Grouped Signed Power-of-Two Algorithms for Low-Complexity Adaptive Equalization

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Abstract—With increasing demand for higher data rate, modern communication systems have grown more complex. Equalization has become more and more important as it is effective in mitigating the multipath fading often occurred in high-data-rate communication systems. However, the implementation complexity of adaptive equalizers is usually too high for mobile communication applications. In this paper, a novel adaptive equalization algorithm and its low-complexity architecture are proposed. This algorithm employs a new grouped signed power-of-two (GSPT) number representation. The GSPT algorithm and several enhanced versions are simulated as adaptive equalizers in a phase-shift keying communication receiver for several practical channels and the GSPT-based equalizers perform as well as the least mean square (LMS) equalizer. Moreover, for comparison, two GSPT-based equalizers and two other equalizers are implemented in field-programmable gate arrays. The GSPT-based equalizers require only about 25%–30% of the hardware resources needed in the LMS equalizer. Also the GSPT-based equalizers are more than twice as fast as the LMS equalizer.

Index Terms—Adaptive algorithms, adaptive filters, low circuit complexity, power-of-two (PT).

I. INTRODUCTION

WITH THE introduction of modern communication standards and the advances in VLSI technology, wireless communication services, such as cellular phones and wireless local area networks, are becoming popular and are growing quite rapidly. As the transmission rate and the modulation level are increased, intersymbol interference (ISI) caused by the multi-path fading channel has become a serious problem [1]. In modern communication systems, equalization has become crucial in tackling the ISI caused by wireless channels. In addition, adaptive equalization is often necessary in mobile communication systems since the channel is time varying.

An adaptive equalizer is in essence a kind of linear adaptive filter that models the inverse transfer function of the channel. The least mean square (LMS) algorithm and the recursive least square (RLS) algorithm are two well-known adaptive algorithms [1]. Although the RLS algorithm has faster convergence speed than the LMS algorithm, its complexity for hardware implementation is too high to be practical. On the other hand, the LMS algorithm is widely used because of its simplicity and

robustness. For many portable devices that operate on batteries, power consumption is an essential issue and simpler equalizers are advantageous since they consume less power.

An LMS equalizer can be divided into two portions: The linear filter and the coefficient updating block. Multiplication operations are necessary in both parts and they make up most of the computation complexity in LMS equalizers. Conventionally, the canonical signed digit (CSD) representation or signed power-of-two representation for filter coefficients reduce the multiplier complexity. With such representations, multiplications required for the filtering process are replaced by simple shift-and-add operations, thus reducing the filter complexity [2]–[4]. Furthermore, several other adaptive algorithms, such as sign–sign LMS, sign–error LMS, sign–data LMS, and log–log LMS [5], have been proposed to further reduce the complexity of the updating block. Actually, several multiplier-less adaptive equalizers have been implemented using these techniques [6], [7]. However, conversion between different number systems is required for coefficients updating in these multiplier-less adaptive equalizers thus introducing additional delay in critical path and additional latency in obtaining the equalizer outputs.

In this paper, we will introduce several algorithms and corresponding architectures for low-complexity adaptive equalizer implementation. Previously, we have proposed a new number representation system [8], which we call grouped signed power-of-two (GSPT) number system. The GSPT number system divides each coefficient into several groups of consecutive signed digits and allows at most one nonzero digit in each group. As a result, the complexity of the linear filter is further reduced since fewer digits need processing in data-coefficient multiplication. Moreover, two LMS-based adaptation algorithms are also introduced [8]. In this paper, we will propose two more adaptive algorithms for the GSPT-based adaptive filter. They operate like the sign–sign LMS algorithm and log–log LMS algorithm, yet with a variable step size and a simple updating mechanism. Coefficients are updated on the GSPT numbers directly and no conversion is necessary. Furthermore, these algorithms are shown to be more suitable for equalization in the Doppler-fading channel environments.

The paper is organized as follows. In Section II, we present the GSPT number system and two GSPT-based equalization algorithms. In Section III, two new GSPT-based equalization algorithms are proposed. In Section IV, we present the simulation results of GSPT-based adaptive equalizers in a quadrature phase-shift keying (QPSK) communication system. Section V compares the implementation complexity of the proposed architectures and two other equalizers. Finally, Section VI concludes this paper.

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II. GSPT LMS ALGORITHMS

The LMS algorithm is widely used in adaptive systems because of its simplicity and robustness. The LMS algorithm for an N -tap adaptive equalizer is described by

$$y[n] = \sum_{k=0}^{N-1} w_k[n] \cdot x[n-k] \quad (1)$$

$$e[n] = d[n] - y[n] \quad (2)$$

$$w_k[n+1] = w_k[n] + \mu \cdot e[n] \cdot x[n-k], \quad 0 \leq k < N \quad (3)$$

where n is the time index, and $w_k[n]$, $d[n]$, $e[n]$, $x[n]$, and $y[n]$ represent the k th filter coefficient, the desired signal, the error signal, the input signal, and the output signal, respectively. In general, we can simplify the LMS adaptive equalizer using the sign-error (sign-data) LMS algorithm by replacing $e[n](x[n-k])$ with simply the sign of $e[n](x[n-k])$. Actually, the sign-sign LMS algorithm can be used to further reduce the implementation complexity using only the sign of $e[n]$ and that of $x[n-k]$ in updating [9]. This suggests that, to guarantee convergence, the direction of the update is more important than the magnitude of the update if the step size is kept small.

In the following, we present the GSPT number system [8] and two associated adaptive equalizer algorithms [8] based on the sign-sign LMS idea. These algorithms can greatly reduce the complexity of both the linear filter and the coefficient updating block of an adaptive filter.

A. GSPT Number System

This number system is called the GSPT number system. The idea is to represent a number by several groups of digits and in each group at most one nonzero signed digit (1 or -1) is allowed. Suppose that a 12-bit GSPT number is partitioned into three groups. Then, a GSPT number can be represented as

$$449 = 512 - 64 + 1 = \underline{0010} \underline{0\bar{1}00} \underline{0001} \quad (4)$$

where each group is marked by an underline and the signed digit ' -1 ' is represented by $\bar{1}$.

B. GSPT LMS Algorithm

As mentioned before, the updating direction is more important than the magnitude of the update. Therefore, only the direction of the term, $e[n] \cdot x[n-k]$, is used for the coefficient updating in the GSPT LMS algorithm. The GSPT LMS algorithm proposed in [8] for an N -tap adaptive equalizer is described by (1)(2)(5). (5) describes the updating equation of the GSPT LMS algorithm as follows:

$$w_k[n+1] = \begin{cases} w_k[n] \uparrow, & \text{if } e[n] \cdot x[n-k] > 0, \\ w_k[n], & \text{if } e[n] \cdot x[n-k] = 0, \\ w_k[n] \downarrow, & \text{if } e[n] \cdot x[n-k] < 0, \end{cases} \quad 0 \leq k < N. \quad (5)$$

When $e[n] \cdot x[n-k]$ is positive (zero or negative), we increase (freeze or decrease) the linear filter coefficient $w_k[n]$. The updating unit having four digits is shown in Fig. 1(a) and an illustration of the updating operation is given in Fig. 1(b).

When a positive trigger (increase) signal, *carryin*, is received, $b_3b_2b_1b_0$ of this updating unit 'shifts up' to increase

Fig. 1. (a) GSPT updating unit. (b) Illustration of the operation in a GSPT updating unit.

Fig. 2. Diagram of the GSPT coefficient updating block.

its value. Likewise, when a negative trigger (decrease) signal, *borrowin*, is received, $b_3b_2b_1b_0$ of this updating unit 'shifts down.' If $b_3b_2b_1b_0$ has been already at the upper (lower) limit 1000 ($\bar{1}000$) and a positive (negative) trigger signal is received, the output signal *carryout* (*borrowout*) is sent to the next more significant unit and $b_3b_2b_1b_0$ is reset to 0000. Fig. 2 shows the coefficient updating block implemented by cascading four updating units in a 12-bit coefficient example.

C. GSPT-A LMS Algorithm: Improved Coefficient Accuracy

In adaptive algorithms, decreasing the step size usually helps improve the steady-state error performance. Even though the step size is not fixed but variable in the GSPT LMS algorithm, we can decrease the step size effectively by increasing the wordlength of the filter coefficients. To improve the steady-state error performance without increasing the complexity of the linear filter, it is well known that we can increase the wordlength used in updating but retain the wordlength for the linear filter. The concept is shown in Fig. 3 and it is called the GSPT-A LMS algorithm [8]. However, its convergence speed is slower than the GSPT LMS algorithm due to the relatively smaller step size.

III. NEW GSPT-BASED ALGORITHMS

The GSPT-based LMS algorithms proposed in [8] can only be used for static or quasi-static channels. So, we propose two new GSPT-based adaptive algorithms. One can be regarded as a variation of GSPT-A LMS while the other is proposed to improve the convergence speed for dynamic channels.

A. GSPT-AV LMS Algorithm: Improved Coefficient Accuracy With Variable Group Size

The algorithms presented previously all have the same number of digits in each group for regularity. However, trun-

Fig. 3. Coefficient updating block of the GSPT-A LMS algorithm.

ating one digit at the most significant group degrades the coefficient accuracy more severely than truncation in the less significant groups. It is advantageous in terms of coefficient accuracy to use fewer digits in the more significant groups and more digits in the less significant groups. With this modification, the set of representable numbers will distribute more uniformly on the real number line. The architecture of such a variable-group-size GSPT-based coefficient updater is almost the same as the GSPT-A coefficient updater shown in Fig. 3 but we can have updating units corresponding to different group sizes. For an example, we may have four groups with group size 2, 3, 3, 4, respectively. We call this method the GSPT-AV LMS algorithm.

B. GSPT-AS LMS Algorithm: Improved Coefficient Accuracy and Convergence Speed

In mobile communication applications, the convergence speed of an adaptive system in the receiver is crucial. To improve the convergence speed while at the same time keeping the complexity at a reasonable level, the idea of the log-log LMS algorithm [5] is adopted. The updating equation of the log-log LMS algorithm is given by

$$w_k[n+1] = w_k[n] + \mu \cdot Q(e[n]) \cdot Q(x[n-k]), \quad 0 \leq k < N$$

where $Q(z)$ selects the PT value that is closest to z . Applying this idea to the GSPT number system, the updating term is first calculated in the same way as the log-log LMS algorithm. Based on the magnitude of the updating term, we can inject a carry/borrow signal to the corresponding updating unit. For example, if a 12-bit integer coefficient is divided into four groups, numbered from 1 to 4 (from MSB to LSB). We increase/decrease the coefficient by sending a trigger to the fourth group if the magnitude of the updating term is smaller than or equal to 2. Similarly, the coefficient is increased/decreased at the 3rd, 2nd, and the 1st groups if the magnitude of the updating term is between 4 and 16, between 32 and 128, and larger than 256, respectively. We call this algorithm GSPT-AS LMS algorithm and it is illustrated in Fig. 4. Note that the updating term can be computed without multiplication since the step size is a power of two.

C. Theoretic Analysis on Coefficient Representation Errors

Let $E(u, v)$, $u < v$, denote the sum of squared errors caused by representing coefficients in $[u, v]$ with the GSPT number system. Considering the GSPT-A scheme with $g = 3$ and infinite number of groups, we can divide the set of nonnegative integers into subsets, $[0, 2^0]$, $[2^0, 2^1]$, $[2^1, 2^2]$, $[2^2, 2^3]$, ..., and then calculate $E(u, v)$ for these subsets one by one.

Fig. 4. Coefficient updating block of the GSPT-AS LMS algorithm.

It is obvious that $E(0, 2^2) = 1$ since $1 = 2^0$, $2 = 2^1$, $4 = 2^2$, but 3 can not be represented as a GSPT number exactly. Similarly, $E(0, 2^3) = 2$ since both 3 and 5 have an error of 1 when represented by the respective nearest GSPT numbers. Next, $E(2^3, 2^4)$ is equivalent to $E(2^3, 2^3 + 2^2) + E(2^3 + 2^2, 2^4)$ and note that $E(2^3, 2^3 + 2^2) = E(0, 2^2) = 1$ because 2^3 is represented by a CSD in the second group. In addition

$$E(2^3 + 2^2, 2^4) = E(2^4 - 2^2, 2^4) = E(-2^2, 0) = E(0, 2^2) = 1.$$

So, $E(2^3, 2^4) = 2 \cdot E(0, 2^2) = 2$.

Carrying on the above procedure, we have derived the following analytical formula for $E(u, v)$ of three specific forms:

$$E(2^{3p}, 2^{3p+1}) = 2 \cdot E(0, 2^{3p-1})$$

$$E(2^{3p+1}, 2^{3p+2}) = 2 \cdot E\left(0, \sum_{i=1}^p 2^{3p-1}\right) + R_p^2 + \sum_{i=1}^{R_p-1} i^2$$

and

$$E\left(2^{3p+2}, 2^{3(p+1)}\right) = 2 \cdot E\left(0, \sum_{i=1}^p 2^{3p-1}\right) + E(0, 2^{3p+1}) + R_p^2 + \sum_{i=1}^{R_p-1} i^2$$

where $R_1 = 2^2$, $R_2 = 2^5 - 2^2$, $R_3 = 2^8 - 2^5 - 2^2$, and so on.

Considering the GSPT-A scheme with $g = 3$ and five groups (four groups for filtering), we can derive $E(0, 2^{11}) = 7188518$. The maximum coefficient representation error is actually $R_3 = 2^8 - 2^5 - 2^2 = 220$. Similarly, for GSPT-AV with group size 2, 2, 3, 4, 4, we conclude that $E(0, 2^{11}) = 48768$ and the maximum coefficient representation error is only 16. In summary, GSPT-AV scheme provides much better coefficient accuracy than the GSPT-A scheme.

IV. SIMULATION RESULTS

To verify the feasibility of the proposed adaptive algorithms in practical applications, an adaptive GSPT-based equalizer for a QPSK communication system over multipath fading channel is constructed for simulation. At the transmitter end, the bit stream is randomly generated and sent to a QPSK modulator. The modulated signal is then upsampled with over-sampling of four and the result is passed to a pulse shaping filter to generate the transmitter output. The transmitter output is then distorted by the multipath fading channel and additive white Gaussian noise (AWGN). At the receiver end, the received signal is first passed through a pulse shaping filter, downsampled, and sent

Fig. 5. Bit error rate versus E_s/N_0 on TU channel: GSPT LMS algorithm and GSPT-A LMS algorithm with different group sizes.

to a T/4-spaced 41-tap adaptive equalizer. Finally, a decision block is used to decide the output symbol for calculating the bit error rate. Static and dynamic multipath fading channel models, including the the global system for mobile communications (GSM) channel model [10] without the Doppler effect and the GSM channel model with the Doppler effect, are used in the simulation.

A. Simulation Results on Static Channels

For simulations conducted on static channels, we use the GSM channel model without the Doppler effect. All GSPT-based algorithms are simulated on the GSM channel to compare their performance. In each run, 40 000 symbols are used and overall 25 runs are simulated in each case.

First, the GSPT LMS algorithm and GSPT-A LMS algorithm are used in the adaptive equalizer to determine the proper group size. We set the group size g to 3 and 4 and the wordlength w adopted for the filter part is set to 12. The simulation results on the typical urban (TU) channel are illustrated in Fig. 5. It is not surprising that the GSPT-A LMS algorithm has better performance than the GSPT LMS algorithm regardless the group size. Furthermore, it is clear that for GSPT-A LMS algorithm, a group size of 4 seems to be sufficient for efficient equalization. Finally, note that the bit error rates of all algorithms are almost the same under low E_s/N_0 , while they are quite different in high E_s/N_0 scenarios. This is because in the high E_s/N_0 scenarios, the coefficient accuracy determines the error performance and thus the most accurate method, GSPT-A LMS with $g = 3$, outperforms all the others.

Next, we compare the performances of GSPT, GSPT-A, GSPT-AV, GSPT-AS, and the conventional LMS algorithms. The GSPT-AV algorithm has five groups with group size 2, 2, 3, 4, 4 for updating and it uses the most significant 11 bits for the linear filter. The other algorithms all use the most significant 12 bits of the updating coefficients in the linear filter. Simulated bit error rate performances shown in Fig. 6 indicate that the GSPT-AS algorithm has best performance among all GSPT algorithms and is quite close to the LMS algorithm. In addition, the GSPT-A and GSPT-AV algorithms have comparable performance and they are both better than the original GSPT algorithm.

From Fig. 6, we can see that GSPT-AV does not perform much better than GSPT-A even though the previous analysis indicated otherwise. The reason is that, compared to GSPT-AV, GSPT-A has a number distribution that is more concentrated on small values. Fortunately for GSPT-A, the adaptive equalizers in the simulation have many taps with small coefficients, thus the improved performance of GSPT-A equalizer.

Fig. 6. Bit error rate versus E_s/N_0 on TU channel: comparison among GSPT, GSPT-A, GSPT-AV, GSPT-AS, and LMS.

Fig. 7. Bit error rate versus E_s/N_0 on TU channel with 50 Hz Doppler frequency: comparison between GSPT-AS and LMS.

Even though the GSPT-AS algorithm has the best error performance, it also has the highest complexity. Considering both the performance and required complexity, we conclude that the GSPT-A algorithm is the most suitable solution for equalizers in static channel environments.

B. Simulation Results on Dynamic Channels

Simulation of the dynamic channel scenario uses the GSM (900 MHz) channel model with the Doppler effect on a QPSK system with 270.8-kHz symbol rate. The Doppler frequency is set to 50 Hz, which corresponds to a relative velocity of 60 km/hr. Under such condition, the dynamic range of the signal through the channel becomes too large to be equalized even by an LMS-based adaptive equalizer. An automatic gain controller (AGC) is introduced to constrain the dynamic range of the received signal. Since the channel varies so fast, adaptation speed becomes the most important issue in the equalizer design. Therefore, only the GSPT-AS LMS algorithm and the LMS algorithm are simulated.

The step size of the LMS algorithm is set to 2^{-7} to track the channel variations continuously. On the other hand, for the GSPT-AS LMS algorithm, the step size is set to 2^{-10} and the wordlength is set to 12 and 15 in the linear filter and the coefficient updating block, respectively. The group size g is set to 3 to achieve better performance in the dynamic channels. The simulation results on TU channel are illustrated in Fig. 7.

V. IMPLEMENTATION AND COMPLEXITY COMPARISON

To make a more realistic comparison of hardware complexity, we implement four adaptive equalizers using SRAM-based field-programmable gate arrays (FPGAs). They are the equalizer based on the LMS algorithm, the proposed GSPT-A LMS algorithm, the proposed GSPT-AS LMS algorithm,

Fig. 8. Example of a 4-tap LMS equalizer.

TABLE I
IMPLEMENTATION COMPLEXITY COMPARISON AMONG LMS, GSPT-A,
GSPT-AS, AND RNS

	LMS		GSPT-A		GSPT-AS		RNS		
	LE	%	LE	%	LE	%	LE	%	ROM (bits)
filter	5586	100	2182	39	2182	39	1890	34	388k
update	6139	100	789	13	1369	22	3276	53	400k
total	11725	100	2971	25	3551	30	5166	44	788k

and the modified LMS algorithm proposed in [11] using residue number system (referred to as the RNS scheme). All equalizers have 11 taps and are implemented based on the architecture shown in Fig. 8. Parameters of the GSPT-A LMS algorithm and GSPT-AS LMS algorithm are as follows: $g = 4$, updating wordlength = 16. The linear filter coefficient wordlength is 12 in all four equalizers and the internal wordlength is 24 for the accumulating path.

How the multiply-and-accumulate (MAC) operations needed in linear filtering and coefficient updating are implemented differs from equalizer to equalizer. In the LMS equalizer, the optimized MAC module provided by the FPGA vendor is adopted. For the GSPT-based equalizers, the updating units are designed with basic combinational gates and are then cascaded to form the coefficient updaters as shown in Figs. 3 and 4. In the linear filtering part of the GSPT-based equalizers, three partial products are generated in each tap. These partial products are then summed together using vendor-provided, pre-designed adders. For the RNS equalizer [11], specially-designed multipliers and adders for the residue number system are designed for both the linear filtering and the coefficient updating blocks.

Table I lists the number of logic elements (LE) needed in an FPGA in implementing these four equalizers. The RNS scheme has longer critical path delay than the GSPT-based schemes due to the necessary number representation conversion required in the feedback loop. Also, it requires higher complexity since the complexity of modulo adders is higher than that of binary adders. Clearly, the GSPT-A LMS algorithm has the lowest complexity and it uses about only 25% of the hardware resource required by the conventional LMS algorithm. For the GSPT-AS equalizer, it requires only about 30% of the hardware resource needed in the LMS equalizer. Moreover, the speed performances

TABLE II
SPEED PERFORMANCE COMPARISON AMONG LMS, GSPT-A,
GSPT-AS, AND RNS

	LMS	GSPT-A	GSPT-AS	RNS
critical delay	144 ns	64 ns	64 ns	79 ns

of the four equalizers are listed in Table II. Both GSPT-based equalizers can operate at more than twice as the speed of the LMS equalizer.

VI. CONCLUSION

In this paper, new low-complexity equalizer algorithms based on a new number system (GSPT) are proposed to improve the adaptation speed of the proposed equalizer. Simulation results of the GSPT-based algorithms indicate that the GSPT-A algorithm performs best in static channels and it has the lowest complexity. On the other hand, the GSPT-AS algorithm is suitable in both static and dynamic channels with a little more complexity than the GSPT-A algorithm. The GSPT-AS algorithm performs as well as the traditional LMS equalizer in both types of channels. FPGA hardware implementations of four equalizers show that the GSPT-A algorithm has by far the lowest hardware complexity, about 25% of that in the LMS equalizer. It also shows that the GSPT-based equalizers are more than twice as fast as the LMS equalizer. With low hardware complexity, high speed, and comparable equalization performance, the GSPT-based equalizers are thus very useful in low-power receivers for broad-band and mobile communications.

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