2011 International Conference on Computer Science and Information Technology (ICCSIT 2011)
IPCSIT vol. 51 (2012) © (2012) IACSIT Press, Singapore
DOI: 10.7763/IPCSIT.2012.V51.99

Study on the DQPSK Demodulation Method based on Virtual Radio

Chen yuan[1], Hou Xiao-min[1], Zheng Hai-xin[1] and Wang Yuan-qin[2]

[1]TT&C Research Center, the Academy of Equipment, Command & Technology, Beijing, China

[2]Institute of Ultra-precision Optical & Electric Instrument Engineering, Harbin Institute of Technology, Harbin, China

Abstract. Discrete short time Fourier transformation (DSTFT) is a common and effective method in digital signal processing and analysis. In the most applications, amplitude information of Fourier transform is utilized instead of phase information. An algorithm based on the sliding DSTFT is presented in this paper. The demodulation of DQPSK signal is achieved using phase information in frequency domain. The principle and process of the algorithm are detail researched. The fast recursive algorithm is given under the two window functions, and the bit error rate (BER) is analyzed. The simulation shows that the DSTFT-based algorithm can get high performance, and facilitate software real-time processing. In the Gaussian white noise channel, when BER is 10^{-4}, the performance of the algorithm is nearly 1dB better than the traditional differential detection method.

Keywords: DQPSK; Sliding DSTFT; Recursive algorithm; all-pole window

1. Introduction

DQPSK is the most widely used modulation method in the current data transmission systems, which adopts π/4-differential quadrate phase shift keying (π/4-DQPSK). The signal demodulation method includes coherent and incoherent. Coherent demodulation has good performance, but it is difficult to achieve effectively in parallel on the general-purpose computer platform because of the using of closed-loop approach. Traditional incoherent differential detection system is relatively simple. However, the incoherent demodulation performance has about the deterioration of 2.3dB, comparing to the coherent detection demodulation. Aiming at the performance deterioration issue of the traditional differential detection demodulation, the improved algorithms have been presented [1-3]. The demodulation performance is significantly improved, but the difficulty and complexity of the methods are correspondingly increased.

A DQPSK software demodulation method based on sliding window is presented. The algorithm is completely achieved by software programming. There is only one core algorithm, and no other factors may result in demodulation errors, so the receiving information is used efficiently, and optimal solution is achieved.

This paper will be structured as follows: In Section Ⅱ and Ⅲ, the principle and process of the algorithm are researched; In SectionⅣ, the optimization method for improving demodulation speed is presented, and in Section Ⅴ the measured results are shown and the discussion based on these results is made. A conclusion is summarized in Section Ⅵ.

2. The Phase Calculation Based On Silding Time Window

A sinusoidal signal phase estimation algorithm is presented [4]. Assumption of time-domain discrete signal is

$$S_i = A\sin(\omega T_s i + \varphi) \tag{1}$$

Where A is signal amplitude, ω is signal frequency, T_s is sampling interval, and φ is initial phase.

Observed m consecutive sampling points among the receiving signal

$$S_{i+k} = A\sin[\omega T_s(i+k)+\varphi] \quad k=0,1,\ldots,m-1 \tag{2}$$

The corresponding phase of the middle point among these points is

$$\varphi_i = \varphi + \omega T_s(i+\frac{m-1}{2}) \tag{3}$$

Equation (3) is written as

$$\begin{aligned} S_{i+k} &= A\sin[\varphi_i + \omega T_s(k-\frac{m-1}{2})] \\ &= A\sin(\varphi_i)\times\cos[\omega T_s(k-\frac{m-1}{2})] \\ &+A\cos(\varphi_i)\times\sin[\omega T_s(k-\frac{m-1}{2})] \end{aligned} \tag{4}$$

Then

$$\varphi_i = \arctan[\frac{\sum_{k=0}^{m-1}\cos[\omega T_s(k-\frac{m-1}{2})]\cdot S_{i+k}}{\sum_{k=0}^{m-1}\sin[\omega T_s(k-\frac{m-1}{2})]\cdot S_{i+k}}\cdot\frac{[1-\frac{\sin(m\omega T_s)}{m\sin(\omega T_s)}]}{[1+\frac{\sin(m\omega T_s)}{m\sin(\omega T_s)}]}] \tag{5}$$

Thus we get the signal phase at a time. Based on this, the m sampling points can be considered as a time window, and the sampling points are moved along the time axis backwards one by one. By this way, the signal phases at all sampling times are obtained.

3. Signal Demodulation Based On Silding Window Phase

The sinusoidal signal of a single carrier frequency can be got by DQPSK, which is

$$S(n) = A\cos(2\pi f_c n / f_s + \theta) \tag{6}$$

Where f_c is carrier frequency, f_s is sampling frequency, θ is additional phase: $\theta = \pi/4, 3\pi/4, 5\pi/4, 7\pi/4$.

The phase modulation information is carried by DQPSK signal. The main process of the DQPSK signal demodulation method using sliding window-based phase is: firstly, the modulation signal sequence which has been read is processed according to the previously discussed phase calculation algorithm, and the instantaneous phase sequence that is close to the signal sequence length can be got, then the phase difference between the adjacent symbols is obtained, and optimal decision point can be found by bit synchronization, finally the phase logic relation of DQPSK signal is judged for recovering the binary information.

The demodulation block diagram is shown in Fig. 1.

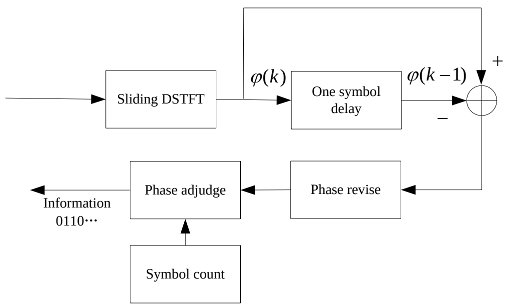


Figure 1.Demodulation block diagram

The demodulation system is composed of phase calculation, phase delay and revise, bit synchronization and phase adjudge.

3.1. Phase calculation

DSTFT is a common and effective method in digital signal processing. In most applications, the amplitude information of DSTFT coefficient is used, but the phase information is ignored. In fact there is an important application of DSTFT, which is to estimate the initial phase of a single frequency sinusoidal signal.

Equation (1) is transformed by L points DSTFT, which equals infinite signal multiplied by the length L of the rectangular window $w(n)$. The signal modified by the window can be expressed as

$$v(n)=\frac{A}{2}w(n)e^{j\varphi_0}e^{j\omega_0 n}+\frac{A}{2}w(n)e^{-j\varphi_0}e^{-j\omega_0 n} \tag{7}$$

According to the charactristic of the Fourier transform, the Fourier transform of $v(n)$ is

$$V(e^{j\omega})=\frac{A}{2}e^{j\varphi_0}W[e^{j(\omega-\omega_0)}]+\frac{A}{2}e^{-j\varphi_0}W[e^{-j(\omega+\omega_0)}] \tag{8}$$

Where $W(e^{j\omega})$ is the Fourier transform of window function.

Equation (8) is shown that DSTFT includes the window function transform in $\pm\omega_0$. Now only the positive frequency is obtained as

$$V'(e^{j\omega})=\frac{A}{2}e^{j\varphi_0}W[e^{j(\omega-\omega_0)}] \tag{9}$$

When $\omega=\omega_0$, $angle\left[V'(e^{j\omega})\right]=\varphi_0$, The phase near the carrier frequency estimated by the Fourier transform is equal to the initial phase of the signal.

3.2. Phase delay and revise

The (k-1)-symbol samples are processed by DSTFT to obtain the first sample phase φ_{k-1} of this symbol. And the k-symbol samples are processed by DSTFT to obtain the first phase φ_k of k-symbol in the same way. According to the characteristics of differential phase modulation signal, the initial phase difference between the two symbols is

$$\Delta\varphi_k=\varphi_k-\varphi_{k-1}=2\pi f_c l/f_s+\Delta\theta_k \tag{10}$$

Where l is the sampling number within one symbol, $\Delta\theta_k=\theta_k-\theta_{k-1}$. So

$$\Delta\theta_k=\Delta\varphi_k-2\pi f_c l/f_s \tag{11}$$

Then $\Delta\theta_k^{'}$ is revised to set range between $0\sim2\pi$ by

$$\Delta\theta_k^{'}=\mathrm{mod}(\Delta\theta_k,2\pi) \tag{12}$$

Table Ⅰ shows the judgment threshold for decision. If the entire length signal is done like this, all the information codes are obtained.

TABLE Ⅰ JUDGEMENT RANGE SET AND DECODING STANDARD

Judgment range	$\Delta\theta_k$ judgment result	Information code a, b
$0\sim\pi/2$	$\pi/4$	0 0
$\pi/2\sim\pi$	$3\pi/4$	0 1
$\pi\sim3\pi/2$	$5\pi/4$	1 1
$3\pi/2\sim2\pi$	$7\pi/4$	1 0

3.3. Bit synchronization

As the window function used for phase calculation has a certain width, there must be some error about critical phase inferred between the symbols. So the phase difference of the optimal sampling points needs to be chosen for judgment. Prerequisite to achieve this step is to find the initial point of the symbol. There are many bit synchronization methods. According to the characteristics of the DQPSK signal, a bit synchronization algorithm based on instantaneous frequency measurement is presented.

For the DQPSK signal, there must be an integer multiple of $\pi/4$ existing near the symbol transform. As here the phase range transition is relatively small and severely affected by the noise, and together with the big error on the phase calculation near the symbol transform, the first point of each symbol is not easy to be accurately located by the phase curve. According to the relationship between frequency and phase, the signal frequency is changed when the signal phase transition occurs. If these frequency-sudden-change points can be searched, the symbol transform can be found either. Firstly, the instantaneous frequency is calculated using the equation given by conference [4]. According to the equation (5), signal frequency f can be obtained by regression and least squares principle.

$$f = 12\sum_{i=1}^{n}[\frac{\varphi_i}{2\pi}(i-\frac{n+1}{2})]\times\frac{1}{n(n^2-1)}\times f_s \tag{13}$$

Where n is the phase point.

After the instantaneous frequency is obtained, the frequency-sudden-change points are found by setting the threshold. The symbol start position is confirmed to achieve the bit synchronization. Finally, the optimal judgment point is selected in the symbol.

4. Optimiztion Algorithm For Improving The Demodulation Speed

Reliability and validity are two important measurement indexes of a communication system. For demodulation, the reliability refers to the BER performance, and the validity refers to the demodulation speed. The DSTFT-based DQPSK signal demodulation system is simple. The entire process course relies primarily on a core software programming algorithm, but the computation of the algorithm is relatively large. Therefore, the algorithm is optimized by designing the fast DSTFT-based implementation to improve the demodulation speed. DSTFT can be efficiently implemented by recursive algorithm such as rectangular window recursive algorithm and all-pole window recursive algorithm.

4.1. Recursive algorithm of rectangular window

Supposed the length of the rectangular window is M, and the length of DSTFT points is M. The center frequency f_c to the spectrum is k_c. The Fourier value $X_1(k_c)$ is the DSTFT calculation value of M sampling data from the beginning of the first sample at the frequency k_c, so

$$X_1(k_c) = x(0) + x(1)e^{-j2\pi k_c/M} + \cdots + x(M-1)e^{-j2(M-1)\pi k_c/M} \tag{14}$$

Rectangular window is moved backward one sample, and the second Fourier value is obtained.

$$X_2(k_c) = x(1) + x(2)e^{-j2\pi k_c/M} + \cdots + x(M)e^{-j2M\pi k_c/M} \tag{15}$$

The only difference between equation (14) and equation (15) is the different data sets, but the exponent of two equations is exactly the same. Equation (15) can be obtained by modifying equation (14) slightly. The relationship is

$$X_2(k_c) = \left[X_1(k_c) - x(0)\right]e^{j2\pi k_c/M} + x(M)e^{-j2\pi k_c} \tag{16}$$

General expression of the above equation is written as

$$X_{n+1}(k_c) = \left[X_n(k_c) - x(n)\right]e^{j2\pi k_c/M} + x(n+M)e^{-j2\pi k_c} \tag{17}$$

Fourier values (complex) of the entire sampling data are obtained by recursive algorithm of equation (16), and the required phase sequence will be got by obtaining the phase of Fourier values. Using recursive demodulation algorithm of rectangle window is just the process of sliding DSTFT implied by recursive computation method. There is no loss of the demodulation performance.

4.2. Recursive algorithm of all-pole window

The demodulation speed is improved significantly by the recursive algorithm of rectangular window. However, the BER performance of the rectangular window is poor. In order to improve the BER performance, all-pole window for recursive computation is researched in this section.

The Z domain expression of p-order all-pole function is

$$W(z) = \frac{1}{(1-\beta z^{-1})^p}, \quad 0 < \beta < 1, p = 2,3,\cdots \tag{18}$$

$W(z)$ has only poles. A proper p can be chosen to make its time window have a single peak and a desired effective width. Supposed discrete time signal is $x(n)$, and the DSTFT window length is M.

$$S_x(M,\omega) = \sum_{n=0}^{M} x(n)w(M-n)e^{-j\omega n} \tag{19}$$

The equation (19) can be expressed by another form.

$$S_x(M,\omega)=\sum_{n=0}^{M}x(M-n)w(n)e^{-j\omega(M-n)}$$
$$=e^{-j\omega M}\sum_{n=0}^{M}x(M-n)w(n)e^{j\omega n} \tag{20}$$

Supposed

$$\tilde{S}_x(M,\omega)=\sum_{n=0}^{M}x(M-n)w(n)e^{j\omega n} \tag{21}$$

Equation (20) can be written as

$$S_x(M,\omega)=e^{-j\omega M}\tilde{S}_x(M,\omega) \tag{22}$$

Supposed $h(M,\omega)=w(n)e^{j\omega n}$, Equation (21) can be written as

$$\tilde{S}_x(M,\omega)=\sum_{m=0}^{M}x(M-n)w(n)e^{j\omega n}=h(M,\omega)*x(M) \tag{23}$$

The Z transform of equation (23) is

$$\tilde{S}_x(z,\omega)=H(z,\omega)X(z) \tag{24}$$

Substituted the all-pole window function into equation (18), we can obtain

$$\tilde{S}_x(z,\omega)=\frac{X(z)}{(1-\beta e^{-j\omega}z^{-1})^p} \tag{25}$$

Supposed the all-pole window order $p=3$, equation (25) is written as

$$\tilde{S}_x(z,\omega)=X(z)+3\beta e^{-j\omega}z^{-1}\tilde{S}_x(z,\omega)$$
$$-3\beta^2e^{-j2\omega}z^{-2}\tilde{S}_x(z,\omega)+\beta^3e^{-j3\omega}z^{-3}\tilde{S}_x(z,\omega) \tag{26}$$

In the time domain, equation (26) is changed as

$$\tilde{S}_x(n,\omega)=x(n)+3\beta e^{-j\omega}\tilde{S}_x(n-1,\omega)$$
$$-3\beta^2e^{-j2\omega}\tilde{S}_x(n-2,\omega)+\beta^3e^{-j3\omega}\tilde{S}_x(n-3,\omega) \tag{27}$$

Based on equation (27), $\tilde{S}_x$ delayed M sampling points can be written as.

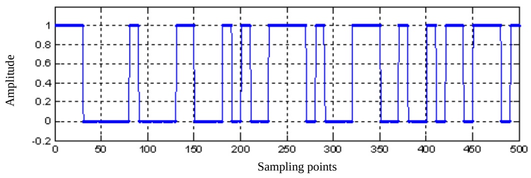
$$S_x(n,\omega)=\tilde{S}_x(n-M,\omega) \tag{28}$$

Using 3-order all-pole window for software demodulation, only the first three Fourier transform value need to be calculated by equation (19). Then the entire signal Fourier values can be obtained by equation (28) without the need for DSTFT operation.

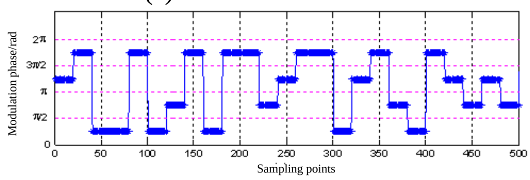
5. Computer Simulaton And Implementation

5.1. Demodulation correctness test

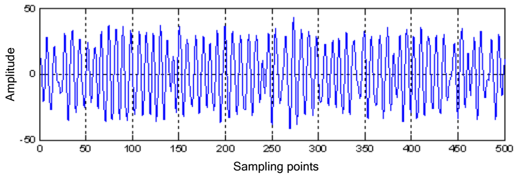
The simulation data is demodulated by DSTFT basic algorithm to verify the algorithm correctness. SNR is set at $E_b/N_0=12.8dB$ $(S/\phi=74.7dBHz)$, and Hanning window is adopted as the analysis window. Signal waveforms are shown at five typical positions a, b, c, d, e by Fig 2.



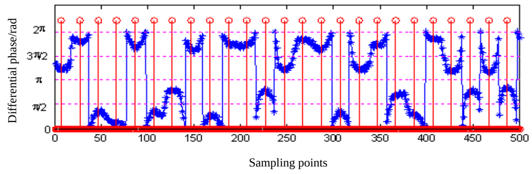
(a) The information code



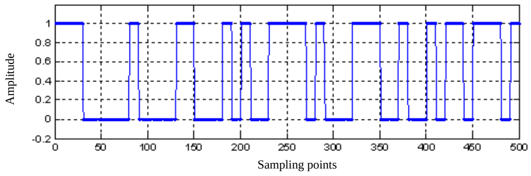
(b) Phase map corresponding to the phase modulation



(c) The signal waveform after adding noise



(d) The differential phase sequence after demodulation

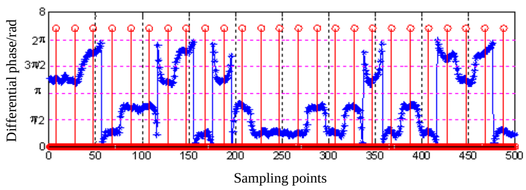


(e) The restored binary code stream after judgment

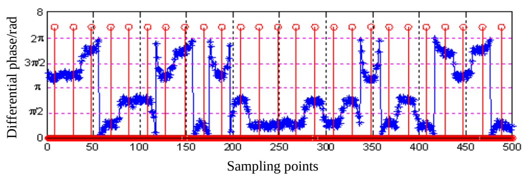
Figure 2.The waveform of modulation and demodulation system

The entire signal processing is show basically by Fig 2. The information code is shown by Fig.2 (a). By serial-parallel conversion and phase map, the binary information is converted into modulated phase information as shown in Fig 2(b). There are four values of $\pi/4$、$3\pi/4$、$5\pi/4$、$7\pi/4$ about modulation phase. Base-band signal including the phase information is modulated by the carrier, and by going through the additive white Gaussian noise channel, the DQPSK signal is shown in Fig 2(c). This is the data required to process by the software demodulation system. Instantaneous phase is calculated by DSTFT, then differential phase sequence is obtained as showing in Fig 2(d), being consistent with the modulation phase sequence as Fig 2(b). Best judgment point positions and judgment range are marked in Fig 2(d). The best judgment points are basically located in the middle of a double-bit code location to prove the correctness of the bit synchronization algorithm. The differential phase value at the best judgment point is demodulated by judgment range $0\sim\frac{\pi}{2}$, $\frac{\pi}{2}\sim\pi$, $\pi\sim\frac{3\pi}{2}$ or $\frac{3\pi}{2}\sim 2\pi$. Finally, the binary stream is exported as showing in Fig 2(e), which is the same as the information code in Fig 2(a) in the case of no BER. Correctness and feasibility of the demodulation and bit synchronization algorithm is verified by waveform comparison of modulation and demodulation systems.

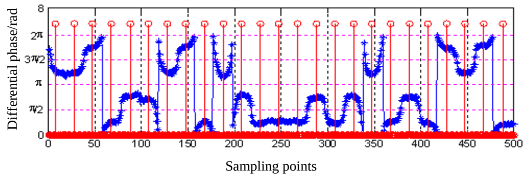
From the above, the DSTFT basic algorithm is adopted for demodulation system, and then the other two implementations of recursive algorithm (rectangle window and all-pole window) are simulated. The differential phase sequence before judgment is shown in Fig3 (b) and (c), comparing with the differential phase of DSTFT basic algorithm in Fig 3(a). It is shown that the differential phase sequences of the two recursive algorithms are the same with the basic algorithm, and rectangle window measured with less precision.



(a) Basic algorithm



(b) Rectangle window



(c) All-pole window

Figure 3.Differential phase demodulation

5.2. EBR performance test

The demodulation algorithm is tested by three groups of simulation data for BER performance. And the result are compared and analyzed.

Firstly, simulation data is demodulated by basic algorithm under different SNR, and the result is counted for demodulation BER. Meanwhile, the result curve is contrasted with the BER curve of the theoretical coherent demodulation and differential demodulation.

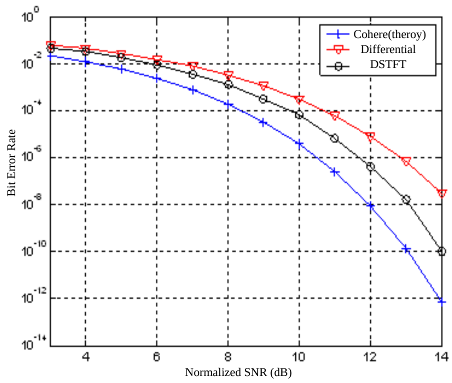


Figure 4.BER performance comparison

Comparing with the theoretical differential demodulation, the performance of the DSTFT-based demodulation method has been greatly improved. In particularly, the BER of DSTFT demodulation is 0.95×10^{-5} when SNR is 11dB. Compared with the theoretical value of coherent demodulation, the deterioration of the novel method is less than 2dB, which can meet the system's requirements.

BER performances of the three implements such as Hanning, Rectangle and all-pole are compared. When BER is $P_e = 10^{-5}$, the values of Normalized SNR E_b / N_0 of three implement are listed in Table Ⅱ.

TABLE Ⅱ ALGORITHM FOR SNR REQUIRED WHEN $P_e = 10^{-5}$

	Hanning	Rectangle	all-pole
$E_b / N_0 (dB)$	10.8	11.1	10.9

The performance of rectangle window recursive algorithm is the worst, and relatively the performance of Hanning window is the best.

Finally, the BER performance of DSTFT is contrasted with traditional demodulation algorithm. And also when $P_e = 10^{-5}$, the values of Normalized SNR E_b / N_0 are listed in Table Ⅲ.

TABLE Ⅲ ALGORITHM FOR SNR REQUIRED WHEN $P_e = 10^{-5}$

	Coherent (theory)	Base-band differential	IF differential	High order differential	DSTFT
E_b / N_0 (dB)	9.6	12.3	12.8	11.2	10.8

Compared with the differential of base-band, IF and high order, there are improvement of 1.5dB, 2dB and 0.4dB respectively on the demodulation performance based on DSTFT. The simulation results show that the performance of DSTFT algorithm is closed to the high order differential demodulation, but DSTFT algorithm has more advantages in high SNR. The system is relatively simple, and the factors affecting the system performance are fewer, so the performance of DSTFT is superior.

6. Summary

The DQPSK signal demodulation algorithm based on the sliding window phase calculation is implied by software programming which belongs to software demodulation. It complies with the software development trend of signal receiving and processing system. The simulation results show that the proposed method can achieve the BER performance of the hardware demodulation requirements. Its performance is superior to the traditional differential demodulation. This method can also be applied to other signal demodulation such as MPSK, DMPSK.

7. Reference

[1] Miller L E，Lee J S.BER Expressions for Differential Detected π/4-DQPSK Modulation. IEEE Journal on Selected Areas in Communication,1983,1(1):74-81.

[2] Divsalar D and Simon M K.Multiple-symbol Differential Detection of MPSK.IEEE Trans. Commun，1990,38(5):300-308.

[3] Mackenthum Jr K M.A Fast Algorithm for Multiple symbol Differential Detection of MPSK.IEEE Trans.Commun,1994,42(2):1471-1474.

[4] Amin M G. A new approach to recursive Fourier transform. Proc. IEEE,1987,75:1537-1538.

[5] Chen W, Kehtarnavaz N, Spencer T W. An efficient recursive algorithm for time-varying Fourier transform. IEEE trans.SP, 1993, 4(7):2488-2490.

[6] Floyd M.Gardner. A BPSK/QPSK timing error detector for sampled receivers. IEEE Transaction on Communications, 1986, 34(5):423-429.

[7] Hamamoto N. Differential detection with IIR filter for improving DPSK detection performance. IEEE Transaction on Communications, 1996:959-966.

[8] B.Li, W.Tong, and P.Ho. Multiple-symbol detection for orthogonal modulation in CDMA system, IEEE Transaction on Vehicular Technology, 2001, 50(1):321–325.