonlinearity includes the works such as a pre-distorter is used at the transmitter side, a signal design scheme is used to reduce PAPR and the distortion extenuating at the receiver side. In order to operate the HPA linearly, the cascaded combination of a digital pre-distorter and HPA is used commonly. The pre-distorter is used to pre-distort the signal, and the cascaded combination of this distorted signal with the nonlinear HPA makes a linear system [3]. The complexity and the cost of pre-distorter are high. So, it is not preferable in IoT-based systems. In this paper, we present a new algorithm for channel estimation and clipping amplitude estimation based on MMSE and Iterative amplitude reconstruction by using frequency-domain block-type training symbols. One way to reduce PAPR is the intentional clipping of OFDM signal. In IoT –based devices, to reduce the PAPR a limiter (clipper) is used instead of using costly HPA. A limiter is used for clipping and the clipper will clip-off the signals above the clipping level. The Cascaded combination of a pre-distorter and HPA is approximated by a limiter. The efficiency of the high power amplifier and battery life can be increased by clipping the high PAPR signal. In each transmission, the CA will change, so the receiver needs to estimate it, to get an updated value since the CA and channel are strange to the transmitter and receiver [4]. In most of the previous works, the multipath channel has been ignored or taken as fully known at the receiver side. The channel is estimated by using MMSE channel estimation and CA is estimated using iterative amplitude reconstruction method. Once the channel and CA are estimated, the symbols are recovered by using iterative detection method [5]. The paper helps on recovering the transmitted signals and joint estimation in the receiver side avoids executing complex algorithms in the transmitter side. The rest of this paper is organized as follows. In Section II, the system model of clipped OFDM and the Iterative amplitude reconstruction method are discussed. The two channel estimation techniques LS and MMSE are discussed in Section III. Section IV discusses the initialization of the alternating optimization algorithm and also, explains about Cramer-Rao lower bound. Simulation results are analyzed in section V and the paper is concluded in Section VI.

II. SYSTEM MODEL OF CLIPPED OFDM

Consider an OFDM system with N subcarriers which uses QAM modulation. The transmitted symbols S = [s0,s1.....sN-1]T is the frequency domain symbol vector for the QAM modulation [4]. By taking the IDFT of frequency domain symbols, we get the time domain symbols as;

$$xn=\frac{1}{\sqrt{N}}\sum_{k=0}^{N-1} s_k e^{-j\frac{2}{N}}, \quad n=0,1,......,N-1 \tag{1}$$

This can be written in matrix form as;

$$X=FHS \tag{2}$$

Where, F is the N x N DFT matrix. There is a slow-fading channel with L + 1 taps and is denoted as, h= [h0, h1.....hL] T. The output of the clipper can be represented as;

$$F(r) = \begin{cases} r, & : r \leq A \\ A, & : r > A \end{cases}$$

$$\varphi(r) = 0 \tag{3}$$

Where, r is the input of the limiter, and A is the CA. When we consider the phase term, then the output of the limiter is given by;

$$g(u; A) = \begin{cases} u, & : |u| \leq A \\ Ae^{j\arg(u)}, & : |u| > A \end{cases} \tag{4}$$

The output of limiter is taken element-wise. But it is difficult to directly work with the output of the limiter. To avoid this difficulty, we use a technique that, the output of the limiter can be denoted as 1 or 0. If Cn =1 then the signal is clipped, else the signal is not clipped.

$$C_n = \begin{cases} 0, & : r_n \leq A \\ 1, & : r_n > A \end{cases} \tag{5}$$

Then, the output of the limiter can be expressed as;

zn= (1-cn) xn + Acnejϕ , n=0,.....,N-1 (6)

We can express (6) in vector form as;

z= (1-c) x + Ac ejϕ (7)

Where 1 denotes an all one vector, ejϕ denotes the phase factor of x, and denotes the Hadamard product. Also, we can express x as, x= r ejϕ.

A Cyclic Prefix (CP) is pre-added to the time-domain symbols the transmitter end, is removed at the receiver side. It helps to remove the Inter-Symbol Interference (ISI) and is removed at the receiver. The matrix form can be expressed as;

u=Hz+w (8)

Where, H is the N X N circulant matrix. By taking the DFT of (7), we have the frequency domain OFDM transmission as;

$$y = \sqrt{N} D_H F_z + w \tag{9}$$

Where DH is the diagonal matrix and w is the noise added to the signal.

Iterative Amplitude Reconstruction

The clipping affects only the amplitude of OFDM signals, but it does not depend on phase. Thus the IAR replaces only the amplitude of detected samples. This is why we call this algorithm as iterative amplitude reconstruction. It is clear from the Fig. 1 that a single pair of IFFT/FFT operation is used in each iteration of the amplitude reconstruction technique.

IFERP
connecting engineers... developing research

ISSN (Online) 2394-2320

International Journal of Engineering Research in Computer Science and Engineering (IJERCSE)
Vol 5, Issue 5, May 2018

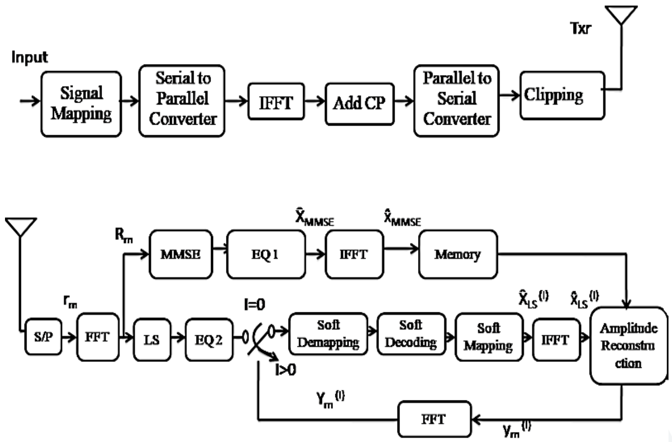

Fig. (a)Transmitter side with clipping and (b) Receiver structure with IAR.

The IAR technique uses two different equalizers to cancel the effect of the channel. Before that, the channel is estimated by using LS and MMSE estimation techniques, which will introduce an increase in the complexity due to the additional equalization technique. Clipping amplitude is calculated by using an algorithm. Clipping amplitude is estimated by sorting the elements in r. The clipped signal can be modeled as the aggregate of an attenuated signal component and clipping noise. The out-of-band components as a result of clipping can be removed by taking DFT of time domain samples [6]. From Fig. 1, we can explain the steps involved in IAR. The clipping amplitude is assumed to know at the receiver.

1. Frequency domain channel observation, Rm[n], is acquired by taking the FFT on the discrete received samples, rm[k], where k= 0 to N-1.
2. Clipped samples are equalized by using an MMSE equalizer (EQ 1) and then, the estimate of the clipped sample, x̂MMSE[k], is obtained and stored in memory by executing IFFT on $\hat{X}$MMSE[n], where n= 0 to N-1.
3. In the second branch, the clipped samples are equalized by using LS equalizer (EQ 2) and the transmitted symbols $\hat{X}$ILS[n] are obtained, where I represent the number of iterations and starts with an initial value of I=0.
4. IFFT is carried out on the decisions in Step 3 to get the LS estimates of the samples and yielding x̂LS [I].
5. The noise is removed in each iteration and we can estimate the signal by comparing it with the clipping amplitude. Then, the amplitude of the clipped signals is reconstructed, and new sequence ymI[k], where k= 0 to N-1 is generated.
6. The sequence ymI[k], where k= 0 to N-1 is converted to the frequency domain, yielding YmI[n], and the transmitted signals $\hat{X}$mI+1[n] are estimated.
7. This completes the Ith iteration, and for more iterations, go back to Step 4 with I=I+1.

III. CHANNEL ESTIMATION TECHNIQUES

The frequency domain channel estimation technique includes channel estimation either by using LS or MMSE.

A. Least Square Channel Estimation

In data-aided channel estimation method makes use of training symbols, the known information to the receiver is inserted in information symbols so that the current channel state can be estimated. By examining the relationship between the known TS and the received symbols, the instantaneous channel impulse response can be estimated. LS algorithm is less complex and easy to implement as it does not require any probability function to determine the channel response [7]. The LS estimate of the channel is given as;

HLS=X-1Y (10)

Where, H is the channel response, X is the input signal and Y is the output signal.

B. MMSE Channel Estimation

Unlike the LS approach, the optimal criterion of the MMSE method is to minimize the MSEs to find an optimal estimator to the unknown parameters. MMSE performs better as compared to LS, as it uses the channel characteristics and signals Noise Ratio (SNR) information to estimate the channel. Let ‘h’ be the channel vector and then the MMSE of the channel is denoted as:

HMMSE=FRhyRyy-1Y (12)

Here Rhy is the cross-correlation matrix between h and y, Ryy is the auto-correlation matrix of y with itself [8].

The MMSE approach can lighten the effect of channel noise in some degree when compared with the LS method. So the MMSE method gives better performance than LS in terms of SNR and MSE. Here the channel is estimated by using MMSE.

IV. INITIALIZATION OF THE ALTERNATIVE OPTIMIZATION ALGORITHM

IFERP
connecting engineers... developing research
ISSN (Online) 2394-2320
International Journal of Engineering Research in Computer Science and Engineering (IJERCSE)
Vol 5, Issue 5, May 2018

The alternative optimization algorithm is used for the joint estimation of channel and clipping amplitude. The channel is estimated by using MMSE technique and CA is estimated by iterative amplitude reconstruction method. This technique uses frequency-domain block-type training symbols. Once the clipping amplitude and channel are estimated, we can use this to discover the transmitted symbols in the remaining block. This is done by using iterative detection method, which is discussed in [5]. The instinct behind this algorithm is that the time variation in CA is a slower than the channel time variance. So, it will not change abruptly for each block, that means it almost same during the transmission of several OFDM blocks.

Based on the initialization used there are two algorithms [4].

1. Initializing by the Channel (Algorithm 1):Here the channel is estimated by using MMSE for the unclipped version of the transmitted signal since the clipping amplitude is unknown at the receiver.
2. Initializing by the Clipping Amplitude (Algorithm 2): The channel is estimated by using MMSE and then the CA is estimated by using iterative amplitude reconstruction.

The CA is estimated by using IAR and channel is estimated by using MMSE.

For LS,

$\hat{h}$ (i) = [VHV]T VHy (12)

For MMSE,

$\hat{h}$(i)= [VHV-σn2In]TVHy (13)

Both algorithms of alternating optimization are sure to converge to a local optimum in every Iteration. So we can find the unique optimal solution of the optimization problem [4]. Once the channel and clipping amplitude estimated, the transmitted signal can be recovered by using iterative detection method [5].

A. Cramer-Rao Lower Bound

The Cramer-Rao Bound is a performance criterion that gives a lower bound to the mean square error of estimation in the set of unbiased estimates [9]. It is used here, as a performance measure. Here, the channel taps are complex valued and CAs are real valued.

CRLB for CA is calculated by;

$$\mathrm{CRLB}\ (A) = \frac{1}{(q-2P_H C_1^{-1} P)} \qquad (14)$$

CRLB for channel is calculated by;

$$\mathrm{CRLB}\ (h) = C_1^{-1} + \frac{1}{(q-2P_H C_1^{-1} P)} C_1^{-1} P P^H C_1^{-H} \qquad (15)$$

V. SIMULATION RESULTS

By computer simulation, we can find the performance of the proposed algorithms. Here we considered sub-carriers (N) such as 128, 256 and 512 and are modulated by using 16 QAM. We consider the signal passes through a Rayleigh fading channel, which has L+1 channel taps. In this model, the channel is fixed in several OFDM blocks, because the channel and clipping amplitude is a slow-time varying phenomenon. The clipping level can be calculated by using the equation:

CL = 20 log10 (A) [dB].

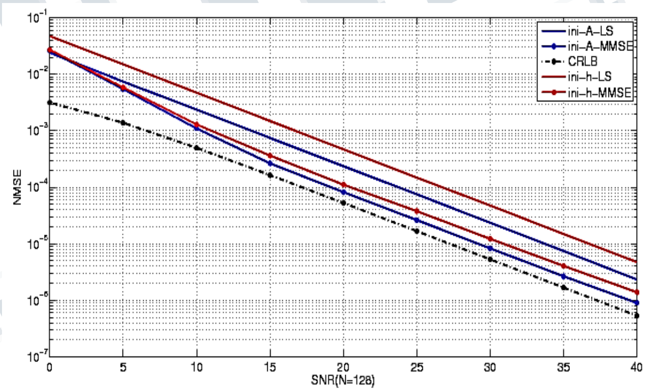

Fig. 2. NMSE performance of LS and MMSE channel estimation by using algorithms 2 and 3, when L+1=7, N=128 and CL=.3.

The Normalized Mean Square Error (NMSE) performance of estimating the channel using the algorithms, algorithm 2 and 3 is shown in the Fig. 2 and Fig. 3 respectively. The channel taps and clipping level of these figures are given by, L+1 = 7 and CL = .3 dB, but Fig. 2 has 128 sub-carriers and Fig. 3 has 256 sub-carriers.

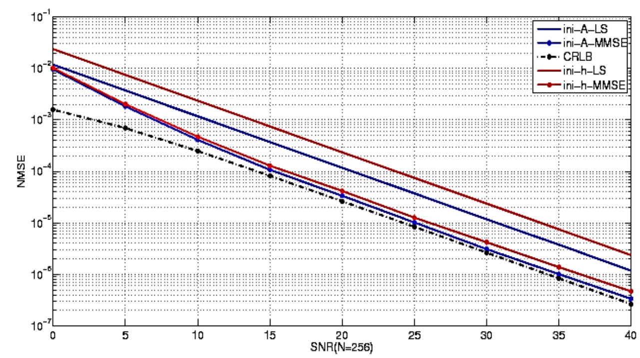

Fig. 3 NMSE performance of LS and MMSE channel estimation by using algorithms 2 and 3, when L+1=7, N=256 and CL=.3.

IFERP
connecting engineers... developing research

ISSN (Online) 2394-2320

International Journal of Engineering Research in Computer Science and Engineering (IJERCSE)
Vol 5, Issue 5, May 2018

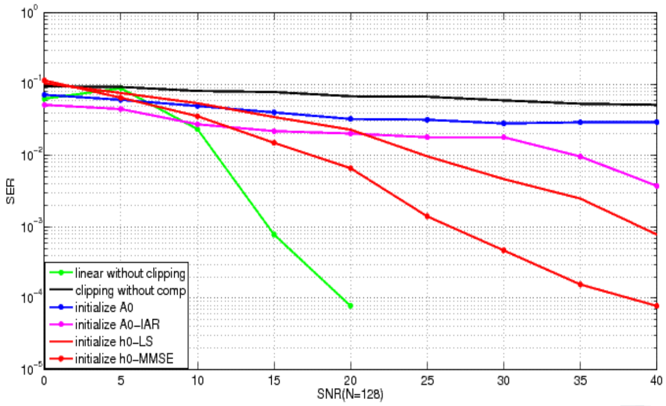

Fig. 4. SER versus SNR performance of IAR method, when L+1=7, N=128 and CL=.3.

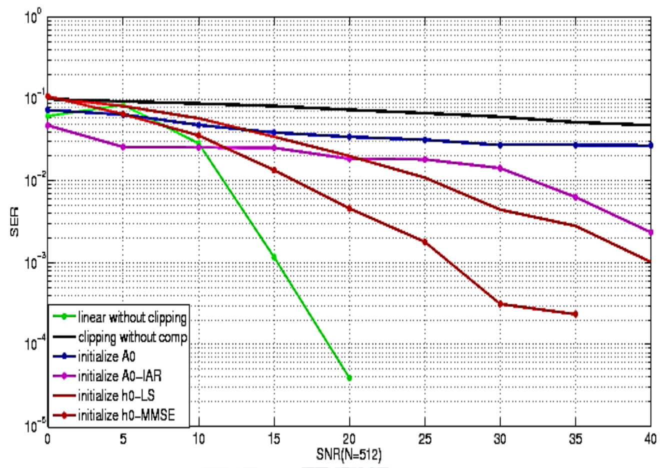

Fig. 5. SER versus SNR performance of IAR method, when L+1=7, N=512 and CL=.3.

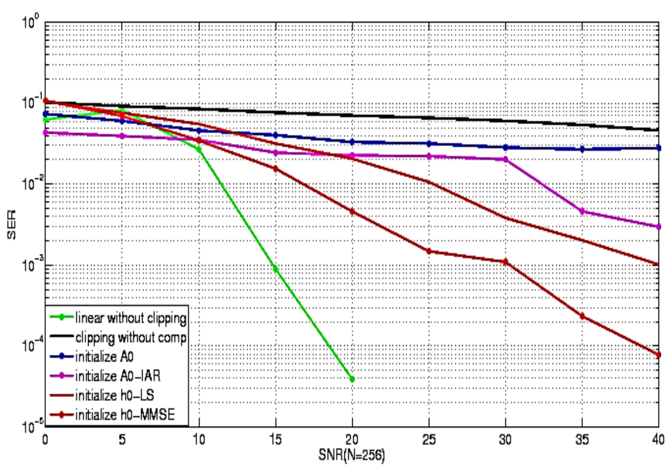

Fig. 6. SER versus SNR performance of IAR method, when L+1=7, N=256 and CL=.3.

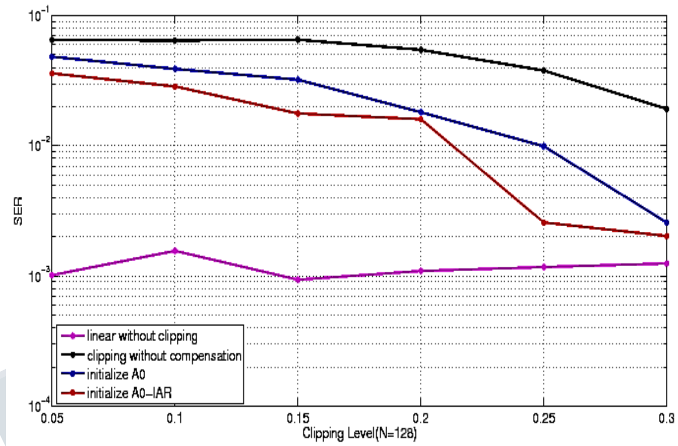

Fig. 7. SER versus CL performance of IAR method, when L+1=7, N=128 and SNR=10.

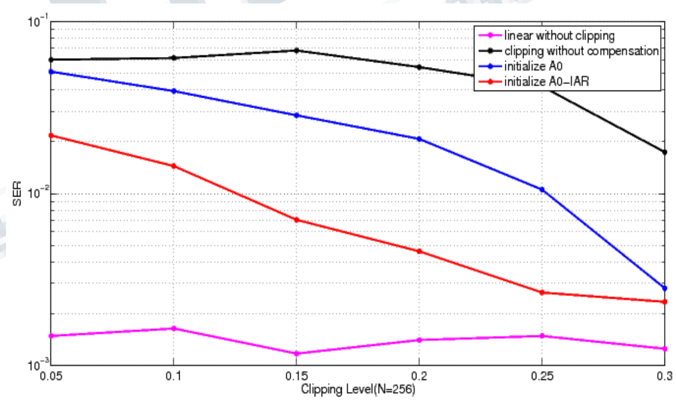

Fig. 8. SER versus CL performance of IAR method, when L+1=7, N=256 and SNR=10.

The Normalized Mean Square Error (NMSE) performance of estimating the channel using the algorithms, algorithm 2 and 3 is shown in the Fig. 2 and Fig. 3 respectively. The channel taps and clipping level of these figures are given by, L+1 = 7 and CL = .3 dB, but Fig. 2 has 128 sub-carriers and Fig. 3 has 256 sub-carriers. A comparison between LS and MMSE channel estimation for both algorithms are shown in the figures. As we can see from the figures the NMSEs are decreased by increasing the SNR and both of them reach CRLB in a medium SNR system. Also, it is clear from the figures that MMSE has better performance as compared to LS in terms of SNR and NMSE. By analyzing all these figures, we can see that the NMSE reaches its CRLB faster for a higher number of sub-carriers. The results are more accurate when the number of sub-carriers and the number of clipped observations are high, which results in more accurate estimates for a given channel length.

The SER performance of two algorithms, when L + 1 = 7, and CL = .3 dB for N = 128 and N = 512, are shown in the Fig. 4 and Fig. 5 respectively. As illustrated in the figures the SER performance of different schemes such as clipping without compensation, linear without clipping, CRLB and also a comparison between LS and MMSE. Here CA is estimated by using iterative Amplitude Reconstruction (IAR). When we estimate the CA and the channel taps using IAR gives the results, which is almost same when compared with the results, in which channel and CA are known at the receiver side. A plot between SNR and SER with the same parameters except for the number of sub-carriers N=256 are shown in the Fig. 6. Again, we can see from the figures that, the SER of different schemes when we estimate the CA and the channel taps using Algorithms 2 and 3 using IAR, almost perfectly match with the case that these are perfectly known at the receiver.

The SER performance of two Algorithms versus CL, when L + 1 = 7, SNR = 10 dB for N = 128 and N = 256, are shown in the Fig. 7 and Fig. 8 respectively. Here, also considered LS and MMSE estimation and found that MMSE has better performance than LS in terms of SER. It is clear from the figure that, the lower bound and upper bound of the iterative algorithms implies linear without compensation and clipping without compensation. All the curves will converge as CA increases two algorithms 2 and 3 give almost same results, in which their difference is negligible. Also, better results will get with the MMSE channel estimation and IAR CA estimation. The main point of these plots is to show that the SER performance is virtually unchanged by the estimation errors.

VI. CONCLUSION

In future IoT-based OFDM networks, we have studied estimation of the channel by using MMSE and clipping level estimation by using IAR at the receiver side are studied in this paper. Today in IoT-based devices, a large number of low-cost low-power nodes transmitting to a highly complex node like BTS. The clipped signals are iteratively reconstructed at the receiver side by using IAR. Based on the type of initialization used that is algorithm 2 and 3 here we have proposed two alternating optimization algorithms. The IAR method reduces the noise and gives an accurate CA, which helps to regain the originally transmitted signal. The estimated theoretical lower bound (CRLB) compares with the performance of these estimates and found that they will achieve this lower bounds. Also, we compare the LS and MMSE channel estimation methods and showed that MMSE has better performance than LS. As a final point of view, we have shown by simulations that, by using the alternative optimization algorithms, the performance of the iterative detection method is almost similar as the receiver already knows the channel and clipping amplitude.

VII. ACKNOWLEDGMENT

I am obediently thankful to the God Almighty. Praise and glory to Him, for all His uncountable bounties and guidance, without which, this work would have never been a reality. I would like to thank my project supervisor, Mr. Ismayil Siyad.C, Assistant Professor, Department of Electronics and Communication for his valuable guidance and timely support throughout the project work. He was always a constant source of encouragement. I also thank my parents and friends for the unceasing encouragement, support and attention. I am also grateful to my life partner who supported me throughout this venture.

REFERENCES

1. Jeffrey G Andrews, Stefano Buzzi, Wan Choi, Stephen V Hanly, Angel Lozano, Anthony CK Soong, and Jianzhong Charlie Zhang, “What will 5g be?” IEEE Journal on selected areas in communications, 32(6):1065–1082, 2014.

2. Yasir Rahmatallah and Seshadri Mohan, “Peak-to-average power ratio reduction in ofdm systems: A survey and taxonomy”. IEEE communications surveys & tutorials, 15(4):1567–1592, 2013.

3. Lei Ding, Guo Tong Zhou, Dennis R Morgan, Zhengxiang Ma, J Stevenson Kenney, Jaehyeong Kim, and Charles R Giardina, “A robust digital baseband predistorter constructed using memory polynomials”. IEEE Transactions on communications, 52(1):159–165, 2004.

4. Ehsan Olfat and Mats Bengtsson, “Joint channel and clipping level estimation for ofdm in iot-based networks”. IEEE Transactions on Signal Processing, 65(18):4902–4911, 2017.

5. Jose Tellado, Louise MC Hoo, and John M Cioffi, “Maximum-likelihood detection of nonlinearly distorted multicarrier symbols by iterative decoding”. IEEE transactions on communications, 51(2):218–228, 2003.

6. Hangjun Chen and Alexander M Haimovich, “Iterative estimation and cancellation of clipping noise for ofdm signals”. IEEE Communications Letters, 7(7):305–307, 2003.

7. E Ahmed, W Aziz, G Abbas, S Saleem, and Q Islam, “Channel estimation for ofdm system using training sequence algorithms”. Adv. Electr. Eng. Syst, 1(3):140–145, 2012.

8. Sweta M Patil and AN Jadhav, “Channel estimation using ls and mmse estimators”. International Journal on Recent and Innovation Trends in Computing and Communication, 2(3):566–570, 2014.

9. Lamia Berriche, Karim Abed-Meraim, and Jean-Claude Belfiore, “Cramer rao bounds for mimo channel estimation”. In Acoustics, Speech, and Signal Processing, 2004. Proceedings.(ICASSP’04). IEEE International Conference on, volume 4, pages iv–iv. IEEE, 2004.