Literature Review on BER Improvement of Adaptive Equalizer

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ABSTRACT

Channel equalization is an important aspect in high speed digital communication required for efficient and reliable data recovery and reception when the data is transmitted over band-limited channel subjected to noise and interference. We investigate channel equalization and introduce hierarchical and adaptive nonlinear channel equalization algorithms that are highly efficient and provide significantly improved bit error rate (BER) performance. Due to the high complexity of nonlinear equalizers and poor performance of linear ones, to equalize highly difficult channels, we employ piecewise linear equalizers.

1. INTRODUCTION

Channel equalization is a channel impairment improvement technique, which compensates for the signal distortion and noise caused due to multipath in time dispersive channels. The digital data is fed into a channel, which can be modelled as an adaptive delay-tapped transversal filter having certain filter coefficients [1]. Due to band-limited, dispersive channel and multipath fading, the transmitted symbols overlap with each other and is distorted termed as intersymbol interference (ISI). In a wireless communication channel when the modulation bandwidth is exceeding the coherence bandwidth ISI takes place as the transmitted pulses are spread into the adjacent symbols. To combat the effects of ISI and noise and to reconstruct the signal and minimize Bit Error Rate (BER), the adaptive channel equalizer is used at the receiver end [4]. When the training is complete transfer function of the equalizer becomes inverse to that of the channel and the filter coefficients are adaptively optimized using adaptive optimization techniques so that the output of the equalizer (estimated signal) matches to that of the delayed version of the transmitted signal (desired signal). Thus, adaptive channel equalization can be viewed as an iterative optimization problem where the objective is to minimize the mean-square error (MSE) such that an estimate of equalizer coefficients is obtained which nullifies the effects of ISI and noise on the signal transmitted through the channel.

Adaptive channel equalization is required as the wireless communication channels are unknown, non-stationary and time-varying channels. Since the adaptive channel equalizer compensates for the effects of the non-linear timevarying channel, a suitable adaptive optimization algorithm is to be applied for updating the equalizer coefficients and thus tracking the variations of the channel. In the recent past, the adaptive channel equalization is developed using soft computing approaches such as evolutionary and swarm intelligence algorithms compared to conventional learning techniques such as Least Mean Squares (LMS), Least Mean Fourth (LMF) and Recursive Least Squares (RLS) and their variants where there is possibility of solution being trapped by local optima. Moreover, there is performance degradation of gradient-based algorithms for non-linear channels.

1.1 Adaptive Channel Equalization Model

Channel equalization is a key area in a digital communication system where the objective is compensation for the channel distortion, which can be achieved by minimization of squared error between the equalizer output and the delayed version of the transmitted signal. The equalization in digital communication scenario is illustrated in Figure 1, where (k) represents the symbol sequence transmitted through the non-linear channel.

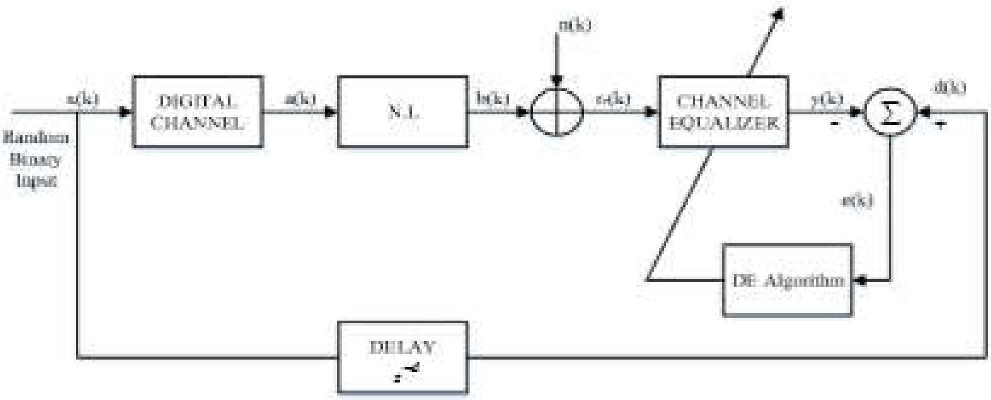

Figure 1 Schematic block diagram of channel equalization

The Additive White Gaussian Noise (AWGN) is the channel noise contaminated to the channel output. This output of the channel acts as input to the adaptive non-linear equalizer. The output of the channel equalizer (k) is subtracted from the delayed version of the desired signal (k) to compute the error $e(k)$. The square of error e 2 (k) is considered as cost function, which is to be minimized such that the equalizer output matches with delayed transmitted source signal.

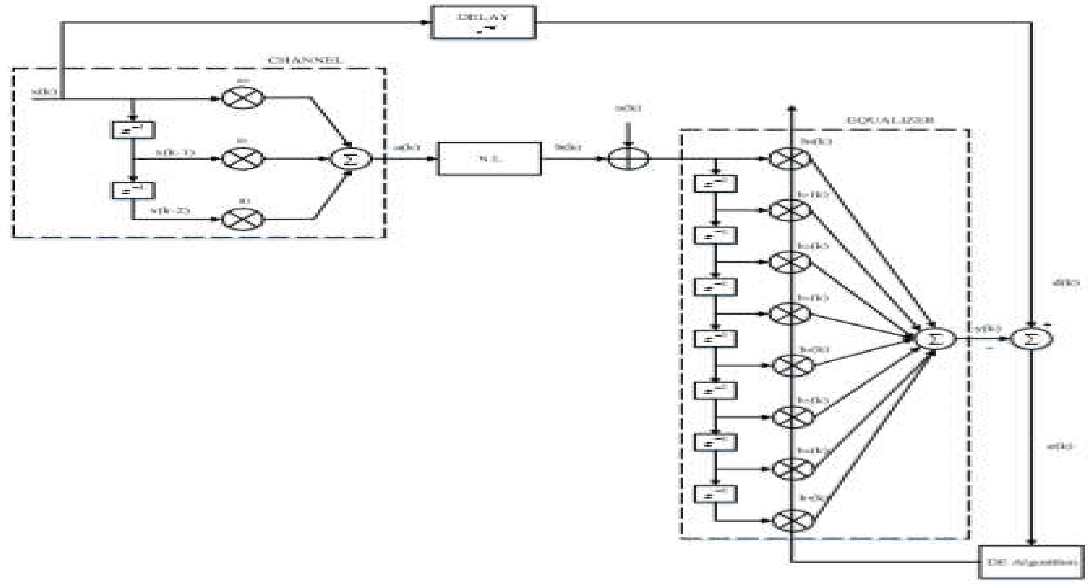

Figure 2 Adaptive channel equalizer models

The coefficients of the equalizer are iteratively updated using DE algorithm to achieve the best possible minimum squared error. Digital communication channels are often modeled as low pass FIR filter. Figure 2 shows the digital channel as a 3-tapped delay filter whose output is associated with nonlinearities and noise, hence it is highly distorted. Therefore, to restore, the transmitted signal the output of the channel is passed as input to the equalizer, which is also modelled as an adaptive delay tapped filter.

2. LITERATURE REVIEW

In [1], the pilot arrangement for OFDM system has been investigated. The comb type pilot channel estimation is implemented for the channel interpolation. IDFT (Inverse Discrete Fourier Transform) and zero padding are used to convert frequency domain to time domain. DFT (Discrete Fourier Transform) is used to convert time to frequency domain. For low Doppler frequencies, the decision feedback estimations performance is slightly worse than the low-pass interpolation channel estimation.

[2] Proposed our system to compare bit error rate of different modulation schemes, channels and channel equalization. We consider rapidly varying doubly selective channels, such that the channel coherence time is less than one OFDM symbol duration.

In [3], a low rank Wiener filter based channel estimator is proposed to reduce the complexity. This optimal estimator avoids large-scale inverse matrix operation, so MMSE estimator complexity is reduced. Moreover this estimator transmits two training blocks instead of one training block of data. This estimator also pre-calculates the singular value decomposition (SVD) of the channel correlation matrix.

[4]In this paper, the main objective is to demonstrate a Discrete Wavelet Transform (DWT) based video watermarking because video piracy is the major challenge when dealing with copyright protection. For watermarking video we use an image and random signals with different power coefficients are used.
[8]We proposed a low complexity algorithm for equalization, which uses the estimated BEM coefficients directly without creating the channel matrix and equalization method require O (K log K) in operations and O (K) in memory.
[12]Hyper spectral imagery provides the potential for more accurate and detailed information extraction than possible with any other type of remotely sensed data. The main objective of this research paper is to study various techniques used in Change Detection for hyper spectral Images.
[14]OFDM is found as a key multicarrier tweak procedure. It offers various favorable circumstances, for example, high ghostly productivity, low usage unpredictability, less helplessness to echoes and non–linear bending. Because of these points of interest, OFDM innovation is immensely utilized as a part of different correspondence frameworks.
[16]In this methodology, reckoning of ideal PTS weight variables by means of comprehensive pursuit obliges exponential many-sided quality in the quantity of sub pieces. Subsequently, a productive calculation for figuring the ideal PTS weights are acquainted with the point with diminish multifaceted nature contrasted with thorough.

3. PROCEDURE OF CHANNEL EQUALIZATION

The channel equalization using DE is discussed through the following steps:
Step 1: The channel coefficients are initialized. Random binary input (k samples) is generated and passed through the channel.
Step 2: The output of the channel added with AWGN of certain SNR is passed through a nonlinear channel.
Step 3: The population of parameter vectors corresponding to equalizer coefficients are initialized randomly. First target vector is taken from NP number of vectors, which consists of p no. of parameters.
Step 4: The nonlinear channel output subject to noise and distortion is passed as input to the equalizer. Thus, the estimated output of the equalizer is computed.
Step 5: The delayed transmitted signal is considered as the desired signal.
Step 6: The difference between the estimated output of the channel equalizer and the desired signal gives the error signal. Thus, k no. of error signals is generated and the mean of the squared error gives the MSE and this process is repeated for NP no. of times.
Step 7: The mutation, crossover and fitness evaluation and selection processes are carried out
Step 8: The above steps are repeated iteratively until MSE decreases gradually. Once the MSE further ceases to decrease and attains the lowest level all the parameters become identical and the stopping criterion is met.

4. PROBLEM FORMULATION

In telecommunication, channel impairments are the foremost barrier in broad band wireless applications. For this purpose the execution of a filter explicitly adaptive in nature is required in order to model the unidentified wireless channel and to carry out inverse modelling like adaptive equalization. The fundamental initiative of equalization is merely to balance for non-ideal features in wireless channels by stir up supplementary filtering. An adaptive filter can be defined as a filter whose features can be customized to attain several goals and is frequently understood to achieve this change (or "adaptation") without human intervention.

The implementation of static equalizer is very simple and less complexity has been observed but it is not efficient and reliable as compared to adaptive equalizer. It is difficult to design an equalizer unless and until if we don't know the transfer function of transmission system and impulse response of channel. There is a natural trade-off between diversity gain and multiplexing gain, increase in number of antennas improves diversity gain but multiplexing gain decreases. Even though these 4x4 SM Q-OFDMA systems achieve better performance than SISO systems, BER increases than compared to 2x2 STBC OFDMA systems.

The idea is to deploy Quasi Orthogonal STBC codes with the LDPC codes can be able to support four transmit and receive antennas and attaining the Shannon limit with a low BER. This can furthermore increases the system capacity with a maximum possible gain. To increase the quality of the signal at both transmitter and receiver, coding has done. We present various coding techniques to improve the SNR and BER using low complexity approach. To address the issue of decoding complexity in space time trellis codes, Space– time block coding was developed.

At the decoder, the extrinsic information is fed as a priori information. Based on this hard decision is made. Here Interleaver / Deinterleaver module shuffles the coded bits to decorrelate the error and make the error correct as possible. For channel estimation, the training symbols and soft decoded data are utilized to track the channel frequency response and suppress the ISI caused by channel estimation errors in QOFDMA systems. Estimation, equalization and decoding algorithms according to the performance /complexity trade-off shown that, the iterative receiver approach can improve the QOFDMA systems BER performance with acceptable complexity.

4.1 Equalization Techniques

There are many types of equalizers but they all fall into two broad categories: linear and non-linear. Fig. 4.1 show different equalizer structures and algorithms used. DFE is the most common of the non-linear equalization techniques because it is the simplest of the non-linear equalizers and it generally performs well provided the SNR is not at low levels otherwise the DFE suffers from error propagation. MLSE is the optimal technique but the complexity of this technique grows exponentially with the length of the delay spread.

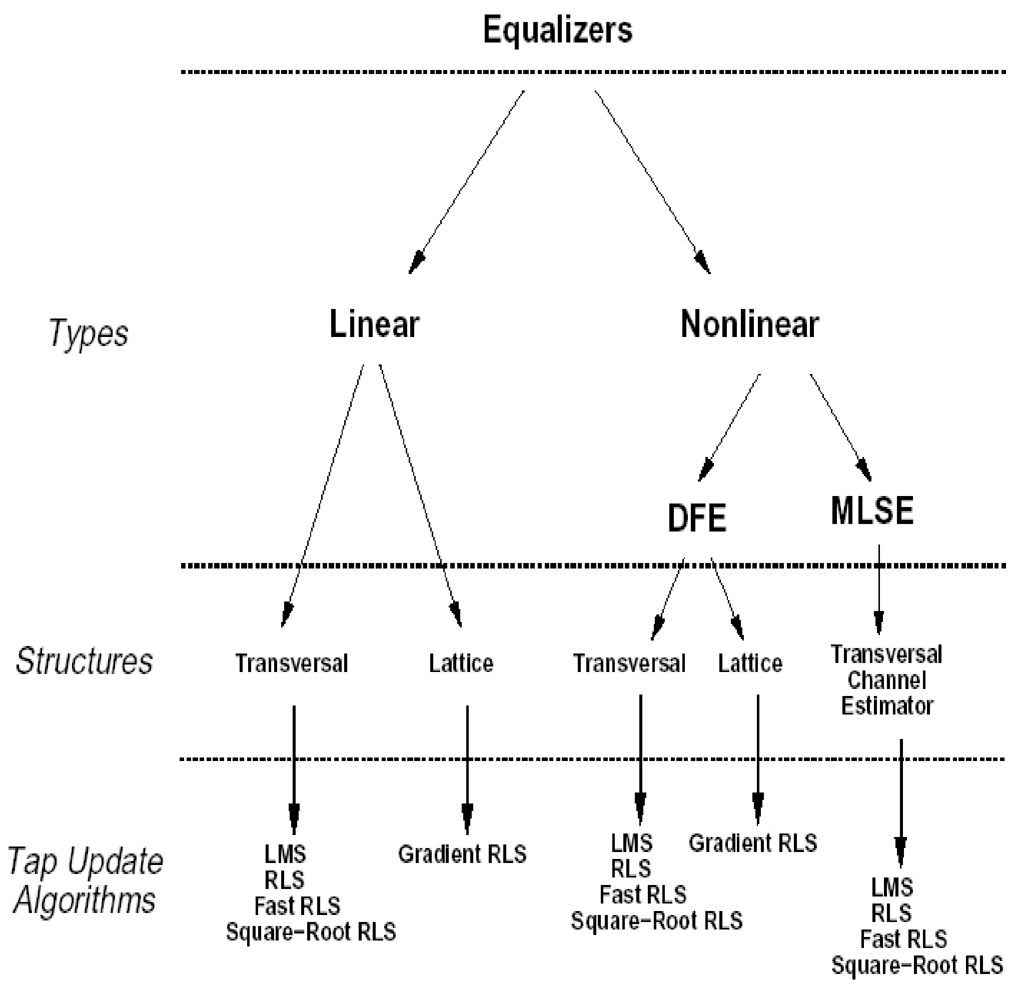

Figure 4.1 Equalizer Types, Structures, and Algorithms

4.2 Minimum Mean Square Error Equalization

This technique of equalization attempts to minimize the average mean square error between the transmitted signal (desired signal d_k) and the predicted received (actual signal d'_k) signal at the output of the equalizer. In other words, one is minimizing $E[d_k-d'_k]^2$. Since the minimum mean square error is linear, the output $\hat{d}_k$ also is linear and can be represented as a combination of the input samples y_k:

A standard problem in equalizers is the noise, and in particular white noise. But because of the matched filter in the equalizer, the white noise is not a problem but rather coloured noise poses a larger issue to compensate. For ease of explanation, one can expand the equalizer into two components, a noise whitening component and an ISI removal component. Thus the new block diagram for the channel and equalization will look as follows in Figure 4.2

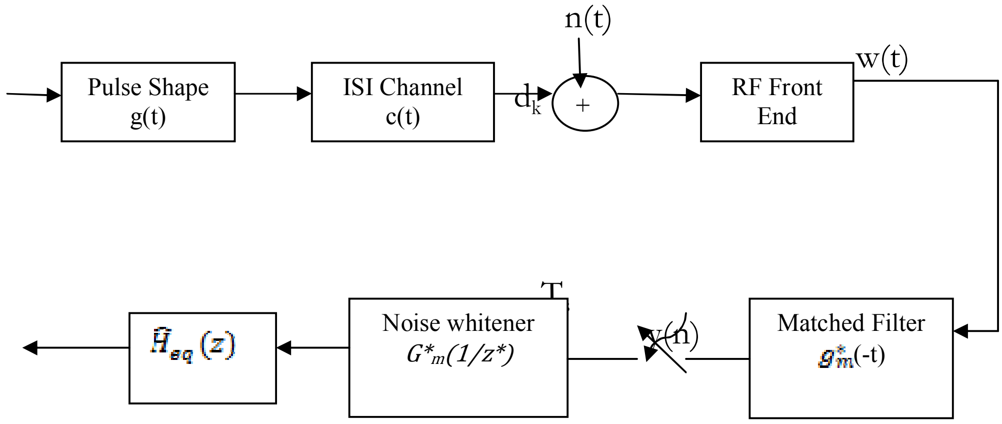

Figure 4.2 MMSE equalizer with noise whitening filter

The noise whitening filter will cause the noise at the output to have a constant power spectrum. It is obvious that if the white noise has a power spectrum of then if the noise whitening filter is, the resultant power spectrum will be constant at N_0. Now, the filter needs to be designed to compensate for the ISI removal component. Similarly as with the Zero Forcing (ZF) equalizer, one can assume the filter $\hat{H}_{eq}(z)$, with input v_n, is a linear filter with N=2L+1.

5. CONCLUSION & FUTURE SCOPE

5.1 Conclusion

In a related recent work, it is shown that the complexity of the receivers can be further simplified by performing the equalization at a per-frequency-bin level rather than at a block level. The use of finite-precision arithmetic in IIR filters can cause significant problems due to the use of feedback, but FIR filters have no feedback, so they can usually be implemented using fewer bits, and the designer has fewer practical problems to solve related to non-ideal arithmetic. They can be implemented using fractional arithmetic. Unlike IIR filters, it is always possible to implement a FIR filter using coefficients with magnitude of less than 1.0. This is an important consideration when using fixed-point DSP's, because it makes the implementation much simpler. The comparisons have been made by using MATLAB 14 tool by comparing the BER.

5.2 Future Scope

Researches on blind equalizations are going on but nobody wants to improve the performance by using training sequence equalization because it consumes lots of bandwidth. Small length training sequence can be generated by using duo binary signaling i.e. reduce the length of training sequence and combine it with delayed version to make it full length training sequence. When dealing with a channel which is rapidly varying with time anyone may use Kalman filter that need training sequence to update its weights. Training sequences needs to be short in length so that utilization of bandwidth can be achieved.

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