Analysis of signal integrity improvement in high frequency digital circuits

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Abstract. The purpose of this paper is to explore the characteristics and applications of high frequency digital signals. With the continuous development of modern communication technology, high frequency digital signals have become an essential and important part of the digital communication field. Firstly, this paper introduces the basic concepts of high-frequency digital signals, including the sampling, quantization and coding processes of digital signals. Secondly, this paper analyses the main characteristics of high frequency digital signals, including frequency, bandwidth, code rate and power. The transmission characteristics of high frequency digital signals are further discussed, including transmission distance, transmission bit error rate and interference immunity. Finally, the paper discusses the applications of high-frequency digital signals in the fields of communication, radar, measurement and control, and provides an outlook on the future development trend of high-frequency digital signal technology. In conclusion, the rapid development of high-frequency digital signals has supported the progress of digital communication technology, and also brought about great changes to the development of human society.

Keywords: high frequency digital signals, sampling, quantization, coding, frequency, bandwidth, code rate, power, transmission distance, bit error rate, interference immunity, communication, radar, measurement and control, future development trend.

1. Introduction

At present, the stability of high frequency digital signals is more often ensured by the Fourier transform, wavelet transform, Laplace transform, Wiener filter and other ways and means, each of the three methods has its own advantages and disadvantages[1]. The advantage of the Fourier transform is that it can interpret signals that cannot be interpreted in the time domain by convolution, but the Fourier transform is only applicable to the analysis of smooth signals, but not to the analysis of instantaneous frequency changes with time and the analysis of excitation signals. Signals that cannot be analysed by the Fourier transform can be made Fourier-transformable by adding an attenuation factor through Laplacian variation. The addition of a finite time decay in the time domain as a reference gives the wavelet signal the ability to be analysed in the time domain, but the coefficients of the wavelet transform still deviate from the real signal, making the signal-to-noise ratio too low. The Wiener filter is the most widely used, as it can be used for continuous, discrete or even random processes, and can be used for vector and scalar signals, as well as for explicit solutions of the filter transfer function. The need to optimise Fourier variation or optimise the integrity of existing high frequency signals has arisen [2].

In the last five years, much research has been carried out on how to improve the signal-to-noise ratio by means of independent vector analysis and the addition of matched impedances. The independent vector approach is based on the analysis of the most common noises and their integration to allow for subsequent noise suppression [1]. Following the logic of analysing the most dominant noise and then masking it, the independent vector analysis method was used to verify the feasibility of the independent vector to analyse and denoise the noise affecting high frequency signal transmission in high frequency networks and to improve the SNR [3].

There are also methods that use adaptive equalisation noise reduction methods to mask out noise in high frequency digital circuits by abstracting it out [4]. It follows the method of first constructing a signal transfer model of a high-frequency digital circuit, and then constructing the spectrum of the high-frequency digital signal in space by segmenting it and then constructing a model of the time-delay-Doppler estimation method, and verifies that the adaptive equalisation noise reduction method can be used to separate the spatial interference noise in the digital circuit and mask it to obtain a filter.

There are also studies of digital signal processing through purely mathematical PM vector coding algorithms. It improves the efficiency of operations to some extent by combining the advantages of PM algorithms and vector coding on a high-dimensional, integer basis [6]. It follows the logic underlying the general application of the Fourier transform in the discrete case, makes optimisations based on the PM algorithm and vector algorithm, etc. for address selection for co-location calculations that are not well solved in large bases, and uses a comparison of the data operation times for various coding algorithms to verify the rationality of the PM and vector algorithms simultaneously.

Wiener filtering is also simultaneously able to be used for information image enhancement in specific, for example, arrays [5]. Its minimum contrast of the sonar array signal according to the minimum mean contrast criterion given no several times response beam formation processing, using the method of Wiener filtering, today's for each direction of the constrained signal for the processing of single-channel Wiener filtering, verified that can be through the array signal for Wiener filtering processing, thus can effectively reduce the impact of noise, enhance the image accuracy of the sonar, and thus help to improve the accuracy of detection.

2. Principles

Reflected noise is generated by impedance mismatches, while crosstalk noise is generated by the interaction of redundant energy from signal interactions. The presence of these two types of noise can greatly insidiously affect the quality of the signal and therefore requires noise processing.

2.1. Principle of independent vectors

Treatment using independent vectors is accomplished by introducing independent volume analysis to the method design. In Fig.1, an improved signal-to-noise ratio is achieved by shielding the most common types of noise found in noise, namely reflected noise and crosstalk noise. A commonly used method is the independent component method, which works by forming an M-component column vector X(t) from M observable signals, and by introducing discrete moments, turning the M signals into the product of an N-dimensional vector S(t) and a mixing matrix A. Thus it is possible to estimate X(t) out of S by another matrix unmixing array B, but from time to time a full approximation cannot be achieved. Instead, the optimisation is done by first bringing the mean value in the mixed signal into the individual mixed signals so that x(t) is n versions of a random vector, and finally whitening the mixed signals, separating the noise signal from the source signal independently. And later, multiple independent vectors were extracted by the FastICA algorithm proposed by Hyvärinen et al. from the University of Helsinki, Finland [7].

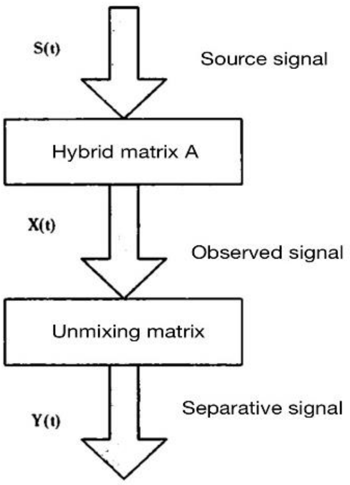

Fig. 1. Flow chart of the independent vector method

2.2. Principle of adaptive methods

The signal transfer model of a digital electronic circuit can be constructed by BPSK modulation to achieve multi-path interference separation of the digital electronic circuit to obtain an acceptance model of the signal. After the model is obtained, a Time Reserve Mirror (TRM) technique is used to reorganise the interference harmonics [8]. After the reorganisation, the signal can be adaptively focused by using the spatial spectrum beamforming method for the adaptive focusing of spatially interfering harmonic signals in digital electronic circuits. Adaptive equalisation noise reduction is then used to separate the noise of the spatially interfering signals in the digital electronic circuit, construct a filtering model and obtain the filtering and modulation results of the channel. And then adaptive matching filtering, based on the knowledgeable focusing process of the spatial interference harmonic signal in the digital electronic circuit using the spatial spectrum beam forming method, the noise separation of the spatial interference signal in the digital electronic circuit is carried out, the channel expansion of the digital electronic circuit is obtained multi-path component, and the tiling focusing is carried out using the beam forming method to obtain the convolution output of the spatial interference harmonic signal of the electronic circuit. The noise interference term will be actively suppressed by adaptive processing to improve the anti-interference capability of the circuit space.

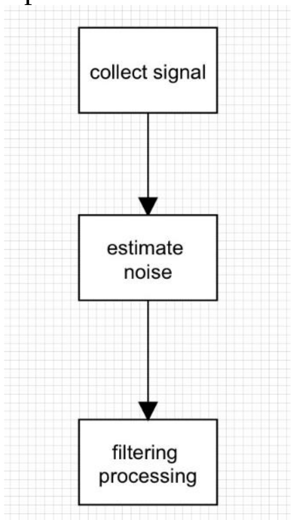

Fig. 2 Flow chart of the reduction methods

2.3. Principle of Wiener filtering

This can also be done by optimising the Wiener filter. The ordinary Wiener filtered signal can be calculated by convolution theorem, in the noise-containing signal by Fourier inversion. As the signal-to-noise ratio increases, the Wiener filter is used to estimate the signal from the noise-containing signal signal close to the original signal. The Wiener filtering of the array signal, on the other hand, allows the calculation of weights to filter the array element signal. A minimum variance distortion-free (MVDR) beamformer is utilised [9]. By estimating the covariance matrix of the array signal, adaptive weights are constructed that vary with time, and the array can be processed through the constraint relations of the weights to perform time-domain signal Wiener filtering on each beam. And then the spatial sub-array smoothing method [10] is used for its processing. And then it can be found that the sonar image after the array Wiener filtering process will be clearer than the image after Wiener filtering process.

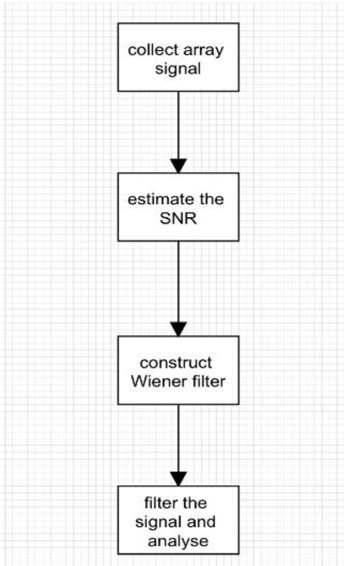

Fig. 3 Flow chart of the Wiener filtering

3. Cross-sectional comparison

The three methods also have their own advantages and disadvantages.

3.1. Analysis of independent vector

The independent vector method automatically separates multiple mixed signals, allowing each signal to be processed independently. In a low signal-to-noise environment, independent component analysis can improve the quality and reliability of the signal by separating and reducing the noise of multiple signals. The main steps of the independent vector method include pre-processing, estimation of the mixing matrix, signal separation and post-processing. Prior to the independent vector method processing, some pre-processing of the data is required to improve the processing effect. For example, the data can be normalised, the direct current component removed and the sampling rate reduced to eliminate redundant information in the signal, reduce the computational complexity and improve the processing efficiency. After that the estimation of the mixing matrix has to be performed

Before signal separation can be carried out, the mixing matrix A needs to be estimated. the mixing matrix is generally unknown, so it needs to be estimated using a blind source separation method. Signal separation can be carried out after the mixing matrix has been obtained. The core idea of the independent vector method is to use the independence of the signal for signal separation, and therefore the independence of the signal needs to be evaluated. Commonly used independence measures include Kurtosis, Shapiro-Wilk Test, Pearson Correlation, etc. And after the signal separation, some post-processing operations may be required to further improve the quality and reliability of the signal. In the latest research, the independent vector method goes through the process of collecting the mixed signal, de-meaning and whitening, setting the number of independent components to be extracted from the mixed signal, followed by constant iterations of orthogonalisation and normalisation until the convergence conditions can be satisfied. In the de-averaging, the independent component analysis is used to remove the noise from the high frequency signal, so that each component of the mixed signal can satisfy the mean value of zero. However, there will still be a lot of noise mixed with the signal in the mixed signal, so the signal has to be whitened to make it separate and independent. Therefore, ica processing techniques can be used to separate the signal by maximising the non-Gaussianity of the independent components by treating the mixed signal as a linear combination of the independent components through a blind source separation technique based on gradient descent. The fastica technique that can be applied to this can be done quickly and with much less computational effort. Figure 4 below shows the waveform of the mixed signal. Figure 5 shows the image after processing using this method.

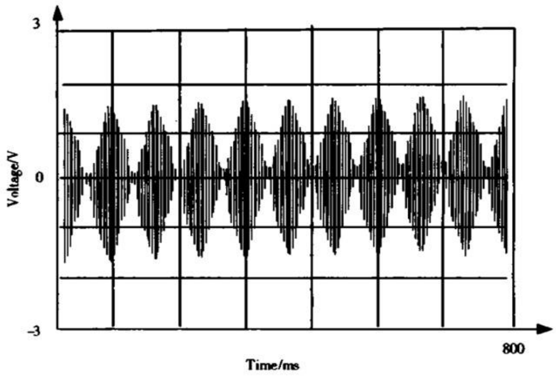

Fig. 4 Waveform of the mixed signal

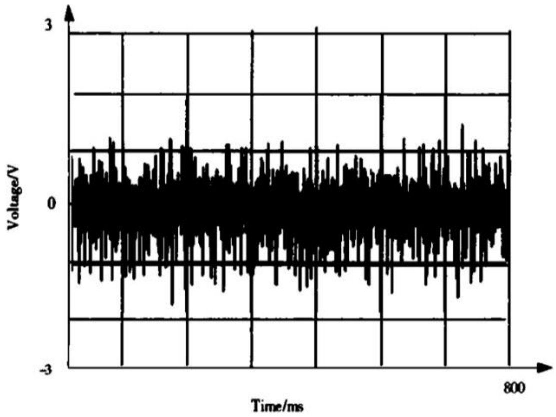

Fig. 5 Waveform after using methods.

3.2. Analysis of adaptive methods

The basic principle of adaptive processing of noise disturbances is to filter and predict the input signal by creating an adaptive filter. The input to the adaptive filter is the original signal and the noise, and the output is the filtered signal. The parameters of the adaptive filter are adaptively adjusted to the characteristics of the input signal and the noise in order to make the output signal as close as possible to the original signal.

Adaptive filters are usually parameterised using either the Least Mean Square (LMS) algorithm or the Least Mean Square (RLS) algorithm, which is an iterative algorithm that calculates a parameter update for the filter based on the current error signal and the input signal, and then uses that update to update the filter parameters. the RLS algorithm is a recursive algorithm that recursively calculates the covariance matrix and The RLS algorithm is a recursive algorithm that updates the filter parameters by recursively computing the covariance matrix and the weight vector. Both algorithms are capable of adaptive tuning, but the RLS algorithm is more computationally intensive and is suitable for scenarios that do not require high real-time performance. The general filtering steps will be split into three. Forward filtering is a common adaptive filtering method that takes the original signal and noise and feeds them into the adaptive filter separately to obtain two output signals, which are then subtracted to obtain the filtered signal. Backward filtering is a more complex adaptive filtering method, which subtracts the output signal of the adaptive filter from the original signal to get the noise signal, then inputs the noise signal into the adaptive filter to get the predicted signal of the noise, and finally subtracts the predicted signal from the original signal to get the filtered signal. Bidirectional filtering is an adaptive filtering method that combines forward and backward filtering, which takes into account both the characteristics of the signal and the noise, and also enables real-time processing. In contrast, newer techniques are now available that separate the noise from the interfering wave signal using digital integrated wave techniques [11]. Adaptive filtering can now also be performed by the spatial spectrum beamforming method by calculating the multi-path component of each channel expansion and thus adaptively matching the spatial interference harmonic noise signal using the fractional order Fourier transform, thus achieving the requirements of the spatial spectrum beamforming method for the weighting and phase adaptive adjustment of the signal using the adaptive processing noise method. The figure 6 below shows the SNR comparison image and BER comparison after processing. It can be seen that the adaptive system has a very significant improvement in the SNR of the signal.

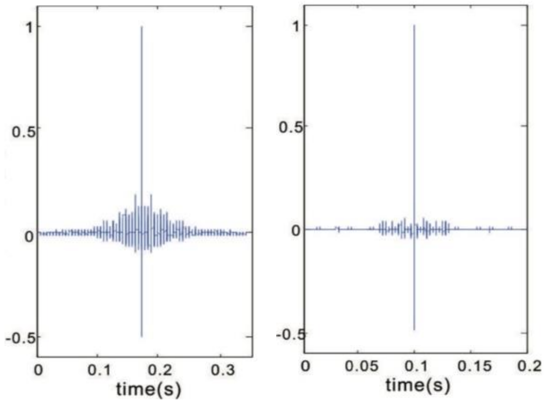

Fig. 6 Before and after using the methods

3.3. Analysis of Wiener filtering

The Wiener filtering implementation of the array signal is divided into single channel Wiener filtering and active sonar array signal Wiener filtering. Single-channel Wiener filtering is achieved by first quantizing the sonar-accepted signal model into a model that passes through the underwater target echo signal and the interference noise signal [5]. The underwater target echo signal and the interference noise signal are uncorrelated, so it can be simply assumed that the input signal X(n) is composed of the superposition of the target echo signal S(n) and the additive interference noise signal N(n), while their Fourier transform is applied to the frequency domain to obtain their spectral representation. The statistical properties of the signal and noise are used to establish the frequency response of the Wiener filter and the frequency response is applied to the spectral representation of the input signal to obtain the spectral representation of the input signal. Finally, the spectral representation of the input signal is transformed into a time domain representation to derive the spectrum of the input signal. With the addition of a sonar array the Wiener filter can be used for sonar signal processing, which can utilise the signals received by multiple sensors in the sonar array to achieve enhancement of the target signal and suppression of noise by weighting and phase adjustment of the signal. The algorithm is based on the Wiener filter, which analyses the autocorrelation and cross-correlation matrices of the signals to derive the optimal weighting coefficients and phase differences, thereby improving the detection and identification capabilities of the sonar system. Active sonar array wiener filtering has been widely used in the fields of ocean exploration, underwater communication and sonar imaging. The spectral signal of the signal received through the M element count sonar array is listed similar to the model of signal processing of a single channel, by adding the components of angular frequency can deform $x(\theta),s(\theta),n(\theta)$ to $X(\omega,\theta),S(\omega,\theta),N(\omega,\theta)$, by this way the number of single channel Wiener filter is changed into a vector, phase adjustment and weighting can be performed, thus allowing a more accurate determination of target location, after which a minimised mean squared error estimate can be made. Once the array signal has been Wiener filtered, it is also possible to combine Wiener filtering with commonly used beamforming methods: the addition of a minimum variance distortion-free (MVDR) beamformer, for example, makes it possible to reduce the effect of air-domain interference.

The Wiener filtering of the active sonar array signal can therefore be understood as active MVDR beamforming and post Wiener filtering. This figure is the conventional beam image of a rigid sphere in the high signal-to-noise ratio case. Figure 7 shows the image after MVDR beam processing, which shows the improved local contrast and also the improved directional resolution of the target echoes.

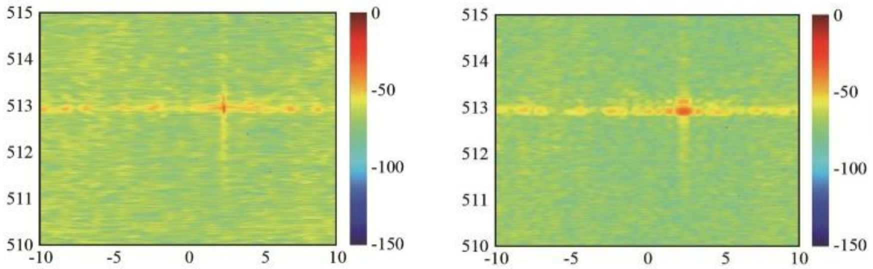

Fig. 7 Before and after using the Wiener filtering

3.4. Comprison

Table 1. The Comparison

	Independent vector method	Adaptive method	Wiener filtering under array signals
Applicable scenarios	Non-Gaussian signals	Various types	All types
Signal processing response	It's computationally intensive and can be slow	It's computationally intensive and can be slow	More calculations and more data to process
Accuracy of signal processing	Best for non-Gaussian signals, depending on the degree of adjustment to the initial value	Best for handling adaptive signals that need to change	Best when statistical properties need to be modelled
Advantages	Wide range of application, can extract signal independent components	Adaptive adjustment of filtering parameters according to signal characteristics	Possibility to model the statistical properties of signals
Disadvantages	Very sensitive to initial values and requires large computational resources	The statistical characteristics of the signal need to meet the requirements of	The accuracy of modelling affects the accuracy of the signal, the

The independent vector method, the adaptive method and Wiener filtering under array signals are all among the common methods used in digital signal processing, and they have different advantages and disadvantages when processing high frequency digital signals.

Independent vector method: The independent vector method performs very well when processing high frequency digital signals, as it can accurately estimate the direction and angle of arrival of the signal, resulting in highly accurate signal separation and denoising. However, the independent vector method requires an a priori setting of the statistical properties of the signal and can be computationally complex for processing signals of high dimensionality.

Adaptive method: The adaptive method is a real-time adaptive digital signal processing method that allows signal separation and denoising without knowing the a priori information of the signal. When processing high frequency digital signals, the adaptive method can dynamically adjust the filter parameters to adapt to different signal characteristics, and therefore has good robustness and adaptability. However, the adaptive method requires a high signal-to-noise ratio for the signal and is prone to algorithm failure in the presence of signal coherence.

Wiener filtering under array signals: Wiener filtering under array signals is a method based on array signal processing that effectively removes noise and interference from the signal. When processing high-frequency digital signals, Wiener filtering under arrays can be used to sample the signal using multiple receivers in the array, thus achieving highly accurate signal separation and denoising. However, Wiener filtering under array signals has certain requirements on the statistical characteristics and spatial distribution of the signal, and requires fine arrangement and calibration of the receivers in the array.

Whereas in the applicable scenarios the independent vector method is often used in a variety of scenarios, the adaptive method is suitable for signal processing where matched filters are available, and array Wiener filtering is mostly used for sonar and radar related detection.

4. Conclusion

When processing high-frequency digital signals, the independent vector method can accurately handle non-Gaussian signals and can transform high-dimensional vectors into vectors for computation; however, the independent vector method requires a priori settings on the statistical properties of the signal. The adaptive method is a real-time adaptive digital signal processing method that can perform signal separation and denoising without knowing the a priori information of the signal. When processing high frequency digital signals, Wiener filtering under an array signal can be used to sample the signal using multiple receivers in the array, thus achieving high precision signal separation and denoising, and this method can also be used in military applications. For there are a large number of high-frequency digital signal processing technology has a wide range of application areas, including wireless communications, radar signal processing, medical imaging, sound processing, audio and video coding and decoding, control systems and other fields. In addition, high-frequency digital signal processing technology is also widely used in important areas such as national defence, security, transportation and aerospace. In wireless communication, the independent vector method can effectively separate the signals of different users to avoid interference and increase in BER. In wireless communication, the independent vector method can effectively separate the signals of different users to avoid interference and BER increase. Adaptive filters can be used to separate the signals of different users and suppress channel interference, improving communication quality and capacity. The use of Wiener filtering of array signals in radar signal processing can remove multipath and spurious interference from the signal and improve the accuracy and reliability of radar signals. Future research in high-frequency digital signal processing methods will tend to be integrated, bringing together different digital signal processing techniques to form a more complete and efficient digital signal processing system. Research on high-frequency digital signal processing methods will continue to progress and develop in the future, bringing more innovations and breakthroughs in the field of digital communications and information processing.

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