Channel Effect Compensation in OFDM System under Short CP Length Using Adaptive Filter in Wavelet Transform Domain

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***Abstract*- Channel estimation in communication systems is one of the most important issues that can reduce the error rate of sending and receiving information as much as possible. In this regard, estimation of OFDM-based wireless channels using known sub-carriers as pilot is of particular importance in frequency domain. In this paper, channel estimation under short cyclic prefix (CP) in OFDM system is considered. An adaptive algorithm based on the set-membership filtering algorithm is used to compensate for this problem. In short CP length, the per-tone equalization (PTEQ) structure is used to prevent inter-symbol interference (ISI). This structure has high computational complexity, so using the set-membership filtering idea with variable step size while reducing the average computation of the system can also increase the convergence speed of the estimates. On the other hand, utilizing the wavelet transform on the branch of this structure in each sub-carrier before applying adaptive filters will in turn increase the estimation speed. The simulation results show better performance than conventional adaptive algorithms. In addition, the estimation and compensation of the channel effect under short CP can be easily accomplished by this algorithm.**

***Index Terms*-** Channel estimation, PTEQ structure, Set-membership filtering, Short CP, Wavelet packet transform.

I. INTRODUCTION

As a multicarrier modulation technique, orthogonal frequency division multiplexing (OFDM) technique has the ability to provide reliable high data rate transmission in different communication scenarios [1]-[2]. OFDM has been widely used in wireless communication systems due to the resistance to timing error and ferequency selective channels. In OFDM systems, the frequency selective channel is converted to a set of flat fading channels, which can be easily compensated by an one-tap equalizer. Hence, channel estimation of the OFDM system is simplified while increasing data transmission rates [3]-[4]. Channel estimation can increase the capacity of the OFDM system by

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improving bit error rates (BER) [5]. In [6], the channel estimation for the sparse channels of the OFDM system is performed using the least squares (LS) and minimum mean square error (MMSE) methods. According to the requirements of the training symbols, there will be three types of estimates, including blind, semi-blind and pilot-based estimates. Channel estimation methods for pilot symbols include LS, linear minimum mean square error (LMMSE), adaptive filters, etc. [7]-[8].

In [9], channel estimation is performed based on block and comb type pilots using LS, MMSE and LMMSE methods. Also, the variability of the pilot distances and appropriate intervals in the time and frequency dimension is discussed. Then, the channel estimation in the data carriers is obtained using interpoltaion. In [10], the performance of the LS estimation method for two fast and slow fading channel modes is analyzed with two types of block and comb pilot arrangement.

In [11], an efficient IQ and channel compensation scheme is proposed based on pilots; this is achieved through a nonlinear least squares (NLS) analysis of the joint channel and IQ imbalance estimation, and a simple symbol detection procedure. For rigor, the Cramer-Rao lower bounds (CRLBs) for both the IQ imbalance parameters and the channel coefficients are also derived.

In [12], a novel joint channel impulse response estimation and impulsive noise mitigation algorithm based on compressed sensing theory is proposed. In this algorithm, both the channel impulse response and the impulsive noise are treated as a joint sparse vector. Then, the sparse Bayesian learning framework is adopted to jointly estimate the channel impulse response, the impulsive noise, and the data symbols, in which the data symbols are regarded as unknown parameters.

Previous articles have been proposed under sufficient CP length. OFDM symbols transmition with short CP leads to ISI and inter-carrier interference (ICI), affecting channel estimation and data detection. If this problem is not resolved, there will be a large error in data detection process [13]. Employing cyclic-prefix (CP) in multi-carrier systems not only protects the signal from inter-symbol-interference, but also allows circular interpretations of the channel which simplifies the estimation and equalization techniques. Nevertheless, the CP information is usually discarded at the receiver side [14].

In [15], for the elimination of ISI caused by short CP, the per tone equalization (PTEQ) structure in the frequency domain is proposed. Implementation of this structure is able to compensate the ISI effect. In [16], a integrated structure for joint estimation of channel and IQ imbalance is presented. It also uses the PTEQ structure to overcome ISI, assuming short CP length. The reason for using short CP is to increase bandwidth efficiency. In this paper, short CP length is considered because of optimal use of the channel, and then, PTEQ-based algorithms will be used. In addition to effectively estimating the channel, it prevents inter-symbol interference and avoids deterioration of the estimation results.

In this paper, in order to increase the estimation speed of adaptive algorithms, in each sub-carrier of PTEQ output data, wavelet transform is applied. Then, the NLMS adaptive algorithm is developed in the new output signal. Given the length of filters required and the number of sub-carriers, applying any adaptive algorithm to this structure will require a lot of computation to converge. Therefore, it is necessary to optimize the computation of the algorithms by maintaining the required convergence speed and steady state error. In the second step, to reduce the computation of the channel estimation process based on the adaptive algorithm, we need to increase the convergence speed again. For this purpose, set-membership filtering (SMF) in adaptive filters is used.

The use of SMF filtering can improve the estimation speed alone by utilizing variable, optimal and noise variance dependent step size. Also, in some iteration, it prevents partial of sub-carriers from being updated and reduces average computation. Therefore, applying a adaptive algorithm to the PTEQ structure in combination with wavelet packet transform and set-membership filtering will have several important features. First, it will increase channel utilization efficiency by reducing the length of CP used. Second, the resulting ISI due to the short CP utilization is eliminated and, ultimately, due to the use of SMF and wavelet packet (WP) transform, has high speed and accuracy in channel estimation. Therefore, by using the proposed SMF-WP-NLMS-PTEQ method, good channel equalization can be performed for the data.

This paper first describes the OFDM-based channel estimation model in section II. Then, wavelet packet transform and set-membership filtering are reviewed in section II. The proposed adaptive compensation algorithm with assumption of short CP length, is presented using SMF-based adaptive filters in wavelet domain and PTEQ structure, in section III. Finally, the simulation results are presented in section IV and conclusions in section V.

II. BACKGROUND TOPICS

In this section, the compensation of channel distortion under sufficient CP length will be studied and corresponding analytical equations will be introduced to model the channel effect so that based on these equations the proposed method can be presented in the next sections. In sufficient CP length, inter-symbol interference does not occur, so a first-order filter as a compensator can minimize existing distortions. But in the case of short CP length, a structure called PTEQ is used to compensate and estimate the data.

A. CP of sufficient length

First, for a sufficient CP, the channel compensation scheme is examined. It is assumed that S represents the OFDM symbol in the frequency domain with $(N \times 1)$ length, where N is the number of subcarriers. Thus the base band symbol in the time domain can be written as equation (1).

$$s = P_{CI} F_N^{-1} S \tag{1}$$

Where P_{CI} is a cyclic prefix (CP) insertion matrix of length υ to symbol S, and F_N^{-1} represents the inverse matrix of DFT.

When the output symbol in the transmitter passes through the semi-stationary channel with length of L_{ch}, the received baseband symbol r can be written as equation (2).

$$r = c \otimes s + n \tag{2}$$

In the equation (2), c represents the base band channel model. Also, n is Gaussian additive white noise.

If we denote the symbol obtained by eliminating CP in the receiver with Z, then it can be written as equation (3) by the Fourier transform of equation (2) and by applying the CP remotion operator.

$$Z = F_N P_{CR} \{\mathrm{r}\} = CS + N \tag{3}$$

In the above relation, P_{CR} is a matrix that performs the CP remotion and Z and N are Fourier transforms of z and n.

The receiver can now use a filter as a compensator to obtain the transmitted symbol. Equation (4) shows the model of this compensator, in which the $Z[l]$ parameter is used as the input of the compensator filter.

$$\tilde{S}[l] = W[l] Z[l] \tag{4}$$

The $W[l]$ coefficient can be calculated using the minimum square error (MSE) criterion, which is expressed by equation (5).

$$\min_{W[l]} \left\{ \left| \tilde{S}[l] - W[l] Z[l] \right|^2 \right\} \tag{5}$$

B. Wavelet packet transform

Wavelets are transform methods that has received great deal of attention over the past decades. This transform is widely used in various fields of engineering such as data compression, speech and ECG processing, image and video analysis and signal transient state extraction. Wavelet transform can be expressed based on a number of basic functions called mother functions [17]. By using two acts of expansion or contraction and time shift, one can produce a family of wavelets based on the mother wavelet.

The continuous wavelet transform of a signal such as $x(t)$ can be written based on equation (6). In this equation the function $\psi(t)$ is the mother wavelet and plays the function of $e^{j\omega t}$ in the Fourier transform.

$$Tx(a,b) = \frac{1}{\sqrt{a}} \int_{-\infty}^{+\infty} x(t) \psi^*\left(\frac{t-b}{a}\right) dt \qquad a > 0,\ b \in R \tag{6}$$

The computational complexity of this transform is one of its major disadvantages. Therefore, in practice to reduce this complexity, this transform is computed as discrete values of a and b. Discrete wavelet transforms are written based on equations (7).

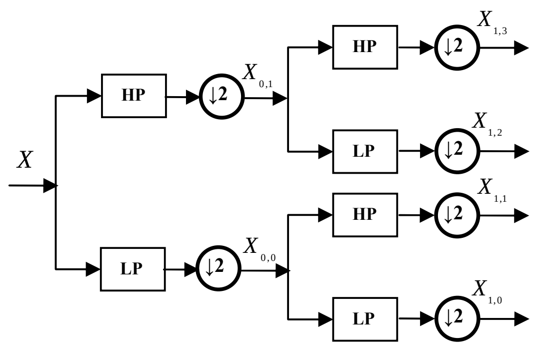

Fig. 1. Wavelet packet decomposition in two scales.

$$\psi_{j,k}(t) = 2^{j/2}\psi(2^j t - k)$$
$$x(t) = \sum_{k=-\infty}^{+\infty} c_k \varphi(t-k) + \sum_{k=-\infty}^{+\infty}\sum_{j=0}^{+\infty} d_{j,k}\psi_{j,k}(t) \tag{7}$$

In this equation $\varphi(t)$ is also called the function of the scale or the father wavelet, and the parameters c_k and $d_{j,k}$ are the coefficients of approximation and details, which can be calculated on the basis of a set of low pass and high pass filters. These filters can be calculated based on the mother and father wavelets according to equations (8).

$$\varphi(t) = \sum_k h_0(k)\sqrt{2}\varphi(2t-k)$$
$$\psi(t) = \sum_k h_1(k)\sqrt{2}\psi(2t-k) \tag{8}$$

In these equations, $h_0(k)$ and $h_1(k)$ are the impulse response of the low-pass and high-pass filters. Based on these filters, the desired signal can be expanded as wavelet coefficients.

For more practical implementation of wavelet and wavelet packet transforms, a filter bank is used. The full wavelet packet decomposition in two scales is shown in Fig. 1.

C. Set-membership filtering

In set-membership filtering (SMF) [18], the filter coefficient vector $\boldsymbol{W}$ is calculated to achieve a predetermined bound on the output error. In instant k of the adaptive process, there can be several different valid values for $\boldsymbol{W}$ that satisfies the chosen bound for the output error.

It is assumed that H_k is the set of all possible values for the $\boldsymbol{W}$ coefficients, for which the associated output error at time instant k is upper bounded in magnitude by γ , i.e.,:

$$H_k = \left\{ w \in R^N : \left| d_k - \boldsymbol{W}^T Z \right| \le \gamma \right\} \tag{9}$$

Based on the following set, the idea of set-membership filtering is obtained.

$$\psi_k = \bigcap_{i=1}^{k} H_i \tag{10}$$

The idea of set-membership filtering states that the coefficient vectors must be updated in such a way that it will always remain within the predefined feasibility set of equation (10) [19].

III. PROPOSED ALGORITHM

In this section, the proposed algorithm for short CP length will be provided. Reducing CP length improves channel utilization. Short CP lengths in multi-path channels will result in inter-symbol interference. In this paper, to increase the efficiency of channel use, inter-symbol interference is accepted and then using the PTEQ structure in the frequency domain and simultaneously with effective estimation of channel, the inter-symbol interference will be eliminated. The estimation and compensation of the channel effect assuming a short CP length cannot be modeled on the basis of the equations described in the previous section. The PTEQ compensation structure to overcome the ISI is originally derived from the integration of a time-domain equalizer (TEQ) and a frequency-domain equalizer (FEQ) [16]. The PTEQ structure, which is able to reduce the effective channel length and compensate for the channel effect in the frequency domain, by making adjustments commensurate with that used in this paper, is illustrated in Fig. 2.

The PTEQ structure has been introduced to increase the accuracy and reduce the complexity of the previous two-stage structure. In this structure, data equalizing is performed for each subcarrier in the frequency domain and performs at a lower sampling rate than the previous structure and thus significantly reduces the cost of its use. A PTEQ is a unified compensation structure, where equalization is performed individually on each subcarrier after taking the DFT of the received signal. In the PTEQ structure, a multi-tap filter is used for each subcarrier as an equalizer. Using this mechanism, the estimation of data in each subcarrier is based on the optimal design of the coefficients of these filters, showing equations (11) to (13) of the corresponding equations.

$$\tilde{S}^{(i)}[l] = W^{(i)}[l]\left(F_{ext}^{(i)}[l]z\right) \tag{11}$$

Where, the matrix of $F_{ext}[l]$ is defined as equation (12) and $L^{'} = L^{''} - 1$. Also, $L^{''}$ is considered as the length of the branches of the PTEQ structure.

$$F_{ext}[l] = \begin{bmatrix} I_{L^{'}-1} & 0_{L^{'}-1\times N-L^{'}+1} & -I_{L^{'}-1} \\ 0_{1\times L^{'}-1} & & F_N[l] \end{bmatrix} \tag{12}$$

In this matrix, the first row represents the difference and the second row the DFT matrix.

And again, the MSE criterion according to equation (13) is used to obtain the PTEQ coefficients.

$$\min_{W^{(i-1)}[l]} \mathbf{E}\left\{\left|S^{(i)}[l] - W^{(i)}[l]\left(F_{ext}^{(i)}[l]z\right)\right|^2\right\} \tag{13}$$

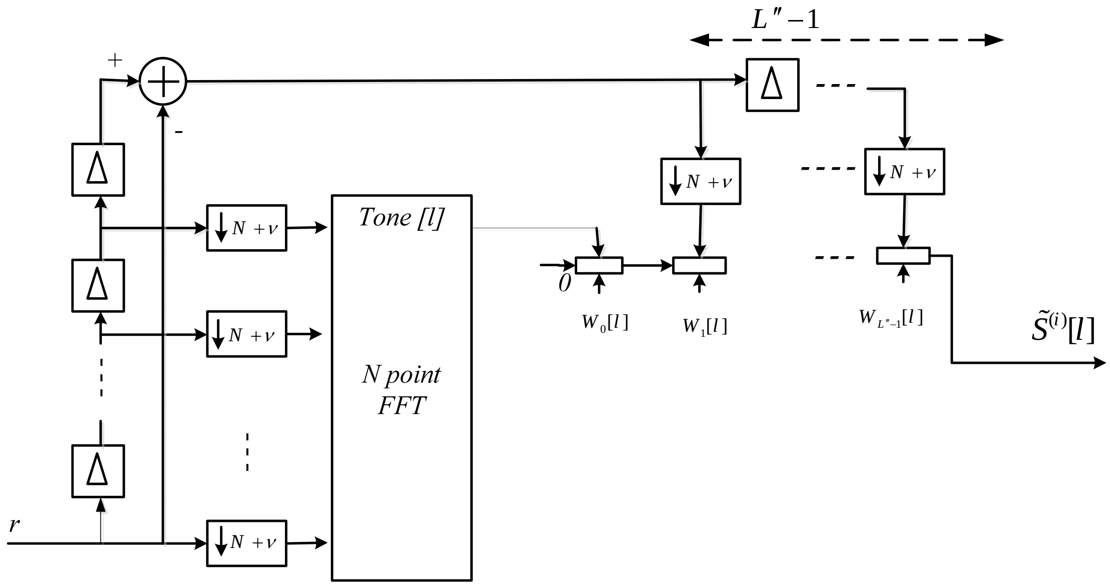

Fig. 2. PTEQ-OFDM receiver structure.

Equation (13) shows the compensator criteria in the PTEQ structure, which the coefficients can be learned by any adaptive algorithm. The speed and accuracy of estimation in communication systems and in channel estimation are essential. Therefore, before the development of adaptive filters, one of the orthogonal transforms in the proposed structure will be applied.

Due to the benefits of wavelet transform and considering that the convergence speed of the LMS algorithm in this domain is faster than the LMS algorithm applied in time domain, discrete cosine and Fourier transform domain [20], therefore, the wavelet transform in the proposed structure will be used. It has been shown previously in Reference [21] and in the system identification scenario that the use of adaptive filtering in orthogonal transform domain such as wavelet and DCT can increase convergence speed, and reduce steady state error. This improvement is due to the decrease in the self-correlation of the input signals. Therefore, to increase the speed of the algorithm in channel estimation, and using the wavelet transform denoising property, in the proposed algorithm, the wavelet transform is first applied to the data behind the branches in each subcarrier. The appropriate adaptive algorithm is then extended to each branch of the PTEQ structure.

After applying the wavelet transform on the PTEQ branches and on the data $F_{ext}^{(i)}[l]\mathrm{z}$, new data will be shown as $\left(F_{ext}^{(i)}[l]\mathrm{z}\right)_{WP}$.

Fig. 2 shows that each subcarrier must be equalized using a multi-tap adaptive filter.

In this paper, first the proposed WP-NLMS-PTEQ algorithm is developed for high speed channel estimation. Then, using set-membership filtering along with wavelet packet analysis is proposed to increase the coefficient training speed and reduce the average computation rate. This algorithm is

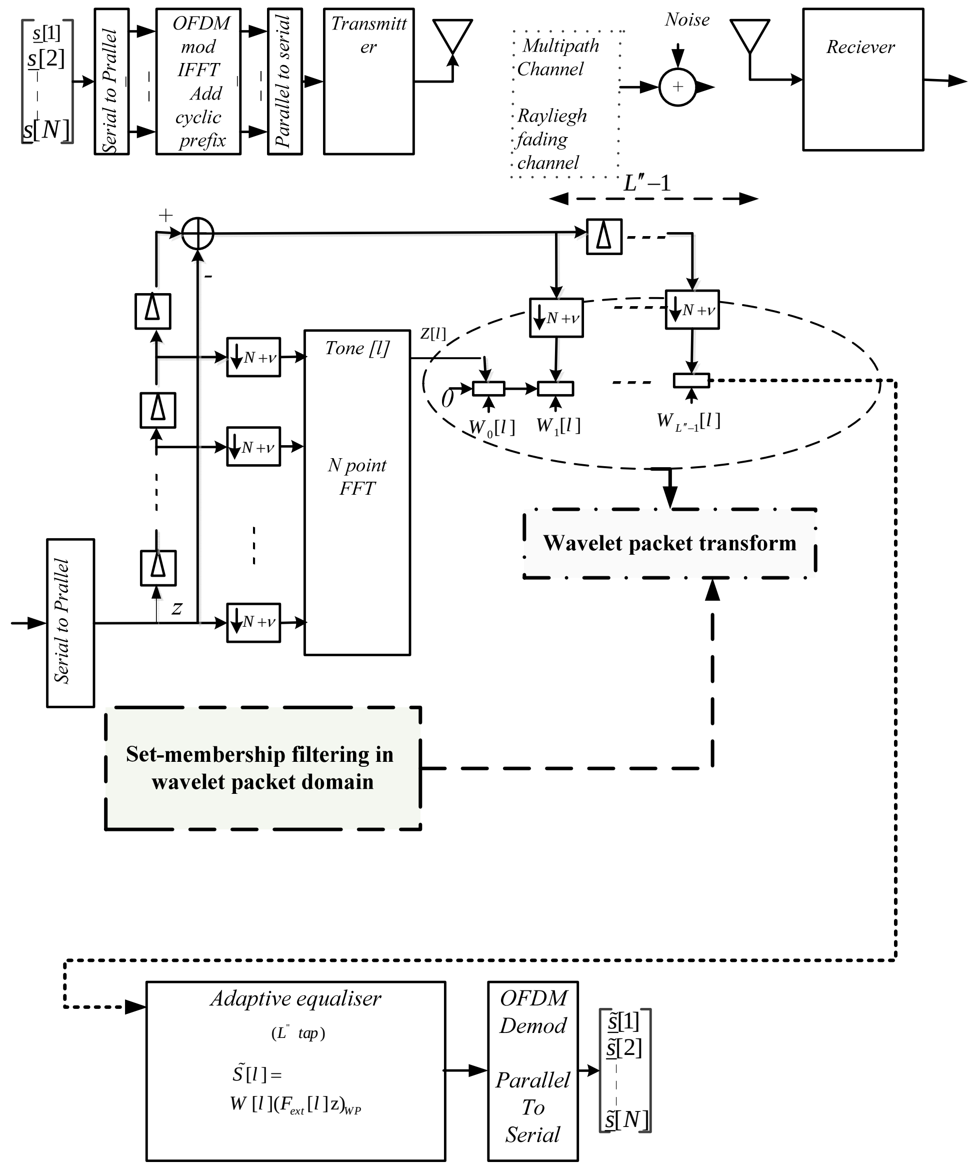

Fig. 3. SMF-WP-NLMS-PTEQ proposed algorithm block diagram

called SMF-WP-NLMS-PTEQ. Fig. 3 shows the block diagram of the proposed SMF-WP-NLMS-PTEQ algorithm, which is described below.

In the SMF method, when the error is less than the predetermined bound value, it is prevented from updating the coefficients and will consequently reduce the amount of computation .On the other hand, the convergence speed is also increased due to the use of the variable and noise dependent step size. The proposed algorithm will consider the estimation problem in the presence of short CP length.

In the case of short CP, NLMS-type algorithm is extended to the framework of the PTEQ structure for channel-effect compensation. Here, we obtain the NLMS technique in the PTEQ structure for a given error in the adaptive filter output. In a PTEQ structure, we can derive the output of the adaptive filter as $\tilde{S}^{(i)}[l]$ instant i by using (15), where $\boldsymbol{W}[l]=[W_0[l],W_1[l],...,W_{L''-1}[l]]^T$, is defined as an $L''\times 1$ filter coefficient vector, and $F_{ext}^{(i)}[l]\mathrm{z}$, is the $L''\times 1$ data vector. The NLMS algorithm is computable for each branch in the PTEQ structure and is derived by using the following constrained minimization problem similar to [22]:

$$\min_{\boldsymbol{W}^{(i+1)}[l]} \left\|\boldsymbol{W}^{(i+1)}[l]-\boldsymbol{W}^{(i)}[l]\right\|_2^2 \tag{14}$$

The limitation of the above criterion is as follows, in which $S[l]$ is a known pilot data.

$$(F_{ext}^{(i)}[l]\mathbf{z})_{WP}^T\boldsymbol{W}^{(i+1)}[l]=S[l] \tag{15}$$

In equation (15), $(F_{ext}^{(i)}[l]\mathrm{z})_{WP}$ represents the transform of $(F_{ext}^{(i)}[l]\mathrm{z})$ using the wavelet packet. The above equations are solved by minimizing the Lagrange coefficients and the cost function of equation (16):

$$\begin{aligned} &J^{(i)}(l)=\left\|\boldsymbol{W}^{(i+1)}[l]-\boldsymbol{W}^{(i)}[l]\right\|_2^2 \\ &+\lambda(S^{(i)}[l]-(F_{ext}^{(i)}[l]\mathbf{z})_{WP}^T\boldsymbol{W}^{(i+1)}[l]) \end{aligned} \tag{16}$$

By solving the equation (16) and minimizing the cost function, the initial update equation to calculating the filter taps can be written in equation (17):

$$\boldsymbol{W}^{(i+1)}[l]=\boldsymbol{W}^{(i)}[l]+\frac{(F_{ext}^{(i)}[l]\mathbf{z})_{WP}^T(\boldsymbol{W}^{(i+1)}[l]-\boldsymbol{W}^{(i)}[l])(F_{ext}^{(i)}[l]\mathbf{z})_{WP}}{\left\|(F_{ext}^{(i)}[l]\mathbf{z})_{WP}\right\|_2^2} \tag{17}$$

On the other hand we have:

$$\begin{aligned} &(F_{ext}^{(i)}[l]\mathbf{z})_{WP}^T\boldsymbol{W}^{(i+1)}[l]-(F_{ext}^{(i)}[l]\mathbf{z})_{WP}^T\boldsymbol{W}^{(i)}[l] \\ &=S[l]-(F_{ext}^{(i)}[l]\mathbf{z})_{WP}^T\boldsymbol{W}^{(i)}[l]=e^{(i)}(l) \end{aligned} \tag{18}$$

Thus, equation (17) can be written as (19). Also, the step size μ is inserted into the equation to control the stability and convergence rate.

$$\boldsymbol{W}^{(i+1)}[l]=\boldsymbol{W}^{(i)}[l]+\mu\frac{(F_{ext}^{(i)}[l]\mathbf{z})_{WP}\,e^{(i)}(l)}{\left\|(F_{ext}^{(i)}[l]\mathbf{z})_{WP}\right\|_2^2} \tag{19}$$

In a PTEQ structure, due to the large number of branches and, as a result, the high number of filter coefficients in the branches, the amount of computation needed to train the system is high. Therefore, it is necessary to reduce the computational complexity of this system. Therefore, the SMF-based algorithm for reducing the average computation and increasing the convergence speed is presented in the following section. Also, in the NLMS algorithm, the step size μ is limited to $0<\mu<2$.

A. Proposed SMF-WP-NLMS-PTEQ algorithm

To obtain the SMF-WP-NLMS-PTEQ algorithm in the PTEQ structure, the step size used in equation (19) is considered as a variable. The update equation of this algorithm is provided in equation (20). Equation (21) also shows the relation of the variable step size.

$$\boldsymbol{W}^{(i+1)}[l] = \boldsymbol{W}^{(i)}[l] + \alpha^{(i)} \frac{(F_{ext}^{(i)}[l]\mathbf{z})_{WP}}{\left\|(F_{ext}^{(i)}[l]\mathbf{z})_{WP}\right\|_2^2} e^{(i)}(l) \tag{20}$$

$$\alpha^{(i)} = \begin{cases} 1 - \dfrac{\gamma}{\left|e^{(i)}[l]\right|}, & if \ \ \gamma > \left|e^{(i)}[l]\right| \\ 0, & otherwise \end{cases} \quad , \gamma = \sqrt{5\sigma_n^2} \tag{21}$$

In equation (21), $\alpha^{(i)}$ is a variable step size and γ represents the error threshold value which can be calculated on the basis of noise variance as $\gamma = \sqrt{5\sigma_n^2}$. The parameter γ is achieved empirically [23]. In the SMF-NLMS algorithm, an upper threshold $\alpha^{(i)}$ is assumed to control and constrain the estimation error [23].

In the SMF (set-membership filtering) method, the coefficients are not updated at times when the error is less than the threshold value, thus reducing the amount of computation, significantly. In addition, the algorithm uses a noise-dependent variable step size, which can increase the convergence speed.

IV. SIMULATION RESULTS

In this section, the results of a number of simulations are shown to demonstrate the efficiency of the proposed algorithm. In the simulations, the FFT size is set to 64, the CP length is $\upsilon = 4$, and the data modulation is 64QAM and 16QAM. Multipath channel with $(L_{ch} + 1 = 16)$ taps is used, in which the taps are independently selected and have a complex Gaussian distribution. In addition, upper band of SMF-based algorithms is also considered to be $\gamma = \sqrt{5\sigma_n^2}$, where σ_n^2 is the noise variance and it is assumed to be known. In curves that do not use the SMF technique, the adaptive filter step size $\mu = 0.03$ is used. In all simulations, the decomposition level of wavelet transform packet will be 2. All results are simulated in a Rayleigh multipath channel with 16-tap length. Only 400 symbols are used to train the system in simulated curves. Every channel realization is independent of previous one and the bit-error-rate (BER) results are depicted from averaging the BER curves over 50 independent channels.

The branch lengths of the PTEQ structure are also considered to be 16. In the simulated figures, in addition to the normal state

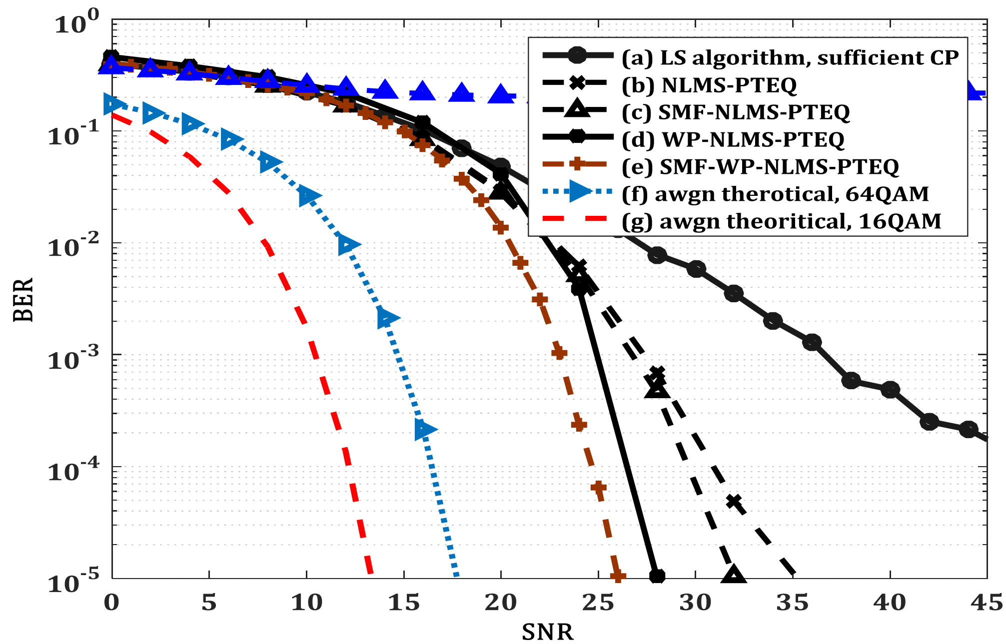

Fig. 4. Error performance curve, 16QAM signal, NLMS training and SMF-WP-NLMS-PTEQ under short CP

of the algorithm, the curves are compared with the LS algorithm used for channel estimation and with sufficient CP length.

Fig. 4 shows a plot of bit error rate (BER) versus signal-to-noise ratio (SNR) for the proposed SMF and WP-based methods for the NLMS-PTEQ algorithm in the PTEQ structure.

The modulation used is 16QAM and the results are compared with LS algorithm with sufficient CP and unequal conditions. The proposed SMF-NLMS-PTEQ, WP-NLMS-PTEQ and SMF-WP-NLMS-PTEQ algorithms, in addition to the LS algorithm, show good results in terms of BER than the conventional NLMS-PTEQ algorithm. Also in the curves (c) and (e) which used SMF technique, updating was done in 57 and 49% of the iterations, respectively. Therefore, the average computation of the estimator system has also decreased in proportion to its update rate. The best results are obtained when both the SMF and WP techniques are combined. All of these curves are compared to the ideal 16QAM and 64QAM modulations on the AWGN channel.

Fig. 5 also shows a plot of BER versus SNR for the proposed SMF and WP-based methods for the NLMS-PTEQ algorithm in the PTEQ structure. The modulation used is 64QAM and the results are compared with LS algorithm with sufficient CP and unequal conditions. The proposed SMF-NLMS-PTEQ, WP-NLMS-PTEQ and SMF-WP-NLMS-PTEQ algorithms, in addition to the LS algorithm, show good results in terms of BER than the conventional NLMS-PTEQ algorithm. Also, in the curves (c) and (e) which used SMF technique, updating was done in 61 and 54% of the iterations, respectively. Therefore, the average computation of the estimator system has also decreased in

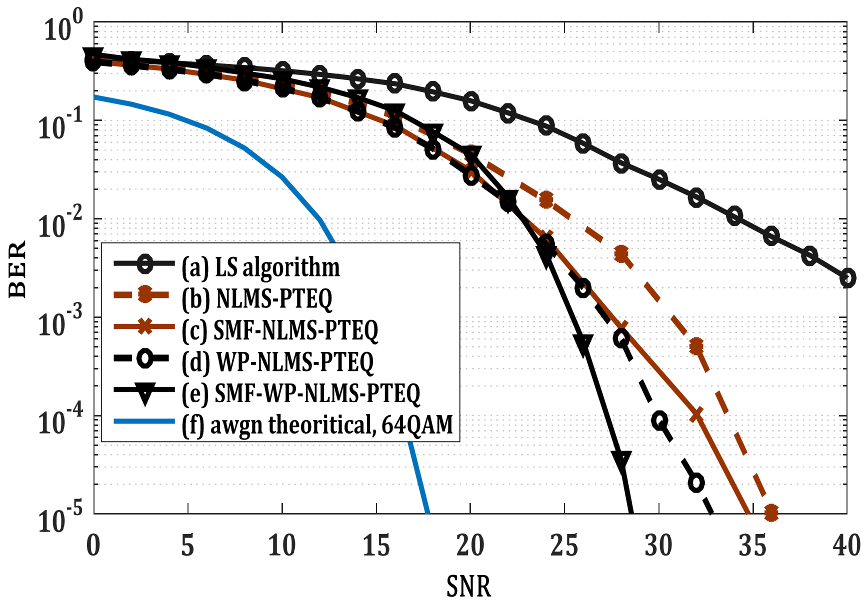

Fig. 5. Error performance curve, 64QAM signal, NLMS training and SMF-WP-NLMS-PTEQ under short CP

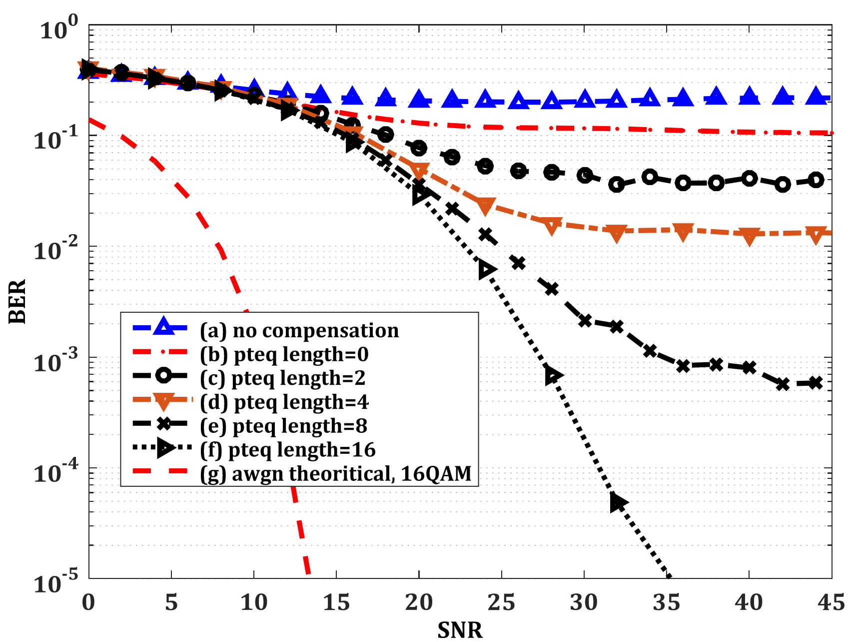

Fig. 6. Error performance curve, 64QAM signal, evaluation of reduced length of PTEQ structure branches under short CP

proportion to its update rate. The best results are obtained when both the SMF and WP techniques are combined. All of these curves are compared to the ideal 64QAM modulation on the AWGN channel.

In Fig. 6, the effect of decreasing L'' is investigated. By decreasing L'', the bit error rate performance will deteriorate and the ISI cannot be completely eliminated in the presence of a short CP.

As shown in Fig. 6, the error performance in the short CP, approaches the non-compensation state, and by increasing the branch lengths of the PTEQ structure, the ISI effects of the short CP length are eliminated. In this simulation, SMF-WP-NLMS-PTEQ algorithm is used for the curves (e) and (f). Also in curves (b), (c) and (d) the SMF-NLMS-PTEQ algorithm is used and the WP technique is not used due to the short length of the branches of the structure. However, the curves show that the use of PTEQ structure in short CP conditions is necessary to ISI effect elimination.

V. CONCLUSION

In this paper, a new effective adaptive based channel estimation has been proposed under short CP conditions. This algorithm is implemented using the structure of PTEQ. The PTEQ structure is capable of compensating channel effect efficiently under short CP length. The proposed method employs an adaptive algorithm in a more optimal way and its implementation in the wavelet transform domain has increased the convergence speed of the channel estimation. The use of PTEQ and wavelet transform along with set-membership filtering concept, has made the proposed algorithm well capable of estimating and compensating channel effects in short CP length. The average computational cost of the system has also decreased and the use of the system has become more cost-effective and implementable in real applications. Overall, the simulation results for the proposed SMF-WP-NLMS-PTEQ algorithm under short CP showed sufficient improvement in BER along with reduced computation, which makes this algorithm nearly ideal in terms of its performance.

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