Literature Survey in Normalised Adaptive Channel Equalizer for MIMO-OFDM

Saranya V.P\(^1\), B Bhuneshwari\(^2\)

PG Scholar Department of Communication Systems, PET Engineering College, India
Assistant Professor, Department of Communication Systems, PET Engineering College, India

Abstract — The chief goal in this paper is to provide an analysis on adaptive channel equalizers technique based on the varied approach. This given paper describes a survey on varied techniques for adaptive channel equalization and problems associated with individual operations. Inter-symbol Interference (ISI) is taken into account as a major obstacle in the wireless communication channel to transfer data. The purpose of the equalizer is to reconstruct the original signal by using filters or other methods, and removing the influence of the ISI, thus enhancing the reliability of data transmission. Existing minimum-symbol-error-rate equalizers were derived based on the symbol-error-rate objective function. Because of the complexity of the objective function the derivation isn’t simple. In the current system, MSER-based channel equalization is concentrated. In existing analysis, the MSER equalizer is derived based on the bit error rate (BER) or SER objective function, that are for BPSK sources and QAM sources, correspondingly. The BER/SER objective functions are non-convex and consequently it's tough to get a closed-form solution. The proposed paper describes comparison between approaches and strategies to estimate the novel blind channel estimation algorithm for a multiple input multiple output (MIMO).

Keywords— Analysis, Blind, Estimation, distorted, interference, MIMO.

1. INTRODUCTION

Recently introduced Multiple Input Multiple Output (MIMO) signal transmission schemes are attractive for recently rising high-speed data transmission wireless communication systems as a result of they provide an increased data throughput (capacity) while not increasing operational bandwidth [S.Chen et al., 2008, M.R. McKay et al., 2010]. Their other advantage is that they are capable to enhance the quality of signal transmission through the utilization of transmitter and receiver diversity. These benefits are potential under the condition that the MIMO channel state information (CSI) is obtainable at the receiver. Traditionally CSI is acquired by sending training sequences (pilot signals) equally spaced along a block of transmit symbols. So as to save the bandwidth and increase spectral efficiency, blind and semi-blind channel estimation strategies are applied to get the CSI.

Inter symbol interference (ISI) could be a limiting factor in several communication systems. ISI can arise from time-varying multi-path fading, which may be severe in, for instance, a mobile communication system. Other channel impairments that contribute to ISI consist of symbol clock jitter, carrier phase jitter, etc. To attain high-speed reliable communication, channel estimation and equalization are necessary to beat the effects of ISI [Rusek and Prlja, 2012].

OFDM systems have recently emerged as good candidates for future generation high-rate wireless systems because of their efficient utilization of bandwidth and robustness to multipath. In this paper it concentrate on blind channel estimation for OFDM systems. The literature along these lines are often classified as follows.

- Single-user OFDM systems with multiple transmit / receive antennae - Multiple transmit / receive antennas together with coding are accustomed to improve diversity and rate. Some representative schemes embrace the space time block codes (STBC) [S.Chen et al., 2006], space time trellis codes, and layered space time. In this case, the channel matrix is non-diagonal. A channel estimation scheme for such MIMO systems is that the two-input one-output MIMO OFDM system studied in [S.Chen et al., 2006], where STBC was applied at the inputs and also the system was estimated by exploiting the structure of the codes. This approach depends on transmission redundancy, i.e., every information symbol is transmitted twice in two consecutive time intervals through two different antennas. In
by reviewing recent surveys [Yeh and Barry, 2000, J.-Y. Li et al., 2007], the aim of this paper is to review some blind channel estimation approaches. It offers a systematic outline of some algorithms in the area of blind channel estimation. Numerous existing algorithms are classified into the moment-based and also the maximum likelihood (ML) strategies. If input is assumed to be random with prescribed statistics, the corresponding blind channel estimation schemes are considered to be statistical. On the other hand, if the source doesn’t have a statistical description, or though the source is random however the statistical properties of the source aren’t used, the corresponding estimation algorithms are deterministic [J.-Y. Li et al., 2007].

The paper further ordered as follows: Section II proposes Basic Description. Section III Algorithms Description whereas statistical result is given in section IV and conclusion is defined in section V.

2. BASIC DESCRIPTION

Minimum-Mean-Squared-Error (MMSE):

Conventional criteria for receive filter design, like the minimum-mean-squared-error (MMSE) criterion and also the least-square (LS) criterion, adopt the squared error as the performance measure. These criteria will minimize the mean or cumulant of the squared error between the filter output and also the target signal. Though, in practice it’s the system’s symbol error rate (SER), not the mean squared error (MSE) that actually matters. It’s been illustrated by numerous simulations that minimizing the MSE doesn’t essentially produce the minimum SER (MSER) performance. Furthermore, the theoretical analysis on MMSE receiver additionally indicates that, when channel coding isn’t applied, the possible SER of MMSE receiver isn’t optimal. The MSER criterion that is directly based on minimizing the SER, has attracted increasing attention in the past two decades. The MSER criterion are often derived back to the error rate analysis in multipath channel. Recently the MSER criterion has shown promising performance in numerous analysis fields, together with channel equalization, multiuser detection, beam forming, power control/allocation, carrier phase recovery, timing recovery and precoding. A thorough review on the application of the MSER criterion is found in. In this paper it concentrates on MSER-based channel equalization.

It will assume a coherent symbol-spaced receiver front-end, in addition to specific knowledge of the signal phase and symbol timing, such the channel are often approximated by identical, discrete-time, baseband model, wherever the transmit filter, the channel, and also the receive filter, are delineated by a discrete-time linear filter, with finite-length impulse response

\[ h[n] = \sum_{k=0}^{M-1} h_k \delta[n-k] \]  

(1)

of length M. The coefficients \( h_k \) are assumed to be time-invariant and known to the receiver that the MMSE criterion is evaluated over both the distribution of the noise in addition to the distribution over the symbols.

It proposes a fast LMS/Newton algorithm that mixes simplicity of the LMS and also the fast convergence of the Newton algorithm. When the autocorrelation input process matrix eigen value dispersion is large and also the performance surface contour is far away from perfect circle, the LMS/Newton algorithm illustrates well
convergence characteristics. Newton technique has high convergence speed.

Various channel equalizer model has the subsequent properties:

1. The MSER criterion is reformulated as the constraint of correct symbol detection. It doesn't ought to calculate the BER/SER expressions and thus includes a simpler derivation.

2. For various source modulation schemes, the proposed model applies a same objective function (with different formulations of constraints). Thus it's an even framework for various source modulation schemes. In this paper it illustrates that BPSK and QAM sources have an identical derivation. Against this, in existing work the BER/SER expressions for difference modulation techniques ought to be derived individually.

The proposed constrained optimization drawback is solved with the Lagrange multiplier technique, which ends in an adaptive algorithm that mimics the classical Normalized Least-Mean-Square (NLMS) algorithm. Compared with existing AMBER equalizers, the proposed equalizer doesn't involve channel parameters and thus includes an easier structure. Additional, a normalization factor is introduced to the proposed equalizer. Simulation results illustrate the new algorithm includes a faster convergence than the adaptive equalizers.

3. REVIEW ON VARIOUS SYSTEMS

Tüchler et al. has mentioned regarding Turbo equalization is an iterative approach to this drawback, in which a maximum a posteriori probability (MAP) equalizer and a MAP decoder exchange soft information in the kind of previous probabilities over the transmitted symbols. A number of reduced-complexity strategies for turbo equalization have recently been introduced in which MAP equalization is replaced with suboptimal, low-complexity approaches. In the given paper, it explores a number of low-complexity soft-input/soft-output (SISO) equalization algorithms based on the minimum mean square error (MMSE) criterion. This embraces the extension of existing approaches to general signal constellations and also the derivation of a novel approach requiring less complexity than the MMSE-optimal solution. All approaches were qualitatively analyzed by observing the mean-square error averaged over a sequence of equalized data. It illustrate that for the turbo equalization application, the MMSE-based SISO equalizers perform well compared with a MAP equalizer whereas providing an incredible complexity reduction.

Qilian Liang and Jerry M. has outlined about a new kind of adaptive filter: type-2 fuzzy adaptive filter (FAF); one that's realized using an unnormalized type-2 Takagi–Sugeno–Kang (TSK) fuzzy logic system (FLS). It apply this filter to equalization of a nonlinear time-varying channel and demonstrate that it will implement the Bayesian equalizer for such a channel, includes a easy structure, and provides fast inference. A clustering technique is accustomed to adaptively design the parameters of the FAF. Two structures are used for the equalizer: transversal equalizer (TE) and decision feedback equalizer (DFE). A new decision tree structure is accustomed to implement the decision feedback equalizer, in which every leaf of the tree could be a type-2 FAF. This DFE immensely reduces computational complexity as compared to a TE. Simulation results illustrate that equalizers based on type-2 FAFs perform much better than nearest neighbor classifiers (NNC) or equalizers based on type-1 FAFs.

WeiShi et al. had mentioned to the problem of inter-symbol interference in communication channel, an adaptive equalization algorithm based on the new quasi-Newton technique is proposed. Using the positive definite symmetric Hesse matrix iteration formula rather than auto correlation function inverse matrix, the proposed algorithm beats the influences caused by the step factor and also the signal auto correlation function estimation on the convergence speed and steady-state error. Simulation results illustrate that the proposed algorithm has fast convergence speed and low bit error rate within the large step.

Atapattu et al. had mentioned regarding Linear adaptive channel equalization using the least mean square (LMS) algorithm and also the recursive least-squares (RLS) algorithm for an innovative multi-user (MU) MIMO OFDM wireless broadband communications system is proposed. The proposed equalization technique adaptively compensates the channel impairments caused by frequency selectivity in the propagation environment. Simulations for the planned adaptive equalizer are conducted employing a training sequence technique to determine optimal performance through a comparative analysis. Results show an improvement of 0.15 in BER (at a SNR of 16 dB) when using Adaptive Equalization and RLS algorithm compared to the case in which no
equalization is used. Generally, adaptive equalization using LMS and RLS algorithms showed to be considerably beneficial for MU-MIMO-OFDM systems.

Constantinos et al. has proved a new system with improved processing on new adaptive equalization algorithms for direct sequence code division multiple access (DS-CDMA) systems operating over time-varying and frequency selective channels. The equalization schemes contain a variety of serially connected stages and find users in an ordered manner, applying a decision feedback equalizer (DFE) at every stage. Both the equalizer filters and also the order in which the users are extracted become of recursive least squares (RLS) manner, efficiently accomplished through time- and order-update recursions. V-BLAST detection ordering is enforced, that is, the stronger signal is extracted first so the weaker users are often more simply detected. The spreading codes are unavailable at the receiver of the first scheme, wherever as the second algorithm uses the RAKE receiver idea, incorporating knowledge of the spreading sequences to offer performance improvement. The bit error rate (BER) performance of the equalizers is estimated via simulations, in each mild and severe near-far environments. Their superiority over existing methods is demonstrated.

While Labat et al. had outlined regarding a novel unsupervised (blind) adaptive decision feedback equalizer (DFE). It may be thought of as the cascade of four devices, whose main components area purely recursive filter(R) and a transversal filter (T): Its main feature is that the ability to handle severe quickly time-varying channels, in contrast to the conventional adaptive DFE. This result is acquired by permitting the new equalizer to change, in a reversible way, both its structure and its adaptation consistent with some measure of performance like the mean-square error (MSE). In the starting mode, R comes first and whitens its own output by means that of a prediction principle, whereas T removes the remaining inter symbol interference (ISI) due to the Godard (or Shalvi–Weinstein) algorithm. In the tracking mode the equalizer becomes the classical DFE controlled by the decision-directed (DD) least-mean-square (LMS) algorithm. With identical computational complexity, the new unsupervised equalizer shows identical convergence speed, steady-state MSE, and bit-error rate (BER) as the trained conventional DFE, however it needs no training. It’s been enforced on a digital signal processor (DSP) and tested on underwater communications signals—its performances are very convincing.

Based on the above discussion many authors have listed regarding their effectiveness of the system relatively outlined that an adaptive system may be an improved choice for perfect channel equalizer selection.

4. STASTICAL RESULTS AND APPRAOCH

Subspace-Based Parameter Estimation Scheme:
Sample estimates $G_i$, of the noise eigenvectors are obtainable and solved in the least squares sense and results in minimize the subsequent quadratic form:

$$\sum_{i=0}^{LN-M-N-1} |GH_i|^2$$  \hspace{1cm} (2)

As it can see, q(H) rely on vector H instead of on the filtering matrix H N. This can be conveniently done by application of following Lemma, which needs the subsequent notations. Notations: Let $V^{(0)},...,V^{(L-1)}$ be L arbitrary N×1 vectors and let V be the LN×1 vector outlined as $V = \left[ V^{(0)T},...,V^{(L-1)T} \right]^T$.

$$V_{M+1} = \begin{bmatrix} V_0^T & \cdots & V_{N-1}^T \cdots & 0 \\ \vdots & \ddots & \ddots & \ddots \\ 0 & \cdots & 0 & V_0^T \end{bmatrix} \hspace{1cm} (3)$$

$$V_{M+1} = \begin{bmatrix} V_{M+1} \cdots V_{M+1} \end{bmatrix}^T \text{ Dim } L(M+1)X(M+N) \hspace{1cm} (4)$$

By theorem 1, if true autocorrelation matrix was obtainable, the true channel coefficients are the unique (up to a scalar factor) vector H such q(H) = 0. In contrast, when solely an estimate of the autocorrelation matrix is obtainable, the quadratic form has not precisely rank $L(M+1)$. Thus, estimation of H may be obtained by minimizing q(H) subject to a properly chosen constraint avoiding the trivial solution H = 0. Different constraints on H offer different solutions. It have classically thought
of minimization subject to linear and quadratic constraints:
- Quadratic constraint: Minimize \( q(H) \) subject to \( |H| = 1 \). The solution is the unit-norm eigenvector related to the smallest eigen value of matrix \( Q \).
- Linear constraint: Minimize \( q(H) \) subject to \( cHH^T = 1 \). Where \( c \) could be a \( L(M + 1) \times 1 \) vector. The solution is proportional to \( Q^{-1}C \) [N.B.Jayaraj, 2010].

The first choice is more natural however involves the computation of an extra eigenvector. The second solution depends on the choice of an arbitrary constraint vector \( c \). The computational cost of the second solution is lower since it amounts to solving a linear system instead of extracting an eigenvector [N.B.Jayaraj, 2010].

Signal Subspace:

It is shown before that minimizing a constrained quadratic form involving the noise eigenvectors provides the channel coefficients. This quadratic form is equivalently rewritten in terms of the signal eigenvectors as:

\[
q(H) = N |H|^2 - \sum_{i=0}^{M+N-1} |S_i H_N|^2
\]  

(5)

Deterministic Subspace Approach:

Subspace technique based on the property that the channel is in a unique direction. It might not be robust against modeling errors, particularly when the channel matrix is close to being singular. The second disadvantage is that they’re typically more computationally expensive.

Without presence of noise, the estimator produces the precise channel using solely a finite number of samples if the identifiability condition is satisfied. Thus, these strategies are most effective at high SNR and for small data sample applications. On one hand, deterministic strategies are often applied to a much wider range of source signals; on the other hand, not using the source statistics affects its asymptotic performance

5. CONCLUSION

Channel estimation is a standard linear system identification problem with the training sequence as the pilot input signal. In several applications, the pilot signals might not be simple to use or they will present an additional problem, for instance requiring more bandwidth in communication systems. Blind channel estimation and equalization eliminates the necessity for a pilot signal and simplifies the necessities for channel estimation and equalization. Particularly, recent developments in blind estimation research have led to a class of rapidly converging and data efficient algorithms that may effectively estimate the channel with a small number of data points. In this paper, it reviewed some of the basic approaches in blind estimation and identified blind channel estimation needs an effective decision making system for processing.

REFERENCES


IJSRET @ 2014


