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Article

Research on Analog-to-Digital Converter (ADC) Dynamic Parameter Method Based on the Sinusoidal Test Signal

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Abstract: The acquisition of dynamic analog-to-digital converter (ADC) parameters is becoming increasingly relevant as electronic information develops at a rapid pace. The fundamental challenge of the spectrum test for ADC is that coherent sampling is difficult to achieve, especially because of the high accuracy of the excitation signal necessary for coherent sampling. Therefore, incoherent sampling is certainly unavoidable in the actual test. If the coherent sampling condition is not reached in the test, spectrum leaks from test data after Discrete Fourier Transform (DFT) analysis, resulting in inaccurate parameters. In this study, a new breakdown and reconstruction method was presented using the Ensemble Empirical Mode Decomposition (EEMD); Hilbert transform and parameter fitting method can accurately estimate the incoherent fundamental and harmonic waves, then reconstruct them to obtain accurate ADC dynamic parameters.

Keywords: ADC test; DFT; EEMD; Hilbert transform; incoherent sampling; spectrum test

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1. Introduction

With the rapid development of modern electronic information, the acquisition of analog-to-digital converter (ADC) is becoming increasingly significant. Therefore, the acquisition of ADC dynamic parameters is more and more important for the correct evaluation of ADC. According to the Institute of Electrical and Electronics Engineers (IEEE) standard for Digitizing Waveform Recorders (IEEE Standard 1057) [1] and IEEE Standard for Terminology and Test Methods for Analog-to-Digital Converters (IEEE Standard 1241) [2], dynamic parameters include signal-to-noise ratio, spurious-free dynamic range, and total harmonic distortion. Figure 1 shows the setting of the ADC traditional dynamic parameters test. In the process of data analysis, Discrete Fourier Transform (DFT) is used to test the spectrum to obtain dynamic parameters. The digital signal in the spectrum test is divided into coherent sampling and non-coherent sampling [3–5]. According to the above standards, coherent sampling is recommended, because when the dynamic parameters of the same ADC are obtained, the dynamic parameters obtained under the condition of coherent sampling are accurate, while those obtained by incoherent sampling are incorrect. The spectrum obtained by coherent sampling is true and complete, the spectrum obtained by incoherent sampling, however, will leak. In order to achieve the condition of coherent sampling, a very pure analog signal is required as an excitation, and the frequency of the input signal needs to be well controlled. Since the general self-built test system needs to consider the small area and low cost of the test circuit, such a system cannot provide high-precision chips and clock generators, resulting in the inability to provide accurate input signals, and the failure to achieve coherent sampling with a self-contained oscillator as the signal source in a small test circuit area. Secondly, the excitation signal required for coherent sampling is strict. It is hard to avoid the noise and jitter of the above self-built test system, which leads to difficulty in coherent sampling. In this way, the general self-built test system can only conduct spectrum tests under incoherent sampling. At this time, a serious leakage of the spectrum would appear, resulting in wrong results. Thus, a method to obtain accurate dynamic parameters under incoherent conditions is urgently needed.

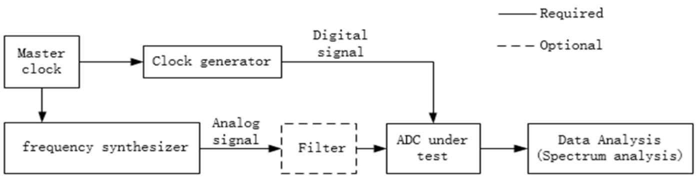

Figure 1. Spectral test setup.

In recent years, many methods have been proposed to suppress spectrum leakage. A long-term method is windowing [6–9]. In these references, if the incoherence is small, the windowing technique is widely used in low-resolution ADC spectrum testing. However, if the spectrum test occurs in the large incoherence or high-resolution ADC test, windowing technology is not sufficient to provide accurate results. Subsequently, the spectrum power of the side lobe of the selected window should be lower than the noise power of the measured ADC, which leads to the outcome that not all windows can effectively suppress spectrum leakage. The other one is the four-parameter fitting method [10–12]. In [11], for example, the iterative process of the four parameters is optimized to obtain better results. However, this method needs to estimate the initial value, and the initial value prediction error will lead to the test failure. Subsequently, when the non-harmonic component determines the Spurious-Free Dynamic Range ($SFDR$), this method cannot provide accurate SFDR values. In [13–18], the interpolation discrete Fourier transform (DFT) method is used, but when the non-harmonic ratio is greater than the harmonic ratio, the obtained results are not optimal. In [19], an additional filter bank is proposed to achieve accurate results, but this method increases the area of the test circuit. In [20], a resampling technique is proposed, which also increases the test area.

All the above methods suffer from one or more of the issues, such as over-reliance on the selection of window types, inability to test high-resolution ADC, whether the estimated value can be estimated as real value, and increased test area. Hence, this paper presents a new spectrum testing method. It can effectively solve the problems of the above methods, carry out a more reliable spectrum testing under incoherent sampling, obtain accurate dynamic parameters, and completely relax the condition of coherent sampling. In the proposed method, incoherence is no longer a problem. Therefore, this method can be used to test any ADC output without knowing the resolution of ADC in advance. Besides, it can reduce the cost of test settings. The low precision signal generator and clock generator can also be used for spectrum testing. In addition, this method can also test the high-resolution ADC without increasing the test system area. First of all, Ensemble Empirical Mode Decomposition (EEMD) and Hilbert transform are used to estimate the frequency parameters of the fundamental frequency. Then, the parameter fitting method is used to fit the incoherent fundamental wave and harmonic wave. Further, the coherent fundamental and harmonic waves are reconstructed, the fitted incoherent fundamental and harmonic waves are subtracted from the initial data, and the reconstructed coherent fundamental and harmonic waves are added. Finally, the reconstructed data is used for DFT calculation to achieve accurate spectrum analysis and obtain accurate ADC dynamic parameters.

The rest of this paper is organized as follows: Section 2 provides the dynamic parameters calculation equation of ADC; Section 3 discusses the ADC spectrum test and introduces the problems of incoherent sampling and coherent sampling; Section 4 details the proposed method and gives the simulation results; Section 5 verifies the method proposed in this paper by using measurement data; Section 6 summarizes the full text.

2. Analysis of ADC Dynamic Parameters

2.1. Signal-to-Noise Ratio (SNR)

The following equation can be used to output an M-bit digital signal:

$$V_{code} = code \times LSB, \tag{1}$$

The Least Effective Bit (LSB) above represents the voltage of the least significant bit in the ADC. For any ADC, the digital signal output after quantization is stepped by the voltage value of $1LSB$. The code represents the dynamic test system output digital coding, and the quantized voltage value of V_{code} can be obtained. Therefore, the quantization noise Err_{rum} can be expressed as:

$$Err_{rum} = V_{pra} - V_{code}, \tag{2}$$

where V_{pra} denotes the actual input voltage value. Then, the average power of quantization noise P_{rum} can be obtained as follows:

$$P_{rum} = \frac{1}{LSB}\int_{-0.5LSB}^{0.5LSB} Err_{rum}^{\;\;2}\, dV_{pra} = \frac{LSB^2}{12}, \tag{3}$$

Since input sine signal, the signal power is calculated as follows:

$$P(f) = \left(\frac{FU}{2\sqrt{2}}\right)^2 = \frac{FU^2}{8} = \frac{(2^m LSB)^2}{8} = 2^{2m-3}LSB^2, \tag{4}$$

where FU represents the full range voltage. Under the condition of full-scale sine wave input, the theoretical maximum SNR of ADC is derived from the quantization noise. On Nyquist bandwidth, SNR can be expressed as:

$$SNR = 10 \times \lg\left(\frac{P(f)}{P_{rum}}\right) = 6.02m + 1.76. \tag{5}$$

Quantized noise is not the only noise source. The calculation results of the SNR are generally obtained by the following DFT data operation:

$$SNR = 10 \times \lg\left(\frac{P(s)}{\sum\limits_{m=2}^{m/2} P(m) - \sum\limits_{k=1}^{h} P(ks)}\right), \tag{6}$$

where $P(s)$ is the power of the input signal, and the denominator represents the power sum of the others except for the direct current component and harmonics components.

2.2. Spurious-Free Dynamic Range (SFDR)

The spurious-free dynamic range is the ratio of the root mean square value of the basic signal to the root mean square value of the maximum distortion component. $SFDR$ represents the effective dynamic region of the measurement system. Once it exceeds this region, the frequency analysis will form a specific threshold problem. Although similar to total harmonic distortion, $SFDR$ also reflects the in-band harmonic distortion characteristics of the measurement system.

$$SFDR = 20 \times \lg\left[\frac{E_{f_{in[rms]}}}{E_{ha\max[rms]}}\right] \tag{7}$$

where $E_{f_{in[rms]}}$ represents the root-mean-square amplitude of the input signal, and $E_{hd\max[rms]}$ denotes the root mean square value of the maximum distortion component.

2.3. Total Harmonic Distortion (THD)

The harmonic component refers to the information that the frequency change in the spectrum is non-integer times of the input signal frequency when the input signal is sampled by the measurement system. Since the nonlinearity of the measurement system is the most important factor in the formation of harmonics, the ratio of the value obtained by the superposition of various harmonic power to the input signal power can be used to represent the THD. h is the total number of harmonics present.

$$THD = 10 \times \lg \left[\frac{\sum_{k=2}^{h} P(ks)}{P(s)} \right], \tag{8}$$

where $P(ks)$ represents harmonic power. lg denotes the logarithm based on 10.

3. Spectrum Test of ADC Coherent Sampling and Incoherent Sampling

At present, the most common method for obtaining ADC dynamic parameters is spectrum testing. By directly obtaining the spectrum of the quantitative results of the test system, the digital quantity is calculated through the Fast Fourier Transform (FFT), and the equation is solved by a certain method to obtain the dynamic parameters of the measurement system. Due to the poor real-time performance of FFT, DFT is generally used for analysis. DFT, as a method of converting time domain information into frequency domain information, has been widely used in the processing of digital information. The digital signal in the spectrum test is divided into coherent sampling and incoherent sampling. Coherent sampling means that in the sampling process, the time obtained by truncation is an integral multiple of the processing signal period. With respect to the incoherent sampling, in the sampling process, the time obtained by truncation is not an integral multiple of the processing signal period. Under incoherent conditions, the DFT method is used for spectrum analysis. At this time, the signal duration is not an integral multiple of the signal period, and the period expansion will lead to phase discontinuity. Then, the spectrum leakage would occur. The spectrum leakage refers to the interaction between the spectral lines in the signal spectrum, which makes the measurement results deviate from the actual value. At the same time, there are some spurious spectra with small amplitudes at other frequency points on both sides of the spectral line, which will lead to a large error in the obtained parameters. In order to reduce the spectrum leakage caused by discrete spectrums, the common method is to realize coherent sampling, after coherent sampling, the spectrum generated by DFT is true and complete. Coherent sampling needs to meet the following requirements:

$$\frac{f_{IN}}{f_{SAM}} = \frac{C_{WIN}}{N_{DATA}}, \tag{9}$$

N_{DATA} denotes the number of the whole period of the input signal, C_{WIN} is the number of complete cycles in the entire system. N_{DATA} and C_{WIN} should be primes each other, requiring N_{DATA} to be an integer power of 2. If the frequency of the input signal f_{IN} is 4 MHz, according to Nyquist, the sampling rate of the whole system f_{SAM} is 40 MHz, so the same assumption N_{DATA} is 16,384 (16 k). If f_{SAM} is 40 MHz, according to the above Equation (9), C_{WIN} is 1638.4; its nearby prime is 1639, and then replace C_{WIN} with the above Equation (9) and get an f_{IN} is 4.00146484375 MHz.

It can be seen from Figures 2 and 3 that when the coherent sampling condition is met, f_{IN} = 4.00146484375 MHz. After DFT analysis, one can conclude that there is no occurrence of leakage. The accurate solution of ADC dynamic parameters, indicates the accurate performance evaluation of ADC. When the coherent sampling condition is not met, such as f_{IN} = 4.0014 MHz or f_{IN} = 4.00146 MHz, DFT analysis results in a serious spectrum leakage, and further inaccurate dynamic parameters are obtained, indicating the inaccurate performance evaluation of ADC. Coherent sampling requires accurate input frequency and pure analog signal. When the input frequency is not accurate, or there is a large noise, it will lead to the emergence of incoherent sampling. This means coherent sampling requires more accurate frequency and noise has a greater impact on it. Considering the requirements for low cost and a small area of the general self-built test system, it is impossible to provide high-precision chip and clock generators, which means it is difficult to achieve the above coherent sampling conditions, in other words, only tests are carried out under incoherent conditions. The spectrum leaks in incoherent sampling, resulting in a test failure. It is necessary to consider adopting various methods to solve the problem of spectrum leakage under incoherent conditions, so as to accurately evaluate the performance of ADC. This paper proposes an effective method to solve the spectrum leakage under the condition of incoherent sampling.

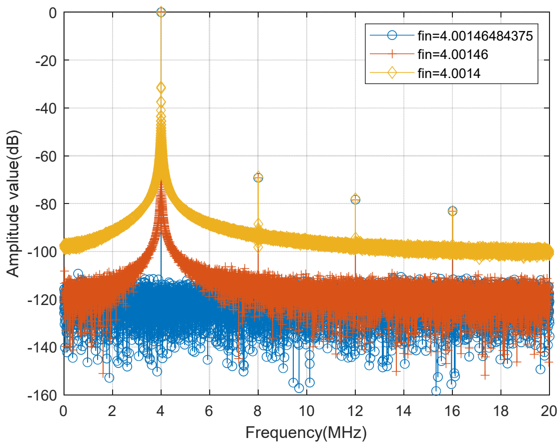

Figure 2. Spectrum comparison of coherent and incoherent sampling.

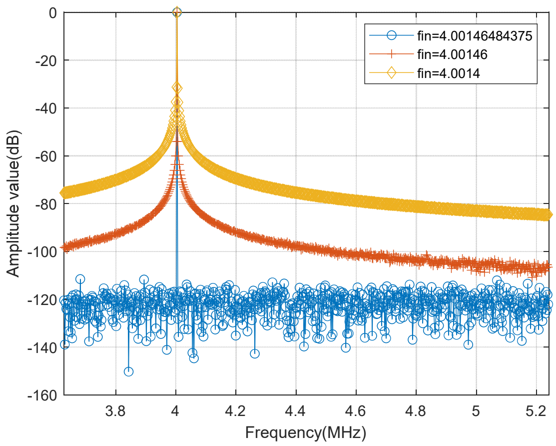

Figure 3. Local amplification of spectral comparison graphs for coherent and incoherent sampling.

4. ADC Dynamic Parameters Extraction Based on EEMD Separation Technology, Hilbert Transform, and Parameter Fitting Technology

Due to the low cost of the self-built test system, it is unable to provide high-precision chips and accurate input frequency. It is difficult to meet the requirements of the coherent sampling, which means that the test system should be tested under the condition of incoherent sampling. Thus, under the condition of incoherent sampling, it is necessary to consider reliable spectrum testing to obtain accurate dynamic parameters. To solve this problem, this paper proposes a dynamic measurement parameter acquisition method based on EEMD sine wave fitting and reconstruction. The EEMD and Hilbert transform are used to estimate the frequency parameters of the fundamental frequency for the first time. Then, the incoherent fundamental wave and harmonic wave are fitted by the parameter fitting method. Further, the coherent fundamental and harmonic waves are reconstructed, the fitting incoherent fundamental and harmonic waves are subtracted from the initial data, and the reconstructed coherent fundamental and harmonic waves are added. Finally, the reconstructed data are used for DFT calculation for accurate spectrum analysis. This can be used to obtain accurate ADC dynamic parameters.

4.1. Extraction, Separation, and Frequency Acquisition of Fundamental Sine Test Signal Based on EEMD and Hilbert

On account of the shortcomings of the Empirical Mode Decomposition (EMD) method, the EEMD method is widely used. EMD method is a new time-frequency analysis method and an adaptive time-frequency localization analysis method, which is proposed by Huang [21]. The deficiency is that the modal component obtained by EMD decomposition has a modal mixing phenomenon. Whereas, EEMD mainly provides a white noise auxiliary analysis method to solve the modal mixing phenomenon. Since added white noise is equally dispersed in a time-frequency space, the time-frequency space will be composed of various scale components divided by the filter bank, according to the basic principle of EEMD analysis. The signal range of multiple scales will be mapped on the appropriate scale corresponding to the background white noise when the average white noise background is applied to the signal. There may be very noisy results in each separate test because each component of additional noise contains both the signal and the added white noise. Since the noise is different in every single test, the noise will be eliminated when all the mean values of the test are adopted, and all the mean values will eventually be regarded as real results. Due to more and more tests, other noises are removed, and the part that is truly durable is the signal itself. EEMD algorithm is an effective method for analyzing and processing nonlinear and non-stationary signals, which solves the problem of mode mixing in signal decomposition. The EEMD method can almost perfectly separate the test information of the fundamental wave.

Harmonic and noise created by the power supply and system peripheral circuits are included in simulation data for the monophonic test system. EEMD technology can efficiently decompose the fundamental wave, but the EEMD method may have a certain endpoint effect. The endpoint data is discarded when applied to practice.

The modal component of the fundamental wave can be acquired by adopting the EEMD strategy. To attain the aim of signal reconstruction, various technical means must be utilized to attain phase, amplitude, frequency, and other parameters. Then, the instantaneous frequency can be attained by the Hilbert transform. Further, the DC component, phase, and amplitude of the fundamental wave are attained by the parameter fitting method. The instantaneous frequency of modal components is obtained by the Hilbert transform.

After the whole test system signal $Y(t)$ is decomposed by EEMD, it is calculated as follows:

$$Y(t) = \sum_{i=1}^{m} x_i(t) + e_m(t), \tag{10}$$

$x_i(t)$ denotes the modal component. $e_m(t)$ represents the residual signal, and m is the number of modal components. Then the signal Hilbert transform can be attained:

$$\hat{x}_i(t) = h(t) * x_i(t) = \int_{-\infty}^{\infty} x_i(t)h(t-\tau)d\tau = \frac{1}{\pi}\int_{-\infty}^{\infty} \frac{x_i(\tau)}{t-\tau}d\tau, \tag{11}$$

In Equation (11) $h(t) = \frac{1}{\pi t}$ and then we can get the corresponding signal of each modal component:

$$r_i(t) = x_i(t) + j\hat{s}_i(t) = \partial(t)e^{j\theta_i(t)}, \tag{12}$$

Subsequently, the amplitude function and phase function in Equation (12) can be expressed as follows:

$$\partial_i(t) = \left[x_i^2(t) + \hat{x}_i^2(t)\right]^{1/2}, \tag{13}$$

$$\theta_i(t) = \arctan(\hat{x}_i(t)/x_i(t)), \tag{14}$$

The instantaneous amplitude and phase of the modal component are represented in Equations (13) and (14), with good instantaneousness. The instantaneous frequency can then be calculated by using Equation (15) since the functional relation of the instantaneous phase has been acquired by Equation (14). Then the instantaneous frequency can be written as follows:

$$f_i(t) = \frac{\omega_i(t)}{2\pi} = \frac{1}{2\pi} \cdot \frac{d\theta_i(t)}{dt}, \tag{15}$$

The solution of instantaneous frequency is local, a function of time and frequency, which cannot be obtained by using the Fourier transform. The fundamental wave of the test signal is decomposed by EEMD, which subsequently acquires all the information required by the modal component. The Hilbert transform of modal components can be used to calculate the instantaneous frequency. The data from both ends of the endpoint is deleted, and the frequency of the basic sinusoidal signal is obtained by fitting the instantaneous frequency of the modal component using the average method. All parameters of the sinusoidal signal are obtained by parameter fitting.

4.2. Sinusoidal Signal Fitting Based on Parameter Fitting

The ideal sine test signal can be described by the following equation:

$$Y(t_n) = A_0 \cos(2\pi f t_n + \varphi_0) + C_0 = B_0 \cos(2\pi f t_n) + D_0 \sin(2\pi f t_n) + C_0, \tag{16}$$

Among them, $Y(t_n)$ denotes the test data of the whole system at t_n times. The whole fitting method is to find A, B, C, and D to complete the minimum sum of residual squares of the following equation when the frequency of the input signal is known:

$$E = \sum_{i=1}^{n} [Y(t_i) - B\cos(2\pi f t_i) - D\cos(2\pi f t_i) - C]^2, \tag{17}$$

Therefore, parameters B, D, and C are the least-squares fitting values of B_0, D_0, and C_0. To find out the matrix constructed by B, D, and C, i in the equation represents the number of iterations.

The implementation phases of the method are obtained as follows:

1. In initialization, the iteration number $i = 1$.
2. Ensure $i = i + 1$ for the next iteration.
3. Obtaining values of A, B, and D by the fitting method.
4. To solve the parameters, the following matrix is established:

$$Y = \begin{bmatrix} Y(t_1) \\ Y(t_2) \\ \bullet \\ Y(t_n) \end{bmatrix}, \tag{18}$$

$$M = \begin{bmatrix} \cos(2\pi f t_1) & \sin(2\pi f t_1) & 1 \\ \cos(2\pi f t_2) & \sin(2\pi f t_2) & 1 \\ \bullet & \bullet & \bullet \\ \cos(2\pi f t_n) & \sin(2\pi f t_n) & 1 \end{bmatrix}, \tag{19}$$

$$X_0 = \begin{bmatrix} B \\ D \\ C \end{bmatrix}, \tag{20}$$

5. The total of residual squares is calculated as follows:

$$E = (Y - MX_0)^T (Y - MX_0), \tag{21}$$

6. When E is the smallest in Equation (21), the least-squares solution of $\hat{X}_0$ is obtained as follows:

$$\hat{X}_0 = (M^T M)^{-1} (M^T Y), \tag{22}$$

7. The amplitude and phase expressions of the fitting function are as follows:

$$y(t_n) = A\cos(2\pi f t_n + \varphi) + C, \tag{23}$$

$$A = \sqrt{B^2 + D^2}, \tag{24}$$

$$\varphi = \begin{cases} \arctan(\frac{-D}{B}); B \geq 0 \\ \arctan(\frac{-D}{B}) + \pi; B \leq 0 \end{cases}, \tag{25}$$

8. The fitting residual is as follows:

$$r(t_n) = y(t_n) - B\cos(2\pi f t_n) - D\cos(2\pi f t_n) - C, \tag{26}$$

The parameters of the sine signal can be derived using the procedures above, resulting in the fitted sine signal being stated as follows:

$$y_f(t_n) = A\cos(2\pi f t_n + \varphi) + C. \tag{27}$$

A represents amplitude, f denotes frequency, φ represents phase, and C denotes direct current component. The sine curve of the fundamental wave has been fitted at this point, and then the residual sine curve signal is obtained by subtracting the fitting sine curve of the fundamental wave from the initial signal:

$$Y_h(t_n) = Y(t_n) - y_f(t_n). \tag{28}$$

The sine curve of the second harmonic is obtained at this point. Because the fundamental frequency is known, the second harmonic frequency may also be calculated. Then, by using the above parameter fitting steps, the amplitude, phase, and DC components are fitted. Thus, the second harmonic sine can be obtained:

$$y_h(t_n) = A_h \cos(2\pi f_h t_n + \varphi_h) + C_h. \tag{29}$$

In addition to the second harmonics mentioned above, residual harmonics can be obtained by using the above steps.

4.3. Signal Reconstruction

From Equations (9) and (27), the fundamental wave signal and harmonic signal obtained by fitting above are reconstructed at this time. We can obtain:

$$y_f(t_n) = A\cos\left(\frac{C_{WIN} + \beta}{N_{DATA}} 2\pi f_{SAM} t_n + \varphi\right) + C. \tag{30}$$

It is known that Equation (30) is expressed as a coherent test when $\beta = 0$, otherwise it is expressed as an incoherent test when $\beta \neq 0$, and Equation (27) is expressed as the sine signal of the fundamental wave obtained by fitting. The frequency can then be determined to be the same. Then it can be obtained:

$$f = \frac{C_{WIN} + \beta}{N_{DATA}} f_{SAM}, \tag{31}$$

Then, according to Equation (31), we can get:

$$C_{WIN} + \beta = N_{DATA} \frac{f}{f_{SAM}}, \tag{32}$$

The f, N_{data}, and f_{sam} in Equation (32) have been correctly known. An accurate value $C_{WIN} + \beta$ can be obtained. Since N_{data} and $C_{WIN} + \beta$ must be integers, and N_{data} must be an integer power of 2. Then, after calculating $C_{WIN} + \beta$, the nearest integer value is taken and revalued C_{WIN_NEW}, and then the frequency of the reconstructed fundamental signal can be obtained as follows:

$$f_N = \frac{C_{WIN\_NEW}}{N_{DATA}} f_{SAM}, \tag{33}$$

The newly obtained f_N is substituted into Equation (27) to obtain the sinusoidal signal of the reconstructed fundamental wave:

$$y_{NEW\_f}(t_n) = A\cos(2\pi f_N t_n + \varphi) + C, \tag{34}$$

Similarly, the reconstructed signal of the new second harmonic can be obtained by replacing f_h the fitted second harmonic:

$$y_{NEW\_h}(t_n) = A_h \cos(4\pi f_N t_n + \varphi_h) + C_h, \tag{35}$$

Residual harmonic can also be reconstructed. The fitted fundamental and second harmonic can be removed from the test data using Equations (27), (29), (34) and (35), and the reconstructed fundamental and second harmonic can be utilized in their place. After replacement, the output is as follows:

$$y_{NEW}(t_n) = Y(t_n) - y_f(t_n) + y_{NEW\_f}(t_n) - y_h(t_n) + y_{NEW\_h}(t_n), \tag{36}$$

Residual harmonics can also be replaced as above. The whole process is shown in Figure 4.

4.4. Simulation Results

In this section, the simulation results that verify the proposed method are given, as well as the results of this method with large or small incoherence. The gap between coherent sampling and incoherent sampling is expressed by β.

MATLAB is used to generate data by 16-bit ADC. The true THD, $SFDR$, and SNR values of ADC are obtained by sending coherent sampled sinusoidal signals. The total number of recorded points is 16,384.

Figure 5 shows the spectrum of 16-bit ADC when the input signal is coherently sampled. The spectrum does not leak, and THD, $SFDR$, and SNR values are obtained from the spectrum.

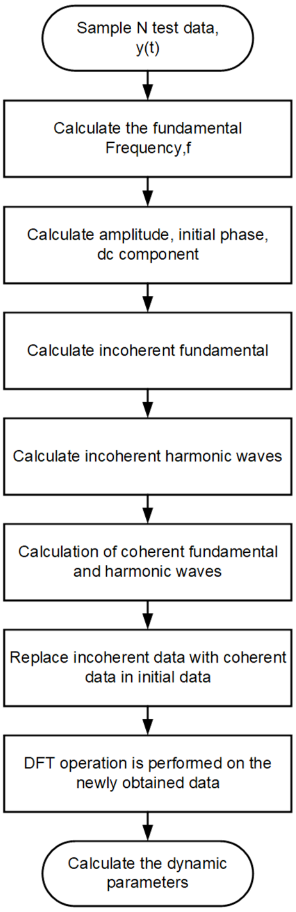

Figure 4. Process step diagram.

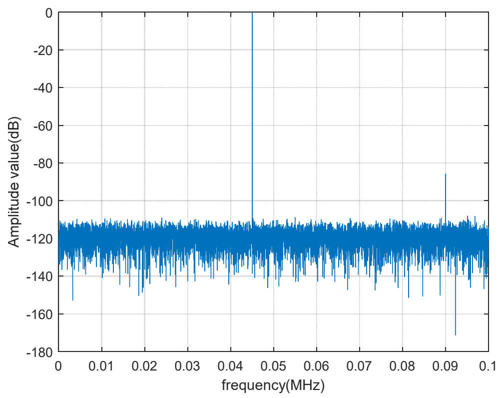

Figure 5. Spectrum of a coherently sampled ADC using direct DFT (N_{DATA} = 16,384, C_{WIN} = 3686).

Figure 6 shows the spectrum of direct DFT for incoherent sampling data. In this case, the value of β is 0.4. When direct DFT obtains the spectrum, the spectrum leaks seriously.

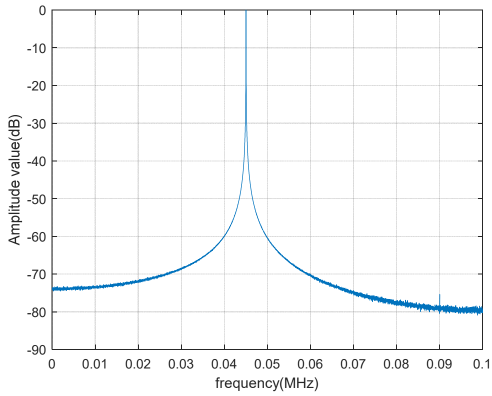

Figure 6. Spectrum of an incoherently sampled ADC using direct DFT (N_{DATA} = 16,384, C_{WIN} = 3686.4).

Figure 7 shows the spectrum when the proposed method is applied to the data under incoherent sampling used in Figure 6. As shown in the figure, the spectrum basically has no leakage.

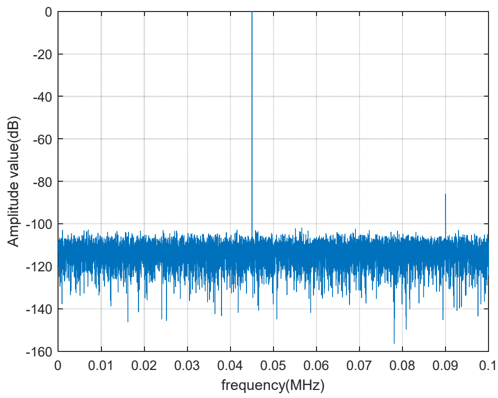

Figure 7. Spectrum of an incoherently sampled ADC after using the proposed method (N_{DATA} = 16,384, C_{WIN} = 3686.4).

It can be seen from Tables 1 and 2 that the SNR, THD, and $SFDR$ values of ADC are obtained by coherent sampling, and the SNR, THD, and $SFDR$ values are obtained by the direct DFT method and the proposed method when the incoherent sampling degree β is 0.02 (small value) and 0.4 (large value). The results show that under the condition of coherent sampling, the dynamic parameters of ADC are accurate, which indicates that the evaluation of ADC is also correct. Under the condition of incoherent sampling, after the direct use of DFT analysis, the spectrum leakage occurs, and the obtained dynamic parameters are significantly different from the data under coherent sampling, indicating the inaccurate evaluation of ADC. The spectrum obtained by the proposed method does not leak, and the difference between the dynamic parameters obtained by the proposed method and the data obtained by coherent sampling is very small, which indicates that the proposed method can obtain the dynamic parameters of ADC under different incoherent degrees and accurately evaluate the performance of ADC.

Table 1. The *THD*, *SFDR*, and *SNR* values obtained under coherent and incoherent sampling, β = 0.4.

	Coherent + DFT	**Incoherent + DFT**	**Incoherent + Proposed Method**
THD (dB)	−101.6	−66.3	−100.5
SFDR (dB)	103.4	68.1	102.6
SNR (dB)	84.8	52.7	84.1

Table 2. The *THD*, *SFDR*, and *SNR* values obtained under coherent and incoherent sampling, β = 0.02.

	Coherent + DFT	Incoherent + DFT	Incoherent + Proposed Method
THD (dB)	−101.6	−89.9	−101.4
SFDR (dB)	103.4	92.6	103.1
SNR (dB)	84.8	75.3	84.7

5. Hardware Platform Design and Implementation

5.1. Hardware Platform Design

The motherboard control board, verification board, and their connection to industrial computer and equipment design are all part of the hardware design. Simultaneously, the motherboard uses Field-Programmable Gate Array (FPGA) as the core device, which primarily completes the control of the output data of the analog-to-digital converter. It performs data caching, cross-clock domain processing, DDR3 interface, read-write control, and data transmission all at the same time. Firstly, FPGA has been widely used in the field of communication and digital signal processing with its high-speed parallel processing capability. Secondly, FPGA can ensure good control frequency to control ADC. The test of ADC dynamic parameters needs auxiliary instruments, such as a high-precision signal generator, high-precision multimeter, and high-precision power supply. These signal sources need to have excellent performance (low phase noise, flat frequency response, and moderate harmonic performance). Additional filtering is applied between the signal generator and the ADC analog input because the harmonic performance of these generators is generally not as excellent as the intrinsic linearity of a specific ADC. Pure excitation signal can be obtained by filtering sine signal produced by a signal generator. Figure 8 depicts the hardware platform for the test.

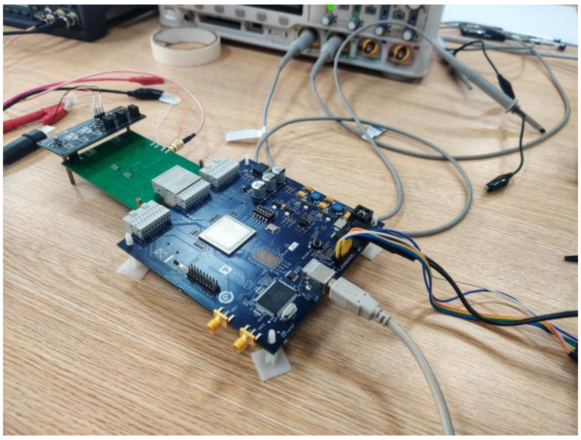

Figure 8. Field test diagram.

5.2. Comparison of Actual Results

The suggested method is verified by the above hardware platform. AD976 is a 16-bit analog-to-digital converter. For the sine wave with coherent sampling input of 40.002441406 KHz, N_{DATA} = 16,384, C_{WIN} = 3277, and for the sine wave with incoherent sampling input of 40 KHz, C_{WIN} = 3276.8. First of all, it can be seen from Figure 9 that after the windowing is used under the condition of incoherent sampling, the spectrum leakage still occurs compared with that obtained by coherent sampling. However, after using the method proposed in this paper, there is basically no spectrum leakage compared with the spectrum obtained by coherent sampling. Secondly, it can be seen from Table 3 that the THD, $SFDR$, and SNR obtained by coherent sampling are −99.7, 101.9, and 83.5, respectively. Then the THD, $SFDR$, and SNR values obtained by windowing in incoherent sampling still have a certain gap from those obtained by coherent sampling. However, there is almost no difference between the values obtained by using the method proposed in this paper and those obtained by coherent sampling, which can indicate that the proposed method is more accurate for ADC evaluation.

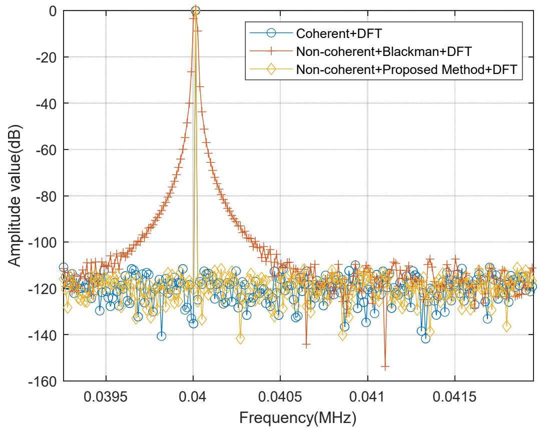

Figure 9. Spectrum comparison of different methods under coherent and incoherent sampling.

Table 3. The *THD*, *SFDR*, and *SNR* values were obtained under coherent and incoherent sampling.

	Coherent Sampling	Blackman Window	Proposed Method
THD (dB)	−99.7	−98.2	−99.5
SFDR (dB)	101.9	100.3	101.8
SNR (dB)	83.5	83.1	83.5

6. Conclusions

In this paper, a new spectrum measurement method was proposed to eliminate the need for coherent sampling. Firstly, the simulation results showed that regardless of the value of incoherent degree β, the THD, $SFDR$, and SNR values obtained by the proposed method under the condition of incoherent sampling are basically the same as the true THD, $SFDR$, and SNR values of the ADC obtained by coherent sampling, which indicates that the proposed method can be applied to the whole Nyquist range and has strong functionality. Secondly, the experimental results showed that compared with the windowing method, the spectrum of this method is basically without leakage, and the consideration of window selection is omitted, which has better accuracy and wider applicability. Moreover, this method does not need to know the resolution of ADC in advance, and can easily test any ADC output, and the obtained THD, $SFDR$, and SNR values are also more accurate. At the same time, because the method can be coherent sampling conditions, a low precision signal generator and clock generator can also be used for spectrum testing. This reduces the cost and area of the general test system. Finally, when the distortion power is equivalent to the basic power, this method may not work. It can be said that this method can effectively deal with the acquisition of ADC dynamic parameters and the accurate evaluation of ADC under incoherent test conditions, expanding the application of ADC in the field of electronics [22].

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