Regular Article

Semi-Blind Timing Skew Calibration in TIADCs Using Second-Order Taylor Approximation and LMS Algorithm

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Abstract– **This paper proposes a technique to calibrate timing mismatches in the digital domain for Time-Interleaved Analog-to-Digital Converters (TIADCs) based on second-order Taylor series approximation for application in high-speed communication and digital signal processing systems. By analyzing the error signal using second-order Taylor series approximation, the proposed technique estimates timing mismatches through the Least Mean Squares (LMS) algorithm to accelerate computation speed and reduce hardware resources. Subsequently, the timing mismatches are corrected based on the second-order Taylor series approximation. The effectiveness of the proposed technique is demonstrated through simulation results in MATLAB software. The simulation results show a significant improvement in the performance of the TIADC. Specifically, the Signal-to-Noise and Distortion Ratio (SNDR) and Spurious-Free Dynamic Range (SFDR) for a 4-channel TIADC are enhanced from 28.29 dB and 33.05 dB to 60.67 dB and 93.66 dB, respectively.**

Keywords– **TIADC, channel mismatch, timing mismatch, calibration, Taylor series approximation, least mean squares algorithm.**

1 Introduction

Nowadays, with the rapid advancement of science, engineering, and technology, communication systems have achieved remarkable progress. Modern wireless communication systems and intelligent signal processing are increasingly digital, particularly in Software-Defined Radio (SDR) systems. In this context, there is a growing demand to bring digital signal processing closer to the antenna. The Analog-to-Digital Converter (ADC) is the device responsible for this digitization process, and it is therefore required to operate at high speed and with high accuracy. A promising solution to meet these requirements is the use of Time-Interleaved Analog-to-Digital Converters (TIADCs). TIADCs increase the effective sampling rate by utilizing M high-precision sub-ADCs that sample the input signal $x(t)$ in parallel at a lower frequency of f_s/M. The outputs of these M ADC channels are then interleaved to obtain a digital output $y[n]$ at the target sampling rate f_s [1]. However, the performance of TIADCs is limited by channel mismatches, including offset, gain, timing, and bandwidth mismatches [2]. Among these, timing mismatch has the most significant impact on the performance of TIADCs. It is challenging to calibrate because it depends on the input signal frequency and is difficult to separate from the signal directly [3]. Therefore, how to calibrate timing errors quickly and accurately is a crucial issue and will be discussed in this paper.

The TIADC model was first introduced in 1980 [1]. Since then, numerous research works have been conducted to enhance the performance of TIADCs, with a focus on correcting channel mismatches. Many of these works specifically address timing mismatches. Based on the form of the calibration signal, there are three methods for calibration: full analog calibration, mixed-signal calibration, and fully digital calibration [4]. Due to the rapid advancement of CMOS technology and digital signal processing techniques, fully digital calibration methods offer superior advantages and have become widely researched and adopted. This approach treats TIADCs as multi-channel systems and various methods have employed filter banks to correct timing mismatches [5–7].

On the other hand, several proposed methods aim to reduce computational complexity. Some methods rely on timing errors between adjacent channels [8, 9], while others are based on the absolute value expectation [10]. Additionally, various other digital domain calibration methods have been proposed [11–16]. Among them, [11] and [12] extend the first-order Taylor series to approximate and then correct timing errors through finite impulse response (FIR) filters. This technique has the advantage of high computational performance and fast calibration. However, each filter requires a set of adders and multipliers, and the resource cost increases linearly with the number of channels. In addition, Kang et al. (2023) [17] introduced a timing-mismatch calibration method based on the Coordinate Rotational Digital Computer (CORDIC) algorithm. Compared to our approach, the CORDIC-based technique can be efficient for certain applications but may introduce additional computational complexity due to iterative rotations.

The methods mentioned above are referred to as blind calibration methods because they do not require knowledge of the input analog signal information. Therefore, these methods usually have lower accuracy and higher complexity than non-blind or semi-blind methods. Non-blind calibration methods rely on complete information about the input signal before calibration, while semi-blind methods require only limited information about the input analog signal [4, 14]. Meanwhile, adaptive calibration methods achieve high accuracy but necessitate multiple iterations, resulting in slower convergence. To overcome these limitations, this paper proposes a new semi-blind timing mismatch calibration method for TIADCs. The proposed method requires only the input signal frequency to be known in advance to perform accurate timing mismatch calibration. It utilizes a second-order Taylor series approximation to analyze the signal for both estimating and correcting timing mismatches. By employing an adaptive algorithm for timing mismatch estimation, the proposed technique achieves high accuracy while reducing computational resource demands. The primary contribution of this paper is the combination of the second-order Taylor series approximation with the LMS algorithm to correct timing mismatches. As a result, the proposed technique demonstrates high performance, fast correction time, and resource efficiency for hardware implementation.

The remainder of the paper is organized as follows. Section 2 presents the mathematical model of TIADC and the proposed calibration method. In Section 3, the paper conducts verification simulations, analyzes the results obtained using MATLAB software, and compares them with relevant works. Finally, the conclusion of the paper is presented in Section 4.

2 Proposed Method

2.1 Mathematical Model of TIADC

To establish the basis for proposing the calibration method, the paper first examines the mathematical model of TIADC. Consider a TIADC with a sinusoidal analog input signal. In this case, the ideal output of the i-th channel is represented as follows

$$x_{i,It}[n] = \sin(2\pi f_{in}.(nM + i).T_s), \tag{1}$$

where f_{in} is the input signal frequency, T_s is the sampling period of TIADC, M is the TIADC number of channels, and $i = 0, 1, ..., M-1$. Considering the TIADC model with only timing mismatch as shown in Figure 1, the output of the i-th channel is rewritten as

$$x_i[n] = \sin(2\pi f_{in}.(nM + i).T_s + \tau_i), \tag{2}$$

where τ_i is the timing mismatch of the ADC in the i-th channel. This mismatch is very small compared to the sampling period T_s. Assuming the input signal is wide-sense stationary and the total timing mismatches among channels sum to 0, i.e

$$\sum_{i=0}^{M-1} \tau_i = 0. \tag{3}$$

The output signals $x_i[n]$ from the channels in Figure 1 are then interleave multiplexed to form the digital output signal of the TIADC $y[n]$ with a sampling frequency of f_s [18].

$$y[n] = x_i\left[\frac{n-i}{M}\right], \text{ where } i = (n \mod M). \tag{4}$$

Calibrating all channel mismatches in a TIADC simultaneously is a complex and challenging task. Therefore, many existing works focus on calibrating specific mismatches, such as offset and gain errors [9, 12, 14, 15, 19, 20]. In this paper, to calibrate timing mismatch, we consider a TIADC model with only timing mismatch, assuming that other mismatches (including offset, gain, and bandwidth mismatches) either do not exist or have been fully calibrated. For timing mismatch calibration, the proposed method employs the Least Mean Squares (LMS) algorithm for estimating timing mismatch and corrects the timing mismatch based on the second-order Taylor series approximation of the TIADC output signal. The general block diagram of the proposed calibration technique is illustrated in Figure 2. Detailed analysis of the proposed method is presented in the following sections.

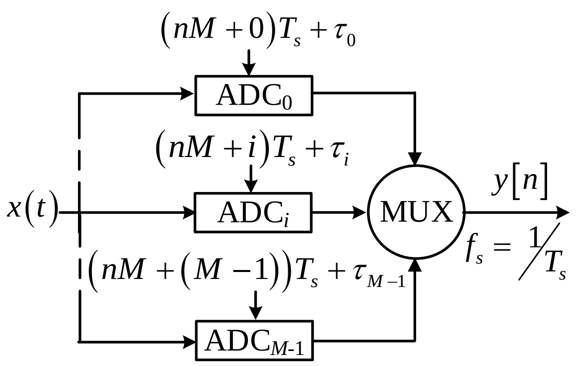

Figure 1. The TIADC model with only timing mismatch.

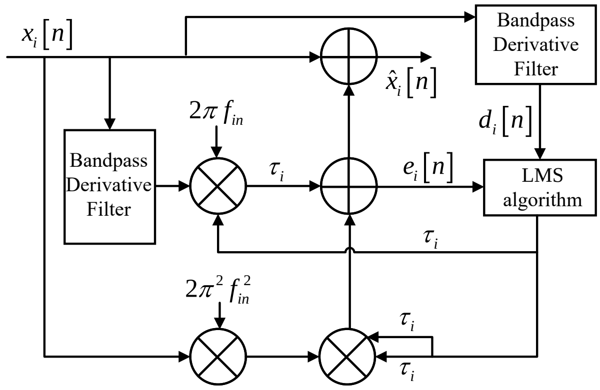

Figure 2. General block diagram of the proposed calibration method.

2.2 Proposed Timing Mismatch Calibration Method

In the frequency domain, the timing mismatch is equivalent to multiplying the ideal signal $x_{i,It}[n]$ by $e^{j\omega\tau_i}$ [12]. Therefore, to eliminate the timing mismatch, it is only necessary to multiply the output of the TIADC by the inverse value of $e^{j\omega\tau_i}$, which is $e^{-j\omega\tau_i}$. It has a second-order Taylor approximation and is expressed as follows

$$e^{-j\omega\tau_i} \approx 1 - j\omega\tau_i + \frac{1}{2}.(j\omega)^2.\tau_i^2. \tag{5}$$

So, the output signal of the TIADC after correcting the timing mismatch is expressed as follows

$$\hat{x}_i[n] \approx x_i[n] - \tau_i.x_i'[n] + \frac{1}{2}\tau_i^2.x_i''[n], \tag{6}$$

where $\hat{x}_i[n]$ is the corrected signal, $x_i'[n]$ and $x_i''[n]$ are the first and second derivatives of the signal $x_i[n]$, respectively, and are expressed as follows

$$\begin{aligned} x_i'[n] &= 2\pi f_{in}.\cos(2\pi f_{in}.(nM+i).T_s + \tau_i) \\ &= 2\pi f_{in}.\dot{x}_i[n], \end{aligned} \tag{7}$$

$$\begin{aligned} x_i''[n] &= -(2\pi f_{in})^2.\sin(2\pi f_{in}.(nM+i).T_s + \tau_i) \\ &= -(2\pi f_{in})^2.x_i[n], \end{aligned} \tag{8}$$

where $\dot{x}_i[n] = \cos(2\pi f_{in}.(nM+i).T_s + \tau_i)$ is the first derivative of the signal $x_i[n]$.

To perform the first and second derivatives of the signal $x_i[n]$, this paper proposes the use of a Bandpass Derivative Filter (BDF) with the structure shown in Figure 3. Using this filter allows it to operate on the input signal at any Nyquist band. The impulse response of the BDF is expressed as follows [4]

$$h_{bdf}[n] = h_d[n] + h_h[n] \times (-1)^{k_{NB}} \left\lfloor \frac{k_{NB}}{2} \right\rfloor 2\pi. \tag{9}$$

where k_{NB} is the Nyquist band order, and $h_d[n]$ and $h_h[n]$ are the impulse responses of the differentiator and Hilbert filters, respectively. The impulse responses of these filters are expressed as follows

$$h_d[n] = \frac{\cos\left(\pi\left(n - \frac{N-1}{2}\right)\right)}{n - \frac{N-1}{2}}, \tag{10}$$

$$h_h[n] = \frac{2}{\pi}\frac{\sin^2\left(\pi\left(n - \frac{N-1}{2}\right)\right)}{n - \frac{N-1}{2}}, \tag{11}$$

where N is the order of the designed filter.

Finally, substituting the formulas (7) and (8) into (6), the signal after timing error correction $\hat{x}_i[n]$ becomes

$$\hat{x}_i[n] \approx x_i[n] - 2\pi f_{in}\tau_i.\dot{x}_i[n] - 2.(\pi f_{in}\tau_i)^2.x_i[n]. \tag{12}$$

According to the formula (12), to correct the timing mismatches at the output of the TIADC, it is necessary to estimate and restore the timing errors on each channel τ_i. To estimate the timing mismatches on each channel τ_i, the proposed method uses the adaptive LMS algorithm. This algorithm provides fast convergence, low computational complexity, and efficient hardware resource utilization.

To find τ_i using the LMS algorithm, the correlation between two error signals, $d_i[n]$ and $e_i[n]$, needs to be calculated. Here, $e_i[n]$ is the recovered error signal for each channel and is expressed as follows

$$e_i[n] = -2\pi f_{in}\tau_i.\dot{x}_i[n] - 2.(\pi f_{in}\tau_i)^2.x_i[n]. \tag{13}$$

To avoid spectral overlap after estimation occurring at positions $k\pi/M$, this paper uses the a Bandpass Derivative Filter to filter the signal $x_i[n]$ before feeding it into the LMS algorithm

$$d_i[n] = x_i[n] * h_{bdf}[n]. \tag{14}$$

To implement adaptive algorithms, it is necessary to calculate the error correlation for each channel $\varepsilon_i[n]$. These signals are determined by the following equation

$$\varepsilon_i[n] = d_i[n] - \hat{e}_i[n] = d_i[n] - \hat{\tau}_i\overline{x}_i[n]. \tag{15}$$

These signals are fed into the adaptive algorithm block to recover the timing errors of each channel $\hat{\tau}_i$ using the LMS algorithm as follows

- Initialize LMS algorithm

$$n = 0, \tag{16}$$

$$\hat{\tau}_i[0] = 0, \tag{17}$$

$$\mu. \tag{18}$$

- Update weights using LMS algorithm

$$\varepsilon_i[n] = d_i[n] - \hat{\tau}_i\overline{x}_i[n], \tag{19}$$

$$\hat{\tau}_i[n] = \hat{t}_i[n-1] + \mu\varepsilon_i[n]\overline{x}_i[n]. \tag{20}$$

The estimated timing error values for each channel are then utilized to correct the timing errors by the expression (12). Note that the accuracy and convergence speed of the LMS algorithm indeed depend on the proper selection of the step size (μ) and initialization parameters. In this paper, these parameters were empirically tuned to achieve optimal performance. Specifically, we performed a series of experiments to test different values of μ and initialization settings, analyzing their effects on convergence speed, stability, and final calibration accuracy. The chosen μ ensures a balance between convergence speed and numerical stability, avoiding excessive oscillations or divergence.

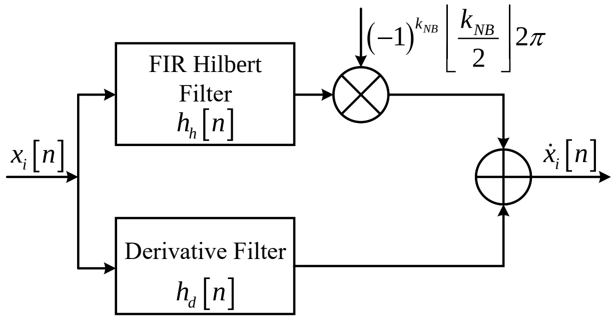

Figure 3. The proposed Bandpass Derivative Filter (BDF) structural diagram.

3 Simulation Results and Analysis

To evaluate the effectiveness of the proposed method, a 14-bit, 4-channel, and 8-channel TIADC are considered for simulation using MATLAB software. The TIADC samples at a frequency of 2.7 GHz. The input analog signal is a sine wave with a frequency of $f_{in} = 0.45 f_s$, where f_s is the sampling frequency. The simulated timing mismatches are set as $\tau = [0.02, -0.015, 0.013, -0.018]$. The number of FFT points is set to 2^{18}. The proposed technique employs a windowing method to design the Bandpass Derivative Filter, and after experimentation, the filter order is chosen to be 31. The results are obtained through Monte Carlo simulations using MATLAB software.

3.1 Estimation Speed

The convergence behavior of the proposed LMS-based calibration method is analyzed through simulation. The selection of the step size (μ) plays a crucial role in determining the speed and stability of the calibration process. We performed extensive empirical testing to optimize μ, ensuring the algorithm converges quickly while maintaining numerical stability. After simulation, the estimation speed of the proposed algorithm is illustrated in Figure 4. According to the simulation results, the estimation values become stable and reach the desired values within 800 - 1000 samples. The convergence speed of the algorithm is fast due to the application of the LMS algorithm. Moreover, this algorithm has low computational complexity and allows for hardware resource reduction. This is the significant advantage of the proposed calibration method.

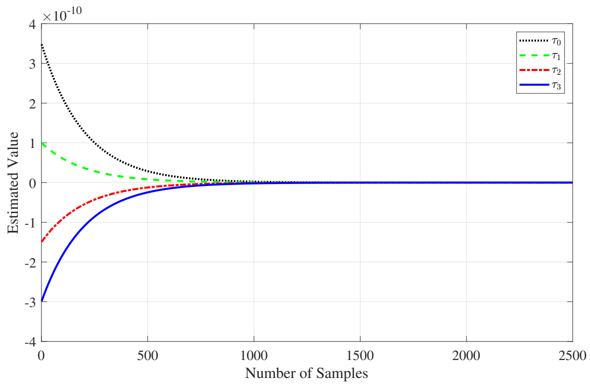

Figure 4. The estimation speed of proposed algorithm.

3.2 Calibration Results

To assess the performance of the proposed algorithm, the paper conducted simulations and obtained results on the Power Spectral Density (PSD) before and after calibration for both 4-channel and 8-channel TIADC. Signal-to-Noise and Distortion Ratio (SNDR) and Spurious-Free Dynamic Range (SFDR) were calculated based on the frequency spectrum results, crucial parameters for evaluating TIADC performance.

The simulation results for the output frequency spectrum of the 4-channel TIADC before and after calibration are illustrated in Figure 5. The simulation results show that the spurs caused by timing mismatches have been completely calibrated, and no longer appear in the output frequency spectrum of the TIADC after calibration. The performance of the proposed technique has also been significantly improved. SNDR has improved by 32.38 dB, from 28.29 dB (before calibration) to 60.67 dB (after calibration). Similarly, SFDR has improved by 60.61 dB, from 33.05 dB to 93.66 dB.

The simulation was also conducted for the 8-channel TIADC, and the results regarding the output frequency spectrum before and after calibration, along with the performance parameters, are illustrated in Figure 6. The simulation results show that the spurs caused by timing errors have also been completely eliminated. The performance of the proposed technique has been significantly enhanced. Specifically, the SNDR has improved by 32.42 dB from 28.25 dB (before calibration) to 60.67 dB (after calibration). Similarly, the SFDR has improved by 60.61 dB from 33.05 dB to 93.66 dB.

In addition, this paper also evaluates SNDR and SFDR with different input signal frequencies. The simulation results are illustrated in Figure 7 and Figure 8, respectively. The simulation results in Figure 7 show that SNDR has improved effectively and stably across the entire Nyquist band. The minimum SNDR improvement reached 28.72 dB. Similarly, Figure 8 shows that SFDR has also improved throughout the whole Nyquist band, with the lowest improvement value reaching 51.94 dB. TIADC has a phenomenon of spectral aliasing at the $0.25 f_s$ frequency. However, the proposed technique still effectively performs calibration.

3.3 Comparison of Proposed Method and Recent Works

The research results in this paper are compared with recent related works, as shown in Table I. The parameters for comparison include the type of channel mismatch, whether the calibration is blind or non-blind, number of ADC channels, bit resolution, input frequency, sampling frequency, convergence time (in samples), improved SNDR (in dB), and improved SFDR (in dB).

To ensure fairness, Table I only compares with works that specifically address the correction of timing errors in TIADC with similar parameters. Blind calibration techniques [12, 15–17] have the advantage of not requiring prior knowledge of input signal parameters. However, these techniques tend to have slower calibration times. In particular, there is a technique with a very slow calibration time [15]. Furthermore, the work in [15] uses multiple iterations to compute the autocorrelation function, resulting in high estimation accuracy but at the cost of increased calibration time and hardware resource consumption. The techniques used in [12, 16] have reduced calibration times, but the blind estimation process still limits the calibration time and performance compared to the proposed technique. The proposed

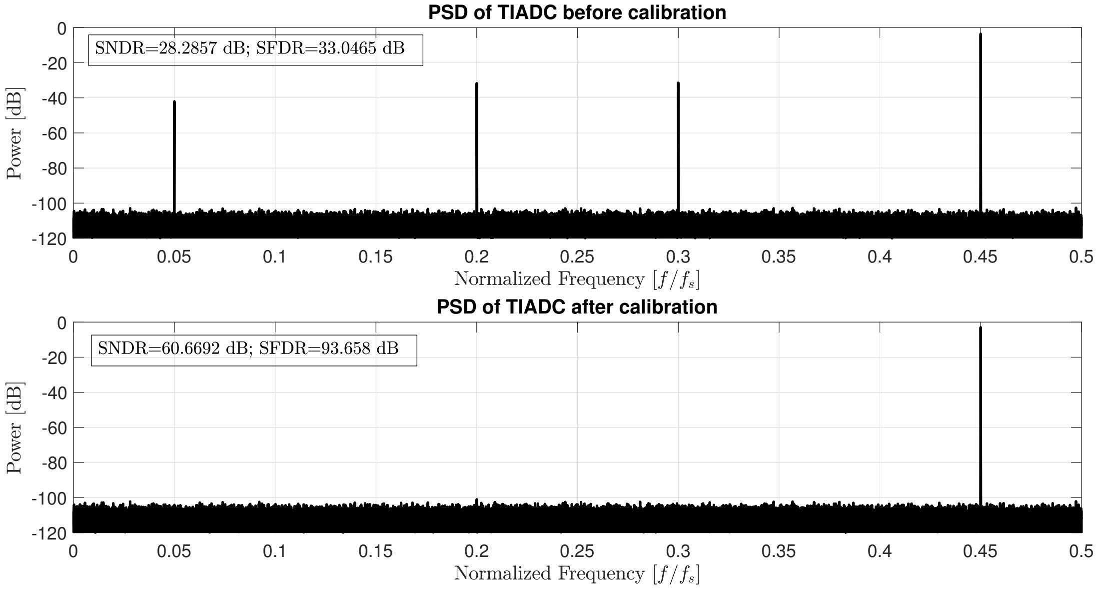

Figure 5. The output frequency spectrum of the 4-channel TIADC before and after calibration.

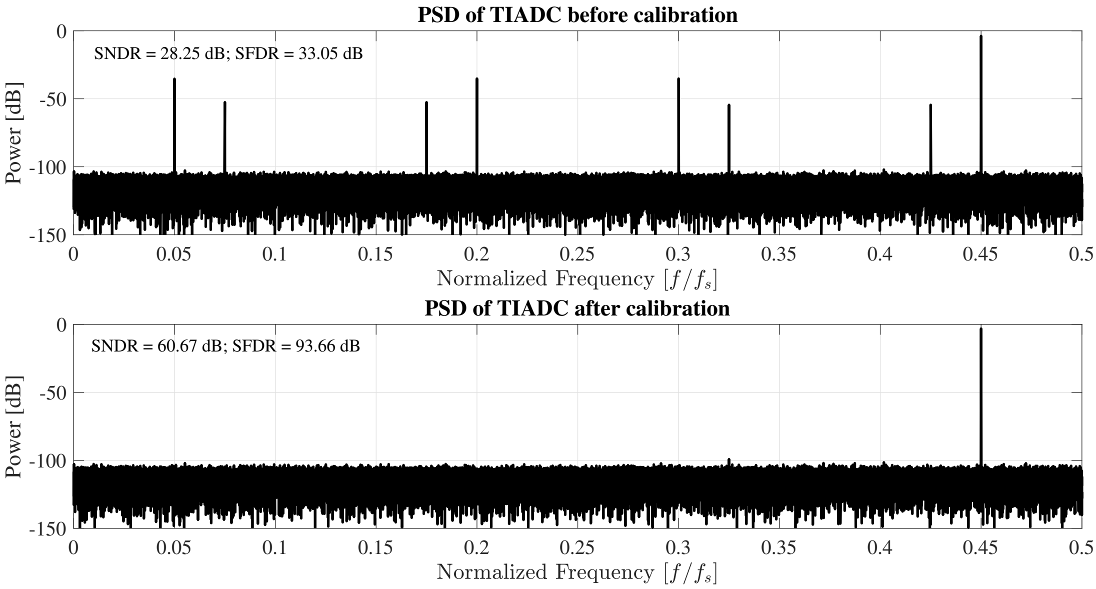

Figure 6. The output frequency spectrum of the 8-channel TIADC before and after calibration.

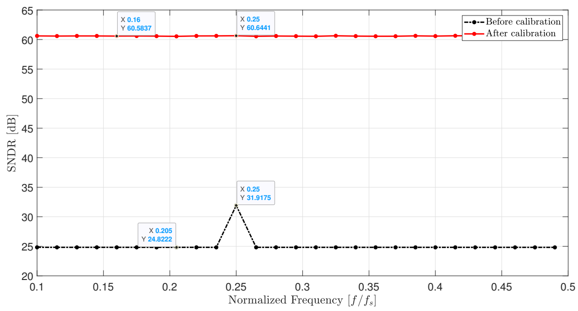

Figure 7. SNDR versus different input frequencies.

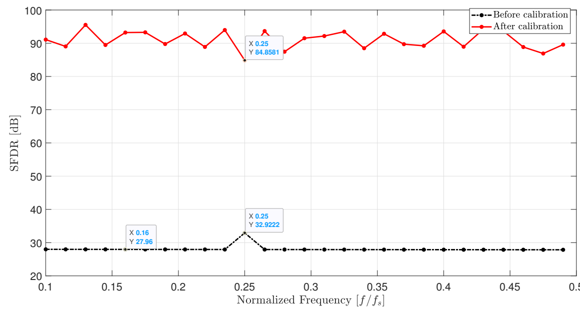

Figure 8. SFDR versus different input frequencies.

method only requires knowledge of the input signal frequency but achieves superior calibration efficiency. This method is well-suited for modern radio communication systems, where transmitters and receivers need to know each other's operating frequencies to capture signals effectively.

Table I
Comparison with Some Other State-of-the-Art Works.

Parameter	[12] 2019	[15] 2021	[16] 2023	[17] 2023	This work
Mismatch types	T*	T*	T*	T*	T*
Blind calibration	Blind	Blind	Blind	Blind	Semi-Blind
Background calibration	Yes	Yes	Yes	Yes	Yes
Number of channels	4	4	4 & 8	4	4 & 8
Number of bits	12	12	14	12	14
Input frequency	$0.15 f_s$	$0.498 f_s$	$0.432 f_s$	$0.437 f_s$	$0.45 f_s$
Sampling frequency	–	1 GS/s	3 GS/s	–	2.7 GS/s
Convergence time (samples)	3.1k	204.8k	6k	1.2k	1k
SNDR improvement (dB)	21 - 38	27.8	24.1	43.9	32.38
SFDR improvement (dB)	22 - 41	44.6	49.43	61.12	60.61

Note: T^* = Timing mismatch

4 Conclusion

This paper has introduced an all-digital semi-blind background calibration method for addressing timing mismatches in TIADCs based on a second-order Taylor series approximation. The proposed method utilizes a Bandpass Derivative Filter and the Least Mean Squares algorithm to speed up computation, reduce hardware resource requirements, and accommodate signals across any Nyquist band. By employing the second-order Taylor approximation to analyze error signals, this technique effectively eliminates timing skews in the TIADC output. Simulation results validate the effectiveness of the proposed method, highlighting its rapid convergence and high calibration accuracy. However, the technique currently requires prior knowledge of the input signal frequency. Future research will focus on developing a fully blind calibration approach in the digital domain to address timing mismatches without needing this prior information.

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