A Symbol-Rate Timing Recovery Method for Low-Energy High-Speed Wireless Communication System

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***Abstract*— This paper presents a new symbol-rate timing recovery approach based on adaptive interpolation to reduce the sampling frequency of analog-to-digital converter (ADC) and power consumption for low-energy high-speed wireless communication system, such as ultra-wideband (UWB) system. Using a principle of maximum absolute-squared sum (MASS), the ADC can select the optimal clock phase provided by a multiple clock generator to sample the incoming signals at the symbol rate. Adaptive interpolation is then introduced to compensate for the performance lose caused by residual timing error due to insufficient amount of clock phases. Both the least mean square (LMS) algorithm and recursive least square (RLS) algorithm can be applied on the adaptive interpolation filter. Simulation results show that adaptive interpolation can improve the performance by decreasing the residual timing error, and the RLS interpolator outperforms the LMS one at the cost of increased complexity.**

I. INTRODUCTION

Timing recovery plays an important role in ensuring precise signal decoding performance as it affects the sampling phase and frequency of analog-to-digital convertor (ADC) for incoming signals. The ADC sampling frequency reflects a trade-off between system performance and power consumption, which has become a crucial issue in the design of wireless communication systems. An approach proposed in [1-3] applies a fixed multirate sampling clock (at Nyquist rate or a higher rate) for ADC to sample the incoming signals, and then use them to reconstruct the symbol-rate signals with a interpolation algorithm. Another approach proposed for symbol timing synchronization [4,5] employs a polyphase filterbank to perform the interpolation. It eliminates the need of a separate interpolation filter and incorporates maximum likelihood timing techniques. However, both methods above require a sampling rate greater than the Nyquist rate, which will increase the difficulty for circuit design and result in high power consumption for high speed applications.

To decrease the power consumption and use symbol-rate sampling approach in high-speed communication system, a dynamic sample-timing controller (DSTC) and phase-tunable clock generator (PTCG) are introduced in wireless orthogonal frequency division multiplexing (OFDM) systems [6]. The DSTC searches for the optimal sampling phase at the symbol rate provided by the PTCG to reduce extra power consumption. But the performance relies on the number of clock phases, insufficient number of clocks will lead to significant performance degradation. To reduce the number of clocks, an interpolation method based on baud-rate sampling is proposed in [7] by exploiting the structure of pulse amplitude modulation (PAM) signals in Gigabit Ethernet communication, and the detailed implementation is given in [8].

This work proposes a timing recovery method by employing a set of multiphase clocks to reduce the sampling frequency to symbol rate with a method of maximum absolute-squared sum (MASS) to control the sampling. Then adaptive interpolation is introduced to eliminate the residual timing error caused by insufficient amount of clock phases. Both least mean square (LMS) and recursive least square (RLS) adaptive algorithms [9] are applied on the updating of the coefficients of the adaptive interpolation filter.

The rest of the paper is organized as follows: Section II demonstrates the system architecture and detailed implementation of the proposed approach. MASS sampling control was introduced in Section III. Section IV explores the application of LMS and RLS algorithms on the adaptive interpolation filter. Simulation results and conclusions are shown in Section V and Section VI, respectively.

II. SYSTEM MODEL

A. *System Overview*

This approach adopts a digital multiple clock generator to provide a series of clocks running at the symbol rate.

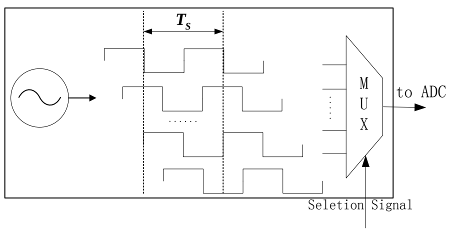

Fig. 1. Multi-clock generator.

As illustrated in Fig.1, these clocks have the same phase space and are treated as clock candidates by a multiplexer to drive the ADC. With the sampling clock selection signal

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provided by the MASS method, the ADC can sample the incoming signals with a selected optimal clock, which is close to the ideal sample instant. The proposed timing recovery system is shown in Fig. 2, including a timing loop [2-3] and an adaptive interpolation block.

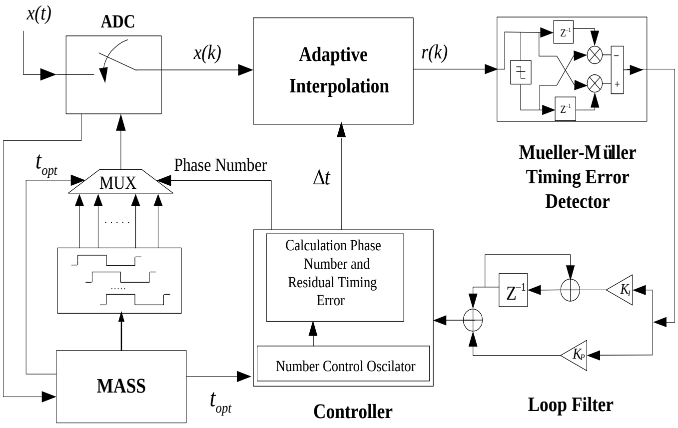

Fig. 2. System architecture of the proposed timing recovery method.

The incoming analog signal $x(t)$ is sampled at the kth symbol corresponding to value $x(k)$ by the ADC with an optimal clock phase selected by MASS. Due to the insufficient number of clock phases of the multiple clock generator or clock jittering, there is usually a residual timing error (as illustrated in Fig. 3) between the sampling instant and the ideal sampling instant, causing a difference between the sampled value and desired value. Hence an adaptive interpolator is employed to implement a process of interpolation with the sampled symbol-rate values to eliminate the difference. The rest of the system loop contains a timing error detector, a loop filter, and a controller.

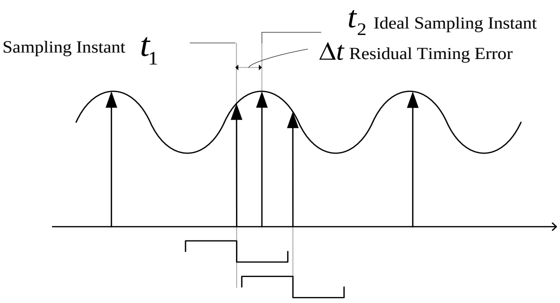

Fig. 3. Residual timing error.

B. System Implementation

Since this system works at a symbol rate, the Mueller-Müller timing error detection algorithm [10] is employed. The timing error information can be calculated by

$$t_{error} = Re[r(k-1)\hat{r}^*(k) - \hat{r}^*(k-1)r(k)] \qquad (1)$$

where $r(k)$ is the kth sampled signal and $\hat{r}(k)$ is its decision value, $*$ denotes the conjugate operation and Re denotes the real part. The implementation structure is shown in Fig. 2. For QPSK modulation, two modules are employed for the in-phase and quadrature components in our simulation.

The loop filter as showed in Fig. 2 is a classic proportional integrator, containing a proportion path and an integration path.

The controller is a modification one to that of used in [2]. Both the phase selection information and residual timing error are derived from the fractional interval μ_k by quantizing μ_k with the number of clock phases.

III. MASS SAMPLING CONTROL

Since we want the ADC to sample at a symbol rate and reconstruct the desired signal, the sampling phase should be optimal after the phase adjusting stage. The MASS sampling control method selects the phase closest to the ideal sampling instant.

Assume signal at transmitter is $x_T(t)$, the total filter response of transmitter and receiver is $f(t)$, then the received signal is given by

$$x_R(t) = x_T(t) * f(t) + w(t) \qquad (2)$$

where $w(t)$ is additive white Gaussian noise (AWGN). Considering the ADC sampling phase offset t, Let $f_t[n] = f[(n-t)T_s]$, where T_s is the symbol period and t is confined to $|t| < 0.5$. The optimal sampling phase t_{opt} according to a max signal-to-ISI [6] is defined as

$$t_{opt} = \arg\max\Big(\frac{|f_t[0]|^2}{\sum_{\substack{n=-\infty \\ n\neq 0}}^{\infty} |f_t[n]|^2}\Big) = \arg\max(SIR_t). \qquad (3)$$

The normalized expected value of sampled signal is given as

$$E\left\{|x_{R,t}[n]|^2\right\} = |f_t[0]|^2 + \sum_{n\neq 0}|f_t[n]|^2 + \sigma_w^2 \qquad (4)$$

where σ_w^2 denotes the power of the AWGN. Based on equation (3), (4) can be rewritten as

$$E\left\{|x_{R,t}[n]|^2\right\} = (SIR_t + 1)I_t + \sigma_w^2 \quad (I_t = \sum_{n\neq 0}|f_t[n]|^2). \qquad (5)$$

Then we can see that the optimal sampling phase corresponds to a maximum $E\left\{|x_{R,t}[n]|^2\right\}$. However, the calculation of the expected value is unrealistic. So we use the maximum absolute-square sum of finite symbols instead of $E\left\{|x_{R,t}[n]|^2\right\}$, i.e. at the stage of adjusting phases, each symbol is sampled by all the clock phases with different t, then choose the phase that results in the maximum absolute squared sum as the optimal phase t_{opt}. At this stage, the sampling phase is selected by the controller of the loop.

IV. ADAPTIVE INTERPOLATION

As illustrated in Fig. 3, a residual timing error may exist between the sampled signal and the ideal value, so in this section we exploit how interpolation is used to compensate for this difference. Suppose the sampled incoming signal at kth symbol is

$$x(k) = x(kT_s) = \sum_n a_n h(kT_s - nT_s - t_1) = a_k * h_k \qquad (6)$$

where a_k denotes the kth symbol value, h_k denotes the sample of the total impulse response before the timing recovery and t_1 is the timing phase offset that can be expressed as the sampling instant, as illustrated in Fig.3. Since the ideal sampling instant is t_2, the desired sampled signal is

$$r(k) = r(kT_s) = \sum_n a_n g_{k-n} = g_k * a_k \qquad (7)$$

where $g_k = h(kT_s - t_2)$ corresponds to the optimal sample of the total impulse response. Then $r(kT_s)$ can be rewritten as

$$r(kT_s) = g_k * (h_k^{-1} * x(kT_s)) = x(kT_s) * (h_k^{-1} * g_k) \qquad (8)$$

where h_k^{-1} is the sample of the inverse of the continuous impulse response $h(t)$. So the desired signal can be obtained by filtering the sampled signal with a linear filter f_k ($f_k = h_k^{-1} * g_k$), i.e. the interpolator.

Considering that the h_k^{-1} and g_k are unknown, by using first order Taylor approximation [7], we can get g_k by

$$g_k = h_k + \Delta t \cdot h_k^{'} \qquad (9)$$

where $\Delta t = t_2 - t_1$ is the residual timing error and $h_k^{'}$ is the sample of the derivation of $h(t)$.Then the interpolation filter f_k can be given by

$$f_k = h_k^{-1} * (h_k + \Delta t \cdot h_k^{'}) = \delta(k) + \Delta t \cdot (h_k^{-1} * h_k^{'}). \qquad (10)$$

On substituting (10) into (8) and let $w_k = h_k^{-1} * h_k^{'}$, we can get the desired signal

$$\begin{aligned} r(kT_s) &= x_k * f_k = x_k * (\delta(k) + \Delta t \cdot w_k) \\ &= x_k + \Delta t \cdot \sum_n w_n x_{k-n}. \end{aligned} \qquad (11)$$

So the desired signal can be obtained by calculating w_n. The solution is to make it an adaptive FIR filter and update it with adaptive algorithms.

A. Leas Mean Square (LMS) Algorithm

First adaptive algorithm introduced is the Least Mean Square (LMS) [9] algorithm. Assume the weight vector of the N-tap filter is

$$w_n = [w_0(n), w_1(n), \cdots, w_{N-1}(n)]^T. \qquad (12)$$

The incoming sampled signals are

$$x_n = [x(n), x(n-1), \cdots, x(n-N+1)]^T. \qquad (13)$$

For the recovered signal $r(n)$, the corresponding desired signal is $d(n)$, which is supplied by the training sequence at the beginning and by the decision value later. So the error $e(n)$ is given by

$$e(n) = d(n) - r(n) = (d(n) - x(n)) - \Delta t \cdot \sum_n w_n x_{k-n}. \qquad (14)$$

Then we can get the gradient vector as

$$\nabla e^2(n) = \frac{\partial e^2(n)}{\partial w(n)} = -2e(n) \cdot \Delta t \cdot x(n). \qquad (15)$$

Substituting (15) into the updating of the weight vector, we can get the recursive equation

$$\begin{aligned} w_{n+1} &= w_n - \frac{1}{2} \cdot \mu \cdot \nabla e^2(n) \\ &= w_n + \mu \cdot e(n) \cdot \Delta t \cdot x(n) \end{aligned} \qquad (16)$$

where μ is the step size of the LMS algorithm.

B. Recursive Least Square (RLS) Algorithm

To gain a faster convergence speed and better tracking ability, we can also use the Recursive Least Square (RLS) [9] algorithm to update the weight vector $w(n)$ adaptively.

Considering the RLS updating procedure for $w(n)$ and the data structure of recovered signal in (11), we need to make some modifications to use the RLS algorithm.

First, the desired signal is rewritten as the difference between the desired signal and the sampled signal, given by

$$d(n) = d(n) - x(n). \qquad (17)$$

Second, the input vector of the adaptive filter is modified as the multiplication product of the sampled signal and the residual timing error, as

$$x(n) = \Delta t \cdot x(n) \qquad (18)$$

Substituting (17) and (18) into the recursive process of the RLS, we can get the adaptive updating method for w_n, given by

Initialization: $w_0 = \mathbf{0}$ $P(0) = \delta^{-1} I$.

$$k(n) = \frac{P(n-1) x_n \Delta t}{\lambda + \Delta t x_n^H P(n-1) x_n \Delta t} \qquad (19)$$

$$\xi(n) = d(n) - x(n) - \Delta t w_{n-1}^H x_n \qquad (20)$$

$$P(n) = \lambda^{-1}(P(n-1) - k(n) \Delta t x_n^H P(n-1)) \qquad (21)$$

$$w_n = w_{n-1} + k(n) \xi^*(n) \qquad (22)$$

where H denotes Hermite transposition, I is a $N \times N$ unit matrix, δ is a small positive constant and λ is forgetting factor in the range of (0,1).

V. SIMULATION RESULTS

This section investigates the performance of the proposed timing recovery method via computer simulation. In this simulation, QPSK modulation is employed and signals are transmitted in an AWGN channel where the SNR is set to be 30 dB. 200 symbols are used for the MASS module to select the sampling phase, during this period, the sampling phase is provided by the controller and the interpolator coefficients are trained. The numbers of the clock phases are chosen to be 8 and 16 respectively. Simulation considers the worst condition, in which the ideal sampling instant is just between two adjacent phases.

Fig. 4 shows the comparison of the sampled signal and the recovered signal based on RLS interpolation in terms of constellation map. We can see that the residual timing error is well compensated after interpolation, and with the increase of the number of clock phases, a significant performance improvement is obtained.

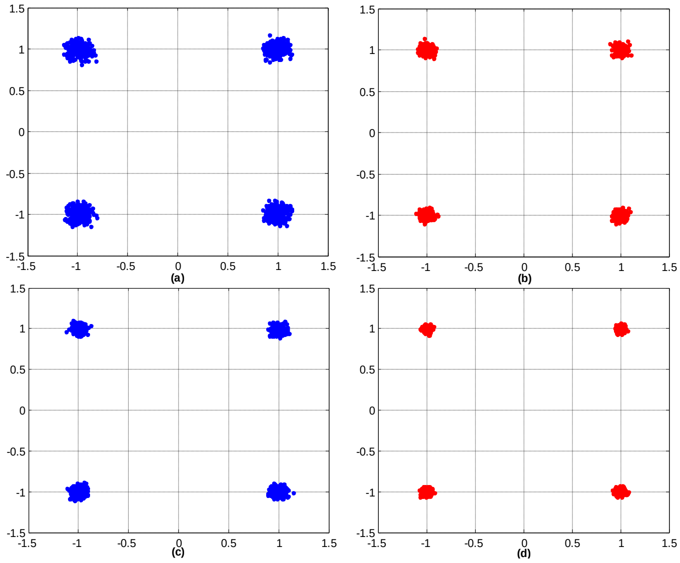

Fig. 4. Constellation maps of (a) sampled signal and (b) recovered signal of 8 phases; (c) sampled signal and (d) recovered signal of 16 phases.

To compare the interpolation performance between LMS and RLS interpolators, the time-averaged signal-to-interference (SIR) [11] is introduced, which is defined as

$$SIR(n) = 10\log\frac{E^2\{r(n)\}}{var\{r(n)\}} \tag{23}$$

where $r(n)$ is the nth recovered signal of interpolator. Fig.5 shows the simulation results of the LMS interpolation and RLS interpolation. The SIR is calculated in 300 independent simulations. It is shown in Fig. 5 that the RLS interpolation has a better performance than LMS even when the sampling phases increase. Since RLS algorithm is more complex than LMS algorithm, the results also reflect a tradeoff between performance and complexity.

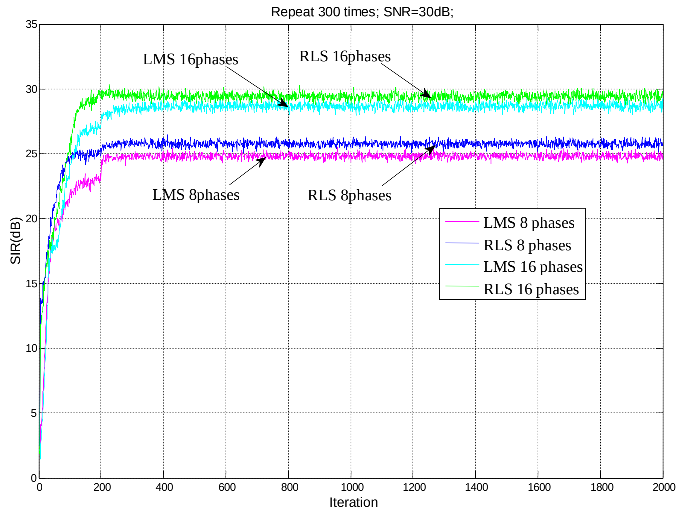

Fig.5. SIR of recovered signal: LMS vs. RLS.

VI. CONCLUSIONS

In this paper, a novel symbol-rate timing recovery approach is proposed for low-energy high-speed wireless communication system to reduce the sampling frequency and power consumption. Continuous signals are sampled at the symbol rate by the ADC based on evenly spaced multiple clocks which run at the symbol rate. A MASS method is used to select the optimal sampling phase. LMS and RLS algorithms based interpolation approaches are introduced to compensate for the performance lose caused by the residual timing error. Simulation results show that sampled signals are significantly recovered after the interpolation process, and increasing in the number of clock phases will also improve the performance. It is also shown that the RLS interpolator outperforms the LMS one given the same amount of clock phases, which reflects a tradeoff between the performance and the complexity.

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