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ADAPTIVE EQUALIZER: EXTENSION TO COMMUNICATION SYSTEM

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ABSTRACT

In Digital Communication system the information or the thought is encoded digitally as discrete signals and then is electronically transferred to the recipient. Adaptive Equalization can be used to improve digital data transmission on wireless links with time-varying multipath distortion.Adaptive Equalizer is an equalizer that automatically adapts to time-varying properties of the communication channel. It is used frequently with coherent modulations such as phase shift keying, mitigating the effects of multipath propagation and Doppler spreading.

KEYWORDS: Bit Error Rate, Inter-Symbol-Interference, Finite Impulse-Response, Adaptive Gain Equalizer, Decision Feedback Equalizer, Quadratic Phase Shift Keying.

INTRODUCTION

Linear reference model has been derived for Zero Forcing (ZF) and Minimum Mean Square Error equalizer by Markus Rupp [1] . Moreover, equalizer problems based on Least Square (LS) approach has also been proposed in [1]. These problems also includes different channel modes, multiple antenna and multiple users. Using this model, iterative and recursive form of equalizer i.e robust has been achieved in [1]. It has also been concluded that Robust Adaptive Equalizer is more stable. In [1] ISI, DFE coefficients and BER has not been targeted which have been targeted in [2] where Look-Ahead Decision Feedback Equalizer (LADFE) has been introduced in [2]. Moreover, Eye Opening Monitor (EOM) has been used in [2]. This EOM measures the magnitude of received signal with different data patterns. In [2] histograms of equalized waveform is also obtained. Inter-Symbol Interference (ISI) induced by channel is estimated in [2]. Moreover, DFE coefficients has also been targeted in [2]. An improvement in Bit Error Rate (BER) has also been observed in [2] for PCB channel lengths 10-,20-,30-,40-cm. In [2], low cost and area saving has not been targeted so switching to [3] in which these are achieved in which an adaptive equalizer with an active source degeneration capacitor has been proposed in [3]. This proposal is based on Miller's Theorem. It is composed of Metal-Insulator-Metal (MIM) capacitor and a sub-amplifier. However, low cost and area saving has been targeted in [3].

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Moreover, by widening the compensation range, ISI compensation has also been achieved in [3]. In [3], faster convergence and tracking capabilities has not been targeted which has been targeted in [4]. In this paper, a new heuristic algorithm for sparse adaptive equalization problem, termed as Stochastic Gradient Pursuits is proposed in [4]. This proposal uses Decision Feedback Equalizer (DFE). DFE is used to mitigate the effects of long multipath channel. Unknown sparsity order parameter has been developed in [4]. Moreover, faster convergence and improved tracking capabilities has also been achieved in [4]. High speed, advantages of power consumption and chip size is not focused in [4] which has been targeted in [5]. Meanwhile, Adaptive Continuous Time Linear Equalizer (CTLE) based on asynchronous undersampling histogram has been proposed in [5]. Optimal equalizing fliter coefficient among various values has been selected by CTLE. Furthermore, it is based on indirect monitoring of data eye opening. This proposal is reliable for various sample size. High speed is targeted in [5] as clock path is eliminated. It also eliminates phase rotary circuitry. A great advantages in power consumption and chip size has been achieved in [5]. The effects of mismatches and its reduction method has not been targeted in [5] which has been targeted in [6] where a method to correct mismatches in an Analog discrete-time Finite Impulse-Response(FIR) filters used as equalizers in digital communication receivers have been introduced in [6]. Using parallel sample-and-holds (S/Hs) and time-interleaved equalizer channels, FIR can be implemented for fast speeding application. The equalizer performance has degraded by the mismatches in parallel S/Hs. Moreover, the mismatches of DC offsets, gain errors, sample-time errors and bandwidth in the S/Hs have been addressed in [6]. The effects of mismatches can be reduced by a different set of adapted coefficients in each equalizer channel have also been achieved in [6]. Better performance in convergence speed and steady-state BER/SER has not been targeted in [6] that has been targeted in [7] in which a new approach to derive the Minimum-Symbol-Error-Rate adaptive equalizers have been introduced in [7]. The constrained optimization problem has been solved with the Lagrange multiplier method which results in an adaptive algorithm with normalization. The proposed equalizer have uniform formulation for BPSK and QAM sources. Moreover, better performance in convergence speed and steady-state BER/SER has been achieved by MSER equalizer in [7]. Better BER and superior performance has not been targeted in [7] that has been targeted in [8]. In this paper,an efficient Time-Domain Adaptive Decision-Directed Channel Equalizer (TD-ADDCE) for Reduced-Guard-Interval Dual-Polarization Coherent Optical Orthogonal Frequency-Division Multiplexing (RGI-DP-CO-OFDM) transmission systems have been proposed in [8]. The phase noise in a decision-directed scheme by extracting and averaging the phase drift of OFDM subcarriers has been estimated by TD-ADDCE. The Channel State Information (CSI) using the decision data and previous estimated CSI in time domain on a symbol-by-symbol basis has updated by it. Without any matrix inversion, TD-ADDCE can perform efficiently. TD-ADDCE has better BER than Frequency Domain ADDCE (FD-ADDCE) has been achieved in [8]. TD-ADDCE can increase the maximum transmission reach by 29% and also superior performance as compared to FD-ADDCE has been targeted in [8]. In [8], the improvement of adaptation accuracy and jitter minimization has not been aimed in [8] that has been aimed in [9].

Meanwhile ,a single-lane, dual –channel, 5-Gb/s serial link re-driver with no Clock Data Recovery (CDR) or Phase-Locked Loop (PLL) has been introduced by Haiqi , et.al [9]. New architecture

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and adaptation algorithm of adaptive equalizer has also been proposed in [9]. A standard 0.13-um CMOS technology has been used in [9]. How well the whole system can be used in different scenario and how the cost of the whole link can be lowered down has been determined by the quality and adaptability of the equalizer in re-driver. Moreover, conflicting effects associated with high and low frequency loop has been resolved. Also the adaptation accuracy has been improved and the jitter minimization has been achieved in [9]. Better speech quality has not been achieved in [9] which has been achieved in [10] in which a decision-directed Adaptive Gain Equalizer (AGE) for assistive hearing instruments have been proposed in [10]. This instrument have been developed using conventional AGE. The conventional AGEs has intended to boost the speech segments of speech signals but are incapable of suppressing noise segment. The limitations of conventional AGE has been overcome in [10]. Moreover, a significant impact has been seen on speech enhancement when signal-to-noise ratio is low. Better speech quality has been achieved in [10] as the proposed approach simultaneously boosts the speech segments and suppress the noise segments in noisy speech. The DDJ and the power consumption problem has not been focused in [10] that has been focused in [11]. In mean time, a 7.5 Gb/s transceiver with a dynamic pre-emphasis calibration and a double pre-emphasis technique have been proposed in [11]. In addition to it, a data width comparison based equalizer with self-adjusting bias control and Band Width (BW) shifting clock generator have also been proposed in [11]. Using the optimum pre-emphasis, the measured jitter of Transmission output data through a channel with 16.88-dB loss has enhanced to 49.9%. under various supply, the noise voltages of the measured jitter of clock generator is reduced by at most 40%. The Data Dependent Jitter (DDJ) and the power consumption problem have been overcome in [11]. Moreover , a Bit Error Rate (BER) under noisy condition has also been achieved in [11]. In [11], the convergence speed and the achievable BER has not been aimed which have been aimed in [12]. In this paper, a new adaptive non-linear equalizer relying on a Radial Basis Function (RBF) have been introduced by Hao , et.al [12]. It is designed based on the Minimum Bit Error Rate (MBER) criterion. This proposal is also known as Online Mixture of Gaussian-Estimator aided MBER (OMG-MBER) equalizer. A mixture of Gaussian based sample by sample probability density function (pdf) estimator is used to model the pdf of the decision variable, for this purpose, a novel online pdf update algorithm has been derived to track the incoming data. With the help of this novel, equalizer parameters has updated sample by sample. The convergence speed and the achievable BER has been targeted in [12]. Moreover, the presented equalizer is known as Least Bit Error Rate equalizer. However, matched filtering, insensitiveness to frequency offset and computational advantages have not been focused in [12] which have been focused in[13] where the application of jointly processing the matched filtering and blind adaptive equalization functions in a receiver for a Nyquist Wavelength Division Multiplexing (Nyquist WDM) system has been introduced in [13]. In this proposal, a Root-Raised-Cosine (RRC) pulse shaping filter have been employed. A blind equalizer with an increased number of taps is able to converge an optimal linear solution combining matched filtering and polarization de-multiplexing have been shown in [13]. A blind equalizer is insensitive to frequency offset has also shown. Moreover, a huge computational advantages by using frequency domain blind equalizer instead of a separate RRC has been targeted in [13]. In [13], Gbps serial link and jitter minimization has not been focused that have been approached in [14] in which a minimum jitter based adaptive Decision Feedback Equalizer (DFE) has been

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proposed in [14]. The adaptation for the optimal tap coefficients of DFE has been also searched in [14]. In order to achieve jitter minimization, the slope of the DFE that counteracts that of the channel has adjusted. The Giga-Bit-Per-Second (Gbps) serial link has been achieved in [14]. Moreover, to minimize data jitter at the edge of data eyes have been targeted in [14].Crosstalk compensation with relatively low computational complexity has not been targeted in [14] that have been targeted in [15]. In this paper, two sparse Frequency Domain Equalizers (FDEs) with low complexity for degenerated Mode-Group Division Multiplexing (MGDM) systems have been proposed by Kai , et.al [15]. The sparse channel impulse response is caused by the strong crosstalk at the mode Mux/Demux and weak coupling in the fiber between different mode groups. In this paper, the first method is based on channel impulse response, in which a mask of taps with significant magnitudes has generated. The second method based on Improved Proportionate Normalized Least-Mean Square (IPNLMS), in which active and inactive taps has adjusted at different rates of convergence. Moreover, crosstalk compensation with relatively low computational complexity has been targeted in [15] where a blind receiver compensation and estimation for long-haul non-dispersion managed system using adaptive equalizer has been proposed in [16]. Imbalances between the four-sampling channels of coherent detector with polarization diversity can rapidly degrade system performance has been observed in [16]. Receiver Skew is an important imbalances which has been used at the time of higher symbol rates, higher modulation orders and low roll-off pulse shapes. Moreover, the performance of receiver skew compensation is based on complex-valued Multiple-Input Multiple-Output (MIMO) 4*2 adaptive equalizer which is tolerant to large residual chromatic dispersion have been achieved in [16]. Also skew estimation from converged equalizer taps has been derived in [16]. In [16] better steady state performance and improved temporal tracking speed has not been aimed that has been aimed in [17].

Meanwhile, a design and implementation of Frequency Domain (FD) Block Least Mean Square (BLMS) and Block Recursive Least Square (BRLS) equalizers for adaptive Zero-Guard-Inherent (ZGI) Coherent Optical (CO) Orthogonal Frequency Division Multiplexing (OFDM) has been proposed in [17]. The proposed BLMS and BRLS equalizers are more effective in rescuing the system from severe physical changes and in tracking continuous variation without the need for additional TS than the ZF equalizer. In the single channel 112 Gb/s QPSK and 250 Gb/s 16-QAM Polarization-Division Multiplexed (PDM), the proposed equalizers increase the transmission distance by 20% and 7% for QPSK, 53% and 35% for 16-QAM as compared with TF based and decision aided ZF equalizers. High non-linear tolerance of the ZF equalizer has been preserved and better steady state performance has also been achieved in [17]. Moreover, improved temporal tracking speed has also been targeted in [17]. A compensation for mis-equalization has not been targeted in [17] that has been targeted in [18] where an Enhanced Active Noise Equalizer (EANE) algorithm to compensate for mis-equalization problem has been introduced in [18]. ANE is one of the popular methods of residual noise reshapping. The ANE application has been restricted by mis-equalization problem, i.e the sensitivity to secondary path modeling errors. Multiple-harmonic EANE has also been studied in [18]. The EANE can adaptively tune the gain factors of ANE for an accurate control of amplitudes of residual noise at reference frequencies. Moreover, the problem of mis-equalization has been overcome in [18]. Accurate performance of EANE at reference frequencies has also

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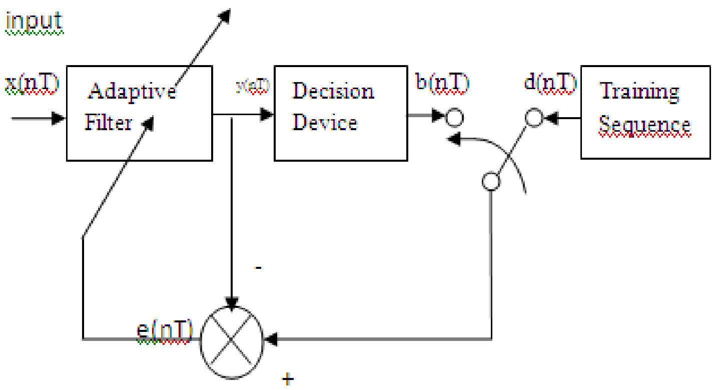
achieved in [18]. To compensate the variation in frequency response and attenuation variation has not been aimed in [18] that has been aimed in [19]. In this paper, an Adaptive continuous-time equalizer integrated in a receiver front-end for short-reach high-speed optical communications through low-cost Step-Index Plastic Optical Fibers (SI-POF) has been presented in [19]. Compensation of the variations in the bandwidth of the filter upto 50 m and attenuation variation of 12dB has been achieved in [19]. Moreover, to work at different data rates ranging from 400 Mb/s to 2.5 Gb/s has also been targeted in [19]. Error in data reception has also been avoided. The fast-converging method has not been targeted in [19] that has been targted in [20].

In mean time, a 20Gb/s adaptive linear equalizer with the coefficients fast-converging method has been presented in [20]. In this work, the adaptation procedure is rearranged to reduce the asynchronous sampling time. Before the low frequency gain has calibrated, the high frequency gain of the linear equalizer is well compensated. However, a considerable amount of time has been required by the equalization coefficients to be determined. The power consumption of the proposed method based on the asynchronous sampling technique can remain within an acceptable range while the data rate goes higher than 10Gb/s. A fast-converging method to adjust the linear equalizer has been achieved in [20]. Moreover, the equalization coefficients have to be determined within 2.68us has also been targeted in [20]. In [20], global convergence and high bandwidth system with fast mobility has not been targeted which has been targeted in [21] where a compressed training adaptive equalization has been proposed in [21]. It is a semi-blind approach for both Single-Carrier Time-Domain Equalization (SC-TDE) and Single-Domain Frequency Equalization (SC-FDE) based communication systems. In this paper, the magnitude boundedness property of digital communication sources have been utilized. Moreover, through convex optimization, low complexity implementations with global convergence have been facilitated in [21]. A link between compressed sensing and adaptive equalization have also been established. The quantity of training symbols in communication packet has also been reduced. Moreover, in low noise scenario the required training length is in the order of the logarithm of the channel spread have been evaluated and high bandwidth systems with fast mobility has also been achieved in [21]. Improved eye-height opening, bandwidth extention and application in speed serial receivers has not been achieved in [21] that has been achieved in [22] where an improved analog equalizer using four level Pulse Amplitude Modulation (PAM-4) signaling have been proposed in [22]. In this paper, boosting-state detection in time domain has performed. To suppress the Pattern-Dependent Jitter (PDJs), an inductorless, cross-stage feedback structure has been employed which broaden the effective tuning bandwidth. Moreover, 42% improved in eye-height opening has been achieved in [22]. Also the bandwidth has extended by 60%, thus the PDJs has came down from 44% to 27.5%. application in high speed serial receivers has also been targeted in [22]. The convergence speed and BER performances have not been targeted in [22] which have been targeted in [23]. In this paper, a 5 Gbps 60 GHz radio-over-fiber (RoF) system using an adaptive Activated Artificial Neural Network Nonlinear Equalizer (ANN-NLE) has been introduced in [23]. In this paper, a variable- alpha activation function ANN-NLE with a faster convergence speed and a stronger equalization ability have been applied to solve the non-linear problems of RoF link. Comparisons between AN-NLE, Volterra and LMS equalizer has also been presented in [23]. Using the proposed ANN-NLE, a higher convergence speed and better BER performances have been achieved in [23]. The 60 GHz RoF system can successfully transmit a 5Gbps BPSK signal over the 10-Km SMF and 1.2-m wireless link under FEC limit of 10^{-3} have been achieved in [23]. The loss budget and tolerance to DGD has not been aimed in [23] that has been aimed in [24] in which a 100-Gb/s/wavelength- based coherent Wavelength Division Multiplexing (WDM) Passive Optical Network (PON) system with a hardware-efficient Digital Signal Processing (DSP) has been proposed in [24]. A new Adaptive Equalization (AEQ) and Carrier Phase Recovery (CPR) has been featured. Here, the conventional AEQ is splitted into 1-tap butterfly Finite Impulse Response (FIR) filter and two non-butterfly FIR filters to halve the number of long FIR filters. This has been done to sacrifice the compensation of Different Group Delay (DGD). Also, a phase offset estimation has been introduced between the two polarizations in the CPR, in order to reduce the number of multiplications by 41%. Moreover, a loss budget of 40.7 dB capable of supporting 8ONU splits over an 80-Km Single-Mode Fiber (SMF) span has also been achieved in [24]. Also, a tolerance to DGD has also been evaluated in [24]. Appropriate compensation of channel losses, low cost, low power consumption, high speed serial link application has not been targeted in [24] that has been targeted in [25]. In this paper, a power saving adaptive equalizer with digital-controlled self-slope detection has been proposed by Yo-Hao, et.al [25]. A serial processing for reducing high-speed detection circuit has also been proposed in [25]. A self-slope detection circuit compares two continuous serial slope instead of parallel slopes. Moreover, through digitization, the self slope detection technique solved the imbalanced swing problems and mismatch. Also, the registers has been used to replace one high speed detection circuit and slicer. Appropriate compensation of channels losses has been achieved in [25]. Moreover, low cost and low power consumption for consumer electronic products has also been targeted in [25]. A high-speed serial link applications has also been focused in [25].

SYSTEM MODEL

Equalizer is used to reduce Inter Symbol Interference (ISI) and additive noise which are added as and when time progress. Equalizer has been used in the receiver section. For linear time varying channel, time varying equalizer is used. So Adaptive Equalizer has been used as it varies with time. The system model of adaptive equalizer has been shown below:



Equalization is the process of correcting channel induced distortion. This process is said to be Adaptive when the filter itself adjusts continuously during the data transmission by operation on the input signal. In Adaptive Equalizer, Finite Impulse Response (FIR) filters has been used. The FIR filter coefficient has updated adaptively step by step fashion

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synchronously with the incoming data. During, the training period, a known sequence is transmitted and a synchronized version of the input signal is generated in the receiver. This generated signal has been applied to the adaptive equalizer as the desired response. The length of this training sequence must be equal to or greater than that of the adaptive equalizer. When the training mode is completed, adaptive equalizer switched to the second mode, the decision- directed mode. In this mode, the error signal equals to

$$e(nT)=b(nT)-y(nT) \quad (1)$$

where y(nT) is the equalizer output. b(nT) is the final correct estimate of the transmitted symbol b(nT). The decision made by the receiver are correct with high probability. Thus, the error estimated are correct most of the time which permit the equalizer to operated satisfactorily. An adaptive equalizer operating in a decision-directed mode is able to track relatively slow variations in channel characteristics.

PROPOSITION

In a message transporting arrangement, signal has been spread through a raucous communication channel and distorted signal is generated due to ISI. To overcome this restraint the most appropriate steadfastness is exploitation of Adaptive Equalizers (AEs) in an efficient manner by tuning the parameters of the AEs without prior knowledge. Existing algorithms provides the better results. However, for more sophisticated communication system demands less error and more accuracy. To handle this bio-inspired algorithm based AE can be designed. In this method filter co-efficient can be tuned adaptively using nature inspired algorithm like Genetic Algorithm (GA) or Particle Swarm Optimization Algorithm (PSO). As a result more accurate result can be achieved.

CONCLUSION

Adaptive equalizer has assumed the channel to be time-varying and tried to design an equalizer filter whose coefficients varied in time according to the change of channel. Inter symbol interference (isi) and additive noise has been estimated each time. It has also assumed that the channel is varying slowly. The most important application is the adaption of the filter parameters has based on minimizing the mean squared error between the final output and the desired signal. Adaptive equalizer has been also used for the following purposes which are noise cancellation, signal prediction, adaptive feedback cancellation, echo cancellation. Among these noise cancellation, signal prediction can easily be accomplished by adopting the above method which not only provide accurate results but also make the design of digital communication systems more dynamic.

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