



Synthetic Sparse Matrix Algorithm for Computer Aided Circuit Analysis

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Abstract. The principle of sparse sampling of circuit analysis coded ultrasonic signal is given, and the sparse sampling data is obtained. The principle of reconstructing the characteristic parameters of defect echo using sparse data is analyzed, and the mathematical model of parameter reconstruction is established. In this paper, the characteristic parameters of echo signal, echo peak delay and amplitude parameters, are reconstructed by using the annihilation filter algorithm. The algorithm of feature parameter reconstruction based on FPGA is designed, the corresponding program is compiled, the simulation experiment of sparse data parameter reconstruction is carried out, and the timeliness difference between FPGA and matlab program reconstruction algorithm is compared and analyzed. Comprehensive sparse matrix algorithm has good numerical stability and can keep the sparsity of coefficient matrix. The storage capacity, the amount of multiplication and division and the time to solve the problem are proportional to the total number of non-zero elements in the coefficient matrix. Therefore, the algorithm is suitable for the analysis of large circuits.

Keywords: Computer Aided; Circuit Analysis; Sparse Coding; Sparse Matrix

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1 INTRODUCTION

With the development of computer-aided instruction, the introduction of computer numerical method into circuit analysis is to integrate different scientific research methods into teaching, which is helpful to simplify calculation and formula derivation. It liberates us from the tedious calculation, and puts our energy on the thinking and research of circuit analysis, and lays a foundation for solving more complex practical problems in the future [1]. The research method of computer numerical simulation has become the third research method besides experimental research and theoretical analysis, and has been widely used in circuit analysis. Potential circuit is a

kind of circuit path or state hidden in the circuit with the circuit design, it can only work in specific circuit conditions [2]. Once the potential circuit is activated, it may produce unexpected functions, or inhibit the expected functions, causing system failure, or even serious accidents, including equipment damage and casualties. Because the potential circuit has great harm, it is necessary to analyze the potential circuit of the designed circuit to find out the potential circuit in the circuit. By improving the circuit or destroying the excitation condition of potential circuit, the harm of potential circuit can be effectively reduced [3].

The essence of circuit sparse sampling is to construct the signal with fri characteristics into the form of power series weighted sum. The key information parameters of the original signal are included in it, and the parameters of the key information parameters are estimated through the spectrum estimation algorithm. In computer aided circuit analysis, Wang et al. [4] proposed reproducing sparse sampling kernel, pointed out that functions satisfying the strang fix condition can be fri sampling kernel, and gave three types of reproducing kernel: exponential reproducing kernel, polynomial reproducing kernel and rational function kernel. Among them, the reproducing sampling kernel can be transformed into rational function kernel, which can be approximated by RLC circuit. Nechma and Zwolinski [5] designed the SOS sampling kernel. The kernel function of the SOS sampling kernel is the weighted sum of sinc functions, which is equivalent to relaxing the restrictions of sampling class sampling kernel, making it infinite in frequency domain and finite in time domain, and can be realized by physical approximation. Jia et al. [6] and his research team proposed fri sparse sampling method for ultrasonic signals for the first time. Ahmad et al. [7] studied the application of fri sparse sampling theory in ultrasonic phased array detection. Rajalakshmi et al. [8] proposed a new method to construct the strang fix sampling core, which fully considered the frequency domain characteristics of the sampling core, making it possible for FRI to reconstruct the signal accurately and perfectly after sparse sampling.

At present, SCA has been widely used in aerospace, military, communication network, nuclear power and other fields abroad. However, this technology started relatively late in China. So far, the domestic development of latent path analysis systems include a computer-aided network tree generation system developed by 12 Institutes of the Ministry of space, uest-611-301 SCAs, a computer-aided latent path analysis software jointly developed by 301 institutes and 611 Institutes of the Ministry of space and Chengdu University of Electronic Science and technology [9]. The automatic software is mainly used in the aerospace field, while SCA research is rarely carried out in other fields. For a long time, the safety and reliability of power electronic system mainly focus on fault diagnosis. Recently, it has been found that there is potential circuit in power electronic system. The research on potential circuit in power electronic system is just beginning, and it is still in the stage of manual analysis. Therefore, it is necessary to develop a computer analysis software for potential circuit in power electronic system to meet the needs of research.

2 CIRCUIT SAMPLING SYSTEM BASED ON SPARSE MATRIX SYNTHESIS

2.1 Scheme Design of Sparse Sampling System

In this paper, a synthetic sparse matrix algorithm is introduced. When necessary, the algorithm reselects the principal component, performs symbolic $L * U *$ decomposition and numerical $L * U *$ decomposition again. When selecting the principal component, it ensures the numerical stability of the elimination process and maintains the sparsity of the coefficient matrix. Therefore, this algorithm is suitable for solving high-order sparse linear algebraic equations in computer-aided circuit analysis. The research of this algorithm shows that: for most practical circuits, the accuracy of the synthetic sparse matrix algorithm is very close to that of the all the principal element elimination method, the filling amount is between 0 and $1.8n$, and the multiplication and division amount is between $4N$ and $15N$ is the order of the coefficient matrix [10]. The storage capacity and solution time are proportional to the total number of non-zero elements of the coefficient matrix, so good results are achieved.

The conventional ultrasonic signal sampling system is based on the traditional Nyquist sampling theorem. The A / D sampling rate is not lower than the Nyquist frequency of the ultrasonic signal. The purpose is to collect all the frequency components of the ultrasonic signal completely. Through a large number of collected data, the reconstruction of the signal becomes simple, and the original signal can be recovered with any precision. The FRI sparse sampling system of pulsed ultrasonic signal needs to complete the time-domain preprocessing of pulsed ultrasonic signal before A / D sampling. Before A / D sampling, the pulse flow of pulsed ultrasonic signal is formed by analog circuit, and the output signal of FRI sampling core is obtained through the FRI sampling core circuit. Taking the maximum local innovation rate of ultrasonic pulse flow signal as A / D sampling rate, the FRI sampling core output signal is sparsely sampled at low rate and equal interval, and the key information parameters of the sampled signal are estimated from the sparsely collected data. Compared with the conventional ultrasonic signal sampling system, FRI sparse sampling system has the following characteristics: before A / D sampling, the sampled signal needs to be constructed into FRI signal to meet the requirements of FRI sparse sampling, so as to reduce the sampling rate. Since the amount of data collected is far less than that of conventional sampling, it is impossible to use conventional reconstruction methods to recover all the information in the original signal. Instead, a relatively complex parameter estimation algorithm is used to estimate the key information parameters of the sampled signal from the sparse sampled data, and the signal is reconstructed with the key information parameters.

According to the content of this paper, the ultrasonic testing signal in the form of pulse excitation can be regarded as the Gaussian signal modulated by the ultrasonic transducer:

$$p(t) = \sum_{n=1}^N \alpha_n e^{-\frac{(t-t_l)^2}{\beta n^4}} \cos(2\pi g(t-t_n)) \quad (1)$$

Where, L is the number of echoes; β is the echo amplitude coefficient; αl is the echo pulse width; tl is the echo arrival time; f_0 is the center frequency of ultrasonic detection signal; and l is the initial phase. The Gaussian pulse envelope $g(t)$ is extracted:

$$f(t) = \sum_{n=1}^N \alpha_n e^{-\frac{(t-t_n)^2}{\beta n^2}} + \phi(t) \quad (2)$$

It can be seen from the above formula that the ultrasonic pulse flow can be represented by a finite number of degrees of freedom β , $tl = 1 \dots L$, which has the characteristics of FRI signal and can be sparsely sampled by FRI.

Furthermore, compared with the pulse ultrasonic signal, the pre-processing of FRI sparse sampling system is different. The original coded ultrasonic signal contains complex information, and does not have the characteristics of FRI signal, which cannot meet the requirements of FRI sparse sampling, and cannot use FRI sparse sampling theory to sample directly at low rate. If the coded ultrasonic signal is directly envelope detected to construct pulse stream, the range resolution will decrease because of the longer oscillation duration of the signal, and the waveform of the coded ultrasonic signal is not regular, so it is difficult to accurately extract its key information parameters. Therefore, we have the research on the construction of coded ultrasonic signal pulse stream based on high-order moments and FRI sparse sampling method. The research content is applied in the pipeline defect coding ultrasonic testing. Combined with the FRI sampling core circuit and other supporting equipment, the FRI sparse sampling scheme of coded ultrasonic signal is designed. The software and hardware of the system are designed according to the coded ultrasonic testing signal of pipeline defects, as shown in Figure 1.

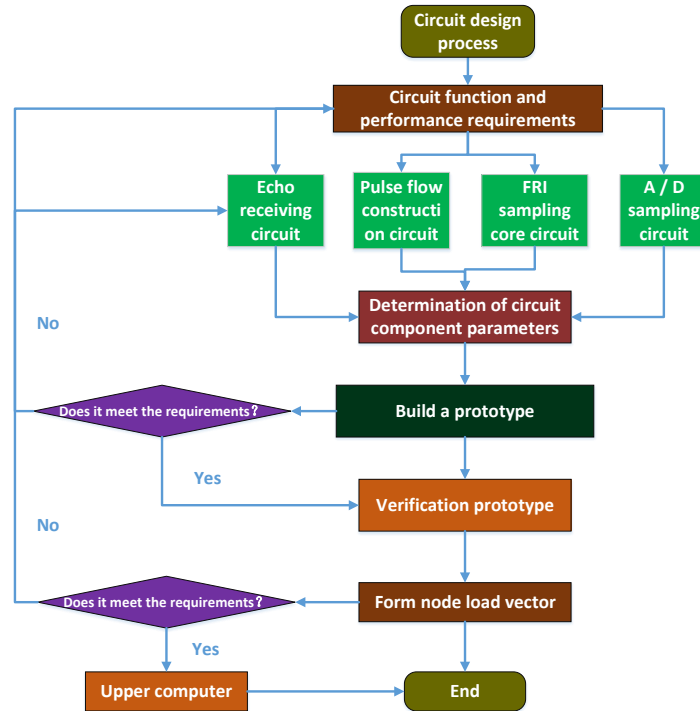


Figure 1: Design of sparse coding circuit.

Firstly, the upper computer drives the ultrasonic transducer to transmit the coded ultrasonic signal to detect the pipeline defects by programming the coded signal generation and excitation circuit. The coded ultrasonic signal passes through the echo receiving circuit, and then the high-order moment pulse flow of the coded ultrasonic signal is obtained through the pulse flow construction circuit. The output signal of FRI sampling core is obtained from coded ultrasonic pulse flow signal through FRI sampling core circuit, and the maximum local innovation rate of high-order moment pulse flow of actual pipeline defect coded ultrasonic testing signal is taken as A / D sampling rate. Through the A / D sampling module, the FRI sampling core output signal is sparsely sampled at equal intervals, and the sparsely sampled data is transmitted to the host computer. The corresponding parameter estimation algorithm is run in the host computer to calculate the sparsely sampled data, from which the key information parameters of the sampled signal are estimated, and then the coded ultrasonic detection signal is reconstructed to obtain the detection results.

2.2 Software Design of Sampling System

In Nyquist Shannon sampling theory, the sampling interval is shortened, the A / D sampling frequency is increased, and a large number of data are collected to ensure the integrity of the frequency information of the sampled signal, in exchange for the undistorted signal, so that the reconstruction algorithm becomes simple, and the complete waveform of the sampled signal can be reconstructed at any precision. In FRI sparse sampling theory, the parameters of information degree of freedom of FRI signal are recovered from the sparse sampling data by using the parametric characteristics of FRI signal and the corresponding parameter estimation and reconstruction algorithm. The following is a brief introduction of the annihilation filter and subspace estimation of these two FRI parameter estimation and reconstruction algorithms.

(1) Annihilation filter

The mathematical model of annihilation filter algorithm is the linear combination of complex exponential functions, while the Fourier coefficients of FRI signal are the linear combination of complex exponential functions in the form of weighted sum of power series. The zeroing filter algorithm is used to estimate and reconstruct the parameters from the FRI sparse data. The root of the zeroing filter is the complex exponent of the Fourier coefficients of the FRI signal. The key information parameters of FRI signal are included in it, so the relationship between the coefficients of the annihilation filter and the Fourier coefficients of FRI signal can be obtained. By solving the formula, the root of the annihilation filter can be obtained, and then the key parameter information of FRI signal can be obtained.

(2) Subspace estimation algorithm

Subspace estimation algorithms include matrix pencil method, rotation invariant signal parameter estimation method and multiple signal classification method. The rotation invariant signal parameter estimation method uses the covariance matrix subspace translation invariance of sparse sampling data for parameter estimation, and the multiple signal classification method mainly uses the Fourier coefficient matrix subspace translation invariance of sparse sampling data for parameter estimation. The subspace estimation algorithm has better anti-noise performance than the zero-filter algorithm, but at the cost of increasing the number of sampling points and requiring more sampling data.

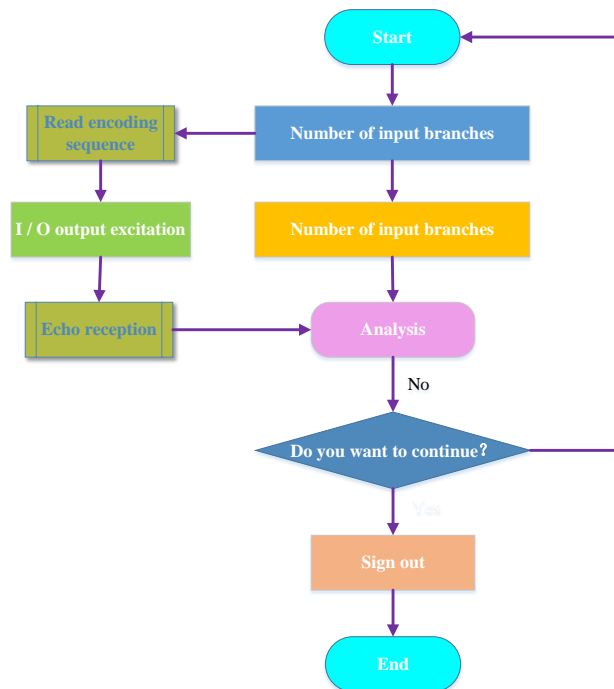


Figure 2: Flow chart of main control program.

The encoding excitation sequence is controlled by FPGA. The encoding length, symbol frequency, symbol combination and repetition frequency of binary frequency coded signal can be changed by programming FPGA separately. The encoding sequence is stored in the external flash in advance, and the main control program is developed in LabVIEW environment. The host computer reads and calls the data through the main control program. At the beginning of detection, the coding sequence is read and output through the assigned I / O pin, the start identification bit is given, the sampling gate is opened, and the A / D sampling is started. After receiving and fri sampling

pretreatment, the coded ultrasonic echo signal is directly input into the A / D sampling module, and the A / D sampling rate is set by the upper computer for sampling. According to the preset sampling gate time, the cut-off identification bit is set to close the sampling gate. The digital signal output by the A / D sampling module is transmitted to the host computer through the bus. The MATLAB software in the host computer reads and runs the parameter estimation algorithm. After the parameter estimation is completed, the result is displayed and the completion identification bit is given for the next cycle. The flow chart of main control program is shown in Figure 2.

3 COMPUTER AIDED PARAMETER ESTIMATION AND RECONSTRUCTION ALGORITHM FOR SPARSE DATA

The coded ultrasonic signal itself does not meet the requirements of fri sampling. After preprocessing, the initial coded ultrasonic signal retains the key information parameters of the initial coded signal and meets the requirements of FRI sampling. The processed signal is processed by fri sampling core to get the output signal of sampling core. The sparse data can be obtained by sampling the output signal at equal intervals according to the maximum local innovation rate of the signal. Finally, the key parameters of the signal are reconstructed from the sparse data by using the parameter reconstruction algorithm, and the signal reconstruction is completed, as shown in Figure 3.

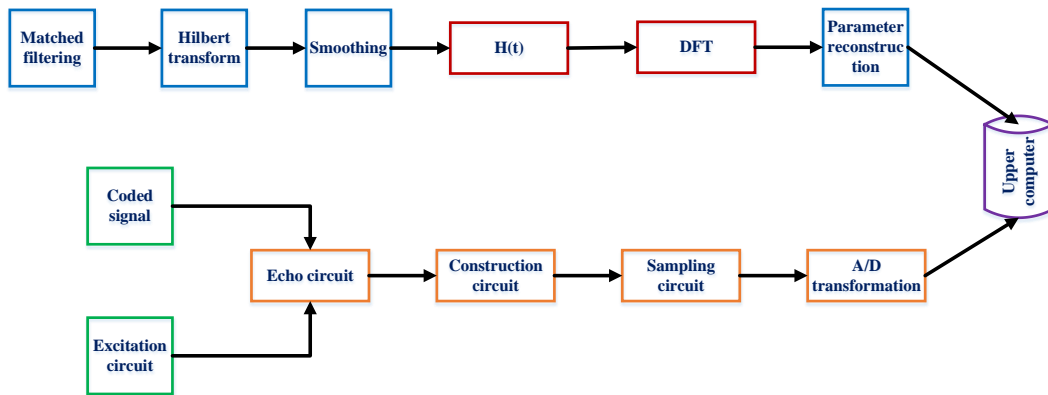


Figure 3: Sparse sampling scheme for coded signals.

The coded ultrasonic signal needs matched filtering, Hilbert transform and smoothing to transform it into FRI sampling pulse stream signal. Matched filter is a common pulse compression method for coded ultrasonic signal. This method is similar to the autocorrelation operation of coded ultrasonic signal. The input-output relationship of matched filter is shown in formula (3).

$$f(t) = \sum_{n=1}^N \alpha_n e^{-\frac{(t-t_n)^2}{\beta n^2}} + \phi(t) \tag{3}$$

$$x_k(t) = \lim_{T \rightarrow \infty} \left[\int_{-l}^l x(s)x(s-t) d_t \right] \tag{4}$$

$v(t)$ in formula (3.2) is the result of matched filtering of coded ultrasonic signal $v(t)$. After Hilbert transform and smoothing, the coded ultrasonic pulse stream signal $\hat{x}(t)$ is obtained, which has the characteristics of FRI signal and can be expressed by limited information degrees of freedom. Figure 4 shows the original time domain waveform of the 3-bit binary coded echo signal, the waveform after pulse compression, and the waveform after Hilbert transform and smoothing. Among them, graph (a), graph (b), graph (c) are three binary 010, 100 and 110, respectively.

It can be seen from Figure 4 that the sidelobe levels of 000, 001, 100 and 111 are high, and the pulse compression effect is not ideal. Compared with 010, 011, 101 and 110, the pulse compression effect is better.

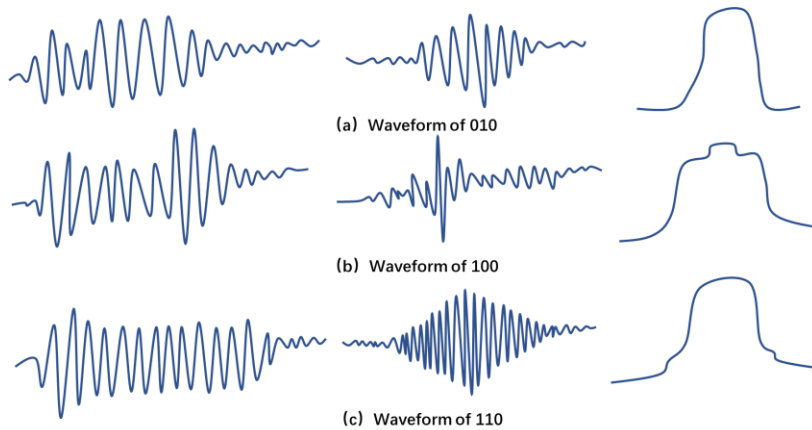


Figure 4: Three waveforms of 3-bit binary coded echo signal.

In the sparse data feature parameter reconstruction algorithm, the Fourier coefficients of the signal have the form of power series weighted sum, and then the spectrum estimation method is used to reconstruct the time delay and amplitude parameters from the Fourier coefficients. The sparse data is obtained by FRI sampling. There are three steps in the process of FRI sampling and reconstruction: 1) the signal $x(t)$ passes through the FRI sampling core $h(t)$, and then the signal $y(t)$ output from the sampling core is obtained; 2) the low rate equal interval sampling of $y(t)$ is performed to obtain the sequence $y[t]$; 3) $Y[n]$ is transformed by discrete Fourier transform to obtain the Fourier coefficient $Y[k]$, and the original signal is reconstructed by the reconstructed parameters to obtain $x(t)$.

The coded ultrasonic signal is transformed into FRI signal after processing:

$$g(t) = \sum_{k=1}^K \beta_n Y[k] e^{-\frac{(s-s_k)^2}{\gamma\sigma}} + \omega_k \varphi(k) \quad (5)$$

Where $a(t)$ is the unit pulse, τ is the period of $x(t)$, and R is the number of pulses in a period. $\{t\} = 1 \dots R$ represents the time delay of the signal, $\{p\} = 1 \dots R$ represents the amplitude of the signal, and satisfies $t \in [0, \tau)$, $P, r \in \mathcal{C}$.

According to Poisson sum theorem, we can get the following results:

$$Y[k] = \sum_{k \in \mathbb{Z}} Y[k] e^{i \frac{2\pi k t}{\sigma}} \quad (6)$$

The Fourier coefficient $Y[m]$ of $y(t)$ can be obtained as follows:

$$Y[m] = \frac{1}{2\pi} \sum_{s=1}^S P_s e^{-i \frac{2\pi s t}{\gamma n^2}} \quad (7)$$

The time delay parameter of signal $y(t)$ is included in the exponential term of $\mathcal{T}[k]$. The amplitude parameter p is the coefficient of $X[k]$. The signal amplitude and time delay parameter $\{p, r\} = 1 \dots R$ can be calculated by the null filter method of spectrum estimation algorithm.

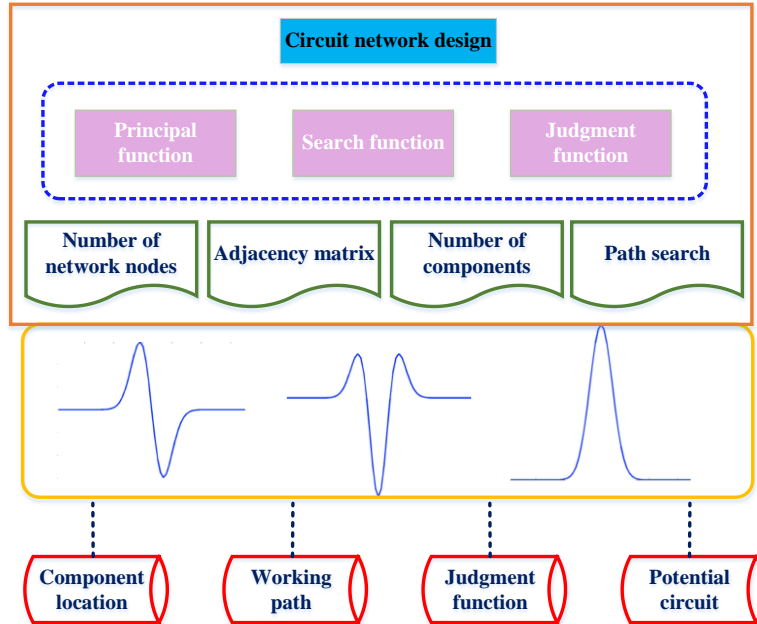


Figure 5: Pulse signal reconstruction parameters.

The parameter reconstruction algorithm of Matlab can use the sparse data obtained from the sparse sampling of the coded ultrasonic signal for parameter reconstruction, as shown in Figure 5. But MATLAB software must run on PC, and the processing speed is not ideal, so this paper implements the algorithm on FPGA to speed up the data processing speed, which is conducive to the improvement of the subsequent imaging speed.

4 EXPERIMENTAL RESULTS AND ANALYSIS

For most practical circuits, the absolute error of the results obtained by combining sparse matrix algorithm and all principal element elimination method is generally less than 10^{-7} . The results show that the number of people to fill is between 0 and $1.8n$, the amount of modification is between $1n$ and $6N$, and the storage capacity is between $23n$ and $28n$. The time of solving equations is more than 25% less than that of sparse principal element in-situ elimination method. Tables 1 and 2 summarize some of the results obtained by the synthetic sparse matrix algorithm. The data in Table 1 shows that for these examples, the amount of filling is not more than $2n$, and the number of modification operations is not more than $6N$. Therefore, the actual storage capacity is less than $28n$. In Table 2, four circuit examples are given to illustrate the calculation amount and percentage of some main steps of the algorithm. The data in Table 2 show that: in the transient analysis of nonlinear circuits, the times of reselection of principal component and symbol decomposition are generally less than 1%, and the times of resolution only changing the right end vector is about 45%. Therefore, the synthetic sparse matrix algorithm can save a lot of machine time.

Order	The number of nonzero elements	Filling quantity	Modification times
18	72	5	27

25	85	3	28
47	232	54	272
91	427	156	478

Table 1: Filling quantity and modification times.

Circuit	Total computation	Symbol decomposition		Numerical decomposition		Positive elimination and Anaphora	
		Calculation amount	Percentage	Calculation amount	Percentage	Calculation amount	Percentage
Circuit 1	144	16	0.12	54	0.38	76	0.53
Circuit 2	185	28	0.16	72	0.39	86	0.48
Circuit 3	831	136	0.17	421	0.52	283	0.35
Circuit 4	1588	240	0.16	768	0.49	582	0.38

Table 2: Analysis of the computational complexity of the algorithm.

It can be seen from the test results that according to the FRI sparse sampling framework, the defects can be effectively detected after the high-order moments of the actual pipeline defect coding ultrasonic testing signal are extracted by the designed FRI sampling preprocessing circuit. At the same time, it is constructed as an ultrasonic pulse stream signal with FRI characteristics, and then the time domain preprocessing of coded ultrasonic signal is realized before A / D sampling through the FRI sampling core circuit developed in the laboratory. The sparse sampling rate is set by the maximum local innovation rate of high-order moment pulse flow, which is much lower than the conventional Nyquist sampling frequency, and the sparse sampling of coded ultrasonic signal is realized. The sparse sampling data is transmitted to the host computer, and the key information parameters representing the detection results are calculated from the sparse sampling data by the method of zeroing filter in the parameter estimation algorithm.

The performance of the coded ultrasonic FRI sampling system is analyzed from several aspects

(1) The sampling rate is related to the amount of data collected. According to the conventional sampling theorem, the A / D sampling rate of coded ultrasonic testing signal used in the test should not be lower than 12.8mhz. In the previous comparative experiment, A / D sampling is carried out at a sampling rate of 20MHz, and the data file size in a sampling period is about 209kb. In the FRI sparse sampling experiment of pipeline defect coding ultrasonic testing signal, the sparse sampling rate is set as 1.2MHz according to the maximum local innovation rate of high-order moment pulse flow. The sparse sampling rate is only about 10% of the conventional sampling rate. The size of sparse sampling data file in a sampling period is about 24Kb, and the amount of data collected is reduced by more than 80%.

(2) Parameter estimation accuracy. The estimation accuracy of four kinds of parameters of 4-bit binary frequency coded ultrasonic signal, i.e., initial wave amplitude ΔP_1 , initial wave delay ΔtH_1 , echo amplitude ΔPH_1 and echo time delay ΔtH_1 , are shown in the form of chart. If there are multiple echoes in one detection, the maximum error value is taken as shown in Figure 6. There are 16 groups of coding forms from "0000" to "1111". The error calculation formula of parameter estimation result is as follows:

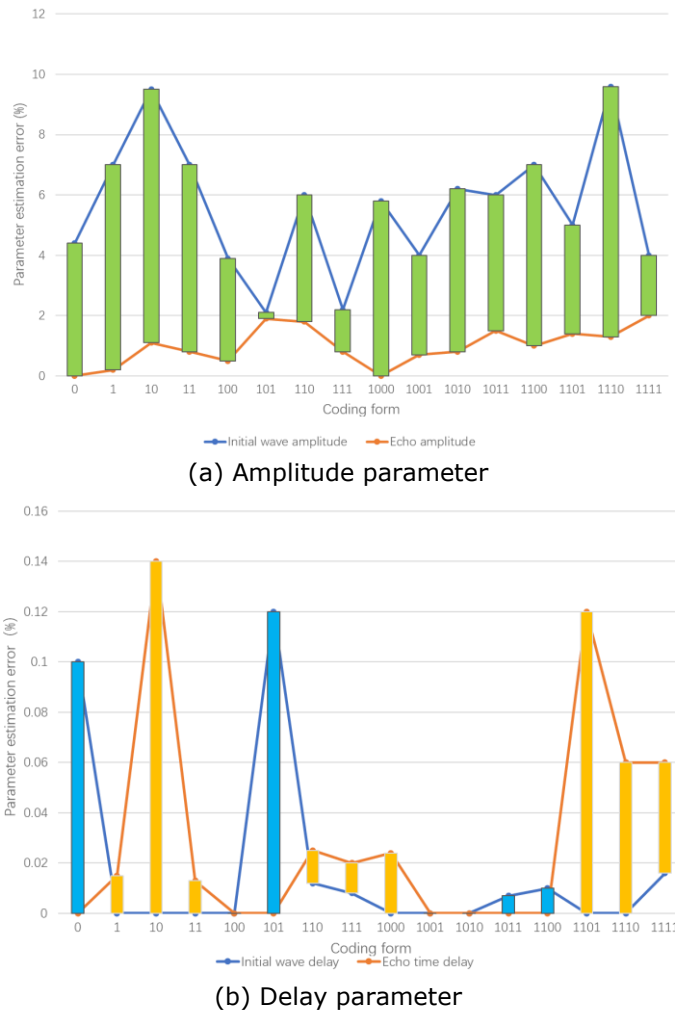


Figure 6: Parameter estimation of FRI sparse sampling data for coded signals.

It can be seen from the chart that the estimation error of time delay parameter is small, and the maximum error is not more than 0.14%, which can accurately reconstruct the arrival time of ultrasonic pulse flow. The estimation error of echo amplitude parameters is slightly larger, the maximum error is close to 10%. This may be due to the saturation phenomenon in the process of building ultrasonic pulse flow, which is caused by adjusting the circuit gain to make the signal amplitude close to the rated value of the simulator.

(3) Range resolution and echo signal-to-noise ratio. The actual echo oscillation duration of coded ultrasonic testing signal for pipeline defects is about $3 \mu\text{s}$. If the longitudinal wave velocity in steel is about 5900 M/s , the range resolution of coded ultrasonic signal is only about 18 mm. After extracting the high-order moments, the echo pulse width of the high-order moment pulse stream is about 100ns, then the range resolution is about 0.5mm, and the range resolution is improved by more than 300%.

The echo amplitude of the original coded ultrasonic detection signal is compared with that of the high-order moment pulse flow. If there are multiple echoes in one detection, the minimum amplitude is taken, as shown in Figure 7. When the noise level is fixed, the extraction of the high-

order moment pulse flow can improve the echo amplitude of the detection signal and the signal-to-noise ratio.

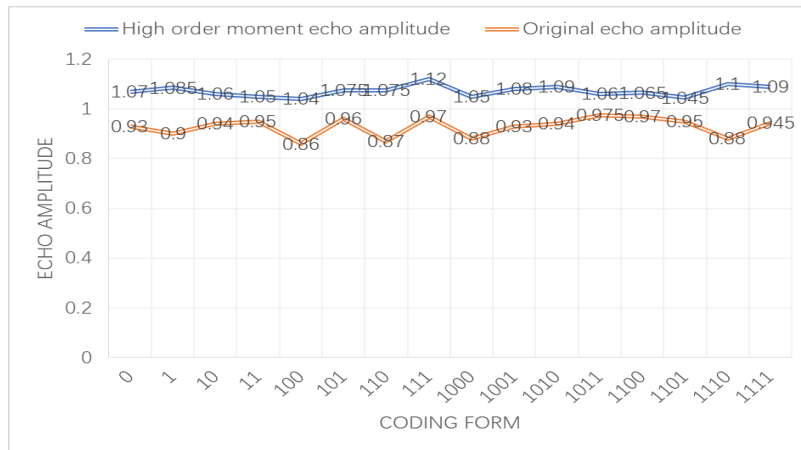


Figure 7: Echo amplitude comparison.

5 CONCLUSIONS

In circuit analysis, power electronic converters often produce a large number of harmonic components due to their switching characteristics, which will pollute the power grid for the grid side and reduce the operation efficiency of the load for the load side. Therefore, it is of great significance for the computer-aided harmonic analysis of power electronic converters. This paper first introduces the idea of computer-aided circuit analysis method, then establishes the adjacency matrix model of power electronic converter according to graph theory, and discusses the algorithm implementation of path search and path judgment based on depth first search algorithm. Finally, the potential circuit computer-aided analysis software is developed, and the potential circuit of Buck resonant switched capacitor converter is analyzed by using the software. The software running results verify the correctness of the proposed method. Through the computer-aided circuit analysis program, we can use the program to calculate the parameters of complex circuit diagram more quickly and conveniently, which greatly improves the efficiency of solving electronic circuit problems.

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