Simulation and Performance Study of MSK System Based on Single Frequency Tracker

ShuaiQianJun
School of Information Engineering
Communication University of China
Beijing, 10024, China
E-mail: sqj@cuc.edu.cn

PanBiyun, Jin Libiao
School of Information &Engineering
Communication University of China
Beijing, 10024，China
e-mail:pan12313128@163.comlibiao@cuc.edu.cn

Abstract—**Adaptive filter technology is mainly used in many different fields such as signal processing, image processing. It is a targeted filter method usually used for removing noise.This paper focuses on the adaptive filtering algorithm in the application of demodulation technology. The least mean square (LMS) algorithm is used to study the demodulation of Minimum Frequency Shift Keying (MSK) signal by adaptive filtering technology and the bit error performance is analyzed. The design uses a Single Frequency Tracker (SFT), which is used to obtain the parameters of the previous time to adjust the current parameters of the momentautomatically. Hence, it can track the frequency, phase and amplitude of the signal. In this paper, we use the single frequency tracker (SFT) to replace the band-pass filterand low-pass filter in the traditional coherent demodulation system.**

Keywords-single:frequency tracker; LMS algorithm; adaptive filter; MSK signal

I. INTRODUCTION

In recent years, with the increasingly widespread application of communication systems, modulation and demodulation of digital signals are becoming more and more important[1]. With the development of large-scale integrated circuit, it is quite common to use software to realize the function of hardware. In this paper, a single frequency tracker (SFT) is used to track the signal, and its core technology is adaptive filtering. In the study of adaptive filtering performance, people found that the adaptive algorithm with coherent demodulation function, it can not only track the signal amplitude, phase， but also frequency change, this is undoubtedly a new demodulation method[2]. Taking MSK signal as an example, the demodulation performance is simulated by computer programming. MSK signal is one of the digital modulation technologies. Digital modulation is the process of converting digital baseband signals into the waveforms that match the characteristics of the transmission channel.The modulation process is to control the amplitude, frequency, and phase of the carrier with the input data.The MSK signal is a 2FSK signal with constant envelope, continuous phase and strictly orthogonal. In addition, MSK signals also have the advantage of minimum bandwidth utilization [3].

With the rapid development of signal processing technology, adaptive signal processing has become an important branch subject of signal and information processing, and has been widely used in fields such as radar, sonar, communication, industrial control, and biology. The traditional filter often cannot get the prior knowledge of the signal, the statistical characteristics of the signal is non-stationary signal changes with time, in this case, the adaptive filter can achieve better filtering performance.The adaptive filtertrack the change of the signal, using the filter parameters obtained a moment ago to adjust automatically the filter parameters in the present, to adapt statistical characteristics of signals and noises, which is unknown or time dependent, to achieve the optimum filtering effect. Adaptive filtering algorithm is an important part of the adaptive filter, the performance of the algorithm directly determines the performance of the filter, the research on adaptive algorithm is one of the most active research topics in the field of adaptive signal processing. The algorithm of adaptive filter is mainly based on various decision conditions. There are usually two kinds of decision conditions: the least mean square error criterion and the least square criterion. Adaptive filtering has two basic algorithms: least mean square (LMS) algorithm and the recursive least squares (RLS) algorithm. Least mean square (LMS) algorithm is a kind of algorithm that is easy to implement, stable and widely used [4].The LMS algorithm tries to make the output signal $y(n)$ close to the ideal signal $d(n)$. The difference between the ideal signal $d(n)$ and the filter output $y(n)$ is $e(n)$, and the expected value of $e(n)$ is minimum, the weight coefficients are modified according to this criterion. Recursive least squares (RLS) algorithm is based on the least mean square error algorithm [5]. The difference is that the RLS algorithm uses the average of time, so the resulting optimal filter depends on the number of samples used to compute the average, In this paper, the least mean square (LMS) adaptive filtering algorithm is adopted, because the LMS algorithm is simple in structure, small in computation, robust and easy to implement.Especially, this algorithm is very practical, it is the first algorithm derived by the statistical analysis method,so it is widely used in system identification, noise elimination, line enhancement, etc.

According to the requirements of the research, simulation and performance study of MSK system based on single frequency tracker can be divided into the following steps: binary baseband signal, differential encoding, MSK modulation and single frequency tracking, median filtering, comparison judgment ， sampling judgment. Finally, the error rate analysis of the system is carried out to observe the anti-noise performance of the system.

978-1-5090-3025-5/17/$31.00 ©2017 IEEE

II. SYSTEM MODEL AND ALGORITHM ANALYSIS

The overall block diagram of the MSK modulation anddemodulation system based on a single frequency tracker is shown in Figure 1 [6]. It mainly consists of two parts: MSK modulation and adaptive demodulation. Firstly, the binary baseband signal is generated, and then the differential encoding is performed by XOR operation. The advantage of using differential encoding is that the current received error symbol does not affect the reception of subsequent symbols, so thaterrors do not last. After modulating the MSK signal according to the modulation principle, the MSK signal is put into a channel containing Gauss white noise [7], and then the adaptive demodulation is performed.Adaptive demodulation includes single frequency tracker, absolute value, median filter, sampling decision and so on. Among them, the adaptive algorithm used in adaptive demodulation is LMS algorithm. The noise immunity of the system is observed by varying the signal to noise ratio.

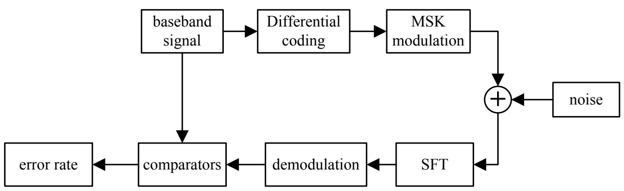

Fig.1Overall block diagram of the system

A. MSK signal modulation

MSK modulation is one of the digital modulation techniques [8]. Digital modulation is the process of converting digital signals into waveforms that match the characteristics of the channel. The modulation is the process that input data to control the amplitude, frequency and phase of the carrier.MSK is called minimum shift keying, and it is an improved type of frequency shift keying (FSK) ， Here "minimum" means that an orthogonal signal can be obtained with a minimum modulation index (i.e., 0.5), which transmits higher bit rates than the PSK [9].

The MSK signal can be represented by two orthogonal components:

$$e_k(t) = p_k \cos\frac{\pi t}{2T_B}\cos\omega_c t - q_k \sin\omega_c t \qquad (1)$$

$$kT_B \le t \le (k+1)T_B$$

The compiling software used in the experiment is Microsoft Visual C++ 6 (VC++). Install the plug-in easyX to observe the simulated waveform.

The input is a binary baseband signal, and the relative code is generated by differential encoding of the baseband signal. The relative code is divided into p_k branches and q_k branches after serial / parallel conversion.The simulated MSK waveform is shown in Figure 2.

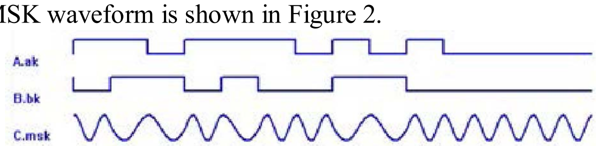

Fig.2 MSK signal modulation

B. The principle of adaptive filter

A single frequency tracker is used to track the signal, and the core technique is adaptive filtering. The reason that using adaptive filtering can achieve the best filtering effect is that the current parameters of filter can be adjusted automatically according to the previous parameters; therefore the adaptive filtering effect is better than the ordinary multi filter effect [10].

The schematic diagram of the adaptive filter is shown in figure 3. It can be seen from the figure, $x(n)$ is an input signal of the adaptive filter, $d(n)$ is the another input signal, $y(n)$ is output signal, another output is error signal $e(n)$, $e(n)=d(n)-y(n)$. The system adjusts the parameters of the filter according to some adaptive algorithm.At last,the input signal $x(n)$ can be tracked by $y(n)$,meanwhile the minimum $e(n)$ can be achieved.

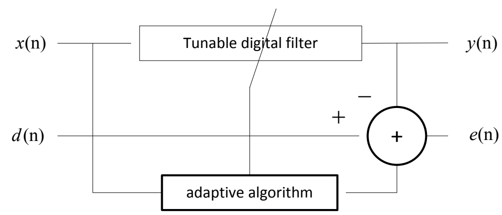

Fig.3 Principle diagram of adaptive filter

C. The Principle of single frequency tracker based on LMS algorithm

Figure 4 shows the principle of an adaptive single frequency tracker based on the LMS algorithm. Two automatically weighted, single frequency adaptive filters can be seen in the diagram. In addition to the original signal, the input has the reference input, and the reference input needs to be guaranteed the same frequency as the original signal. The output also has two outputs.oneis the error output $e(n)$, and the other is $y(n)$ [11].

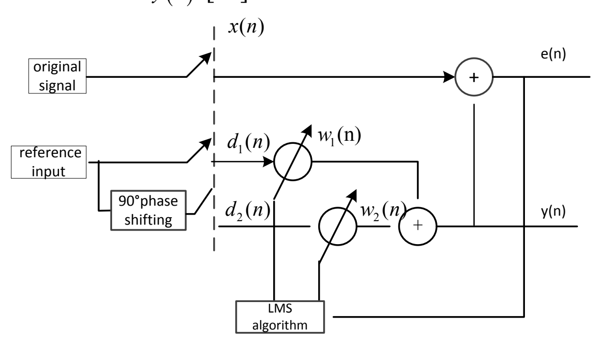

Fig.4 Schematic diagram of adaptive single frequency tracker

The original input is $x(n)$ in the figure, and the reference input is $d_1(n)$ and $d_2(n)$. Suppose the original input is $A\cos(\omega_0 t+\varphi)$， then

$$\begin{cases} d_1(n) = A\cos(\omega_0 nT+\phi) \\ d_2(n) = A\sin(\omega_0 nT+\phi) \end{cases} \qquad (2)$$

The reference input $d_1(n)$ is the same frequency as the original input, and the reference input $d_2(n)$ has a 90^0 phase difference with the original input. The minimum error sum of square is obtained by the original and reference input, and the filter weights ω_1 and ω_2 are controlled by it. $y(n)$ is the filtered output.

$$y(n) = w_1(n)d_1(n) + w_2(n)d_2(n) \quad (3)$$

In adaptive filtering, LMS is often used because it is simple and efficient. As an efficient recursive method, the principle of the LMS algorithm is steepest descent method, in order to minimize the mean square variance between the original input and the reference input, so as to obtain the optimal weight vector [12]. The iterative weighting formula of LMS algorithm is:

$$w(n+1) = w(n) + 2\mu e(n) d(n) \quad (4)$$

μ is the factor controlling the rate of convergence.

$$\mathrm{e}(n) = \mathrm{x}(n) - \mathrm{y}(n) \quad (5)$$

D. Adaptive demodulation principle of MSK based on single frequency tracker

The output signal $y(n)$ of SFT can also be expressed as:

$$y(n) = w_1(n)d_1(n) + w_2(n)d_2(n) = C' \sin\left(\omega_0 nT + \varphi'\right) \quad (6)$$

The synthetic phase is:

$$\varphi' = \arctan\left[\omega_2(n) / \omega_1(n)\right] \quad (7)$$

The synthetic amplitude is:

$$C' = \sqrt{\omega_1^2(n) + \omega_2^2(n)} \quad (8)$$

According to formula (7) and formula (8), we can obtain the syntheticphase φ' and the synthetic amplitude C' by adjusting the weight coefficients ω_1, ω_2. The SFT not only tracks the frequency, but also tracks the phase and amplitude of the signal, so it has a strong tracking ability.

The schematic diagram of the adaptive single frequency tracker based on the MSK signal is shown in Figure 5

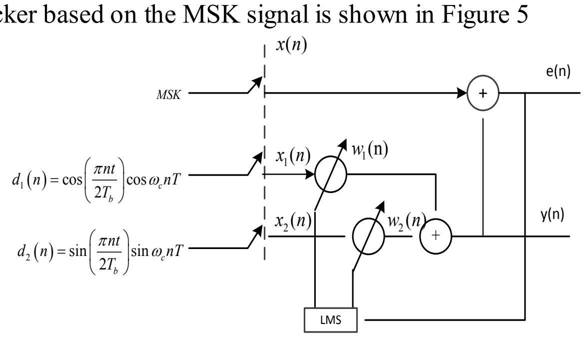

Fig.5 Schematic of adaptive single frequency tracker based on MSK signal

According to the modulation principle of the MSK signal, after theseries-to-parallel conversion processing, the signal is divided into two signals, the two signals are modulated respectively, and the carrier wave is $\cos\frac{\pi t}{2T_B}\cos\omega_c t$ and $\sin\frac{\pi t}{2T_B}\sin\omega_c t$ respectively. They are two strictly orthogonal signals. Therefore, the reference input of the single frequency tracker should also be a strictly orthogonal reference signal with the same frequency.

$$d_1(n) = \cos\frac{\pi nT}{2T_B}\cos\omega_c nT \quad d_2(n) = \sin\frac{\pi nT}{2T_B}\sin\omega_c nT \quad (9)$$

After the MSK signal passes through the single frequency tracker, the original binary baseband signal can be recovered through the steps of absolute value, median filtering and samplingjudgment.

III. EXPERIMENTAL SIMULATION

Because the communication system is bound to be disturbed by noise in the actual transmission process, this paper discusses the demodulation of MSK signal for single frequency tracker in the case of adding Gauss white noise to the channel. In order to observe the demodulation performance of the system for MSK signals under different degrees of noise, the performance of the system can be analyzed by changing SNR and observing the BER of the system [13].

The signal added to the noise is represented as:

$$nmsk = a * msk + noise \quad (10)$$

'a'stands for the amplitude of the MSK signal

The power of the MSK signal is:

$$S = \frac{a^2}{2} \quad (11)$$

The average power of Gaussian white noise is:

$$N = \sigma^2 = 1 \quad (12)$$

The signal-to-noise ratio is:

$$r = \frac{S}{N} = \frac{a^2}{2\sigma^2} = \frac{a^2}{2} \quad (13)$$

$$SNR(dB) = 10\lg r = 10\lg\frac{a^2}{2} \quad (14)$$

When the SNR=8dB is given, the MSK modulation and demodulation simulation waveform is shown in Figure 6.The meanings of waveforms are shown in Table 1.

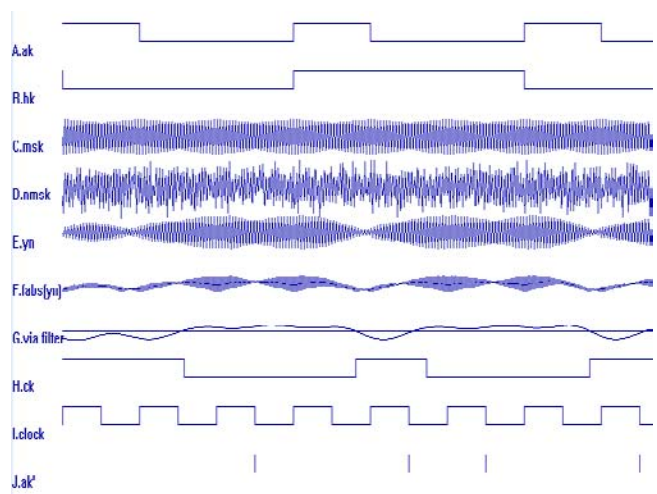

Fig.6 SNR=8dB MSK modulation and demodulation simulation waveform

Comparing the waveforms of MSK and *nmsk*, it is found that the MSK waveform is obviously distorted. After the *nmsk* signal transmitted through a single frequency tracker, the waveform of the output signal*yn* corresponds to the jump of the differential codeBk. When *bk* jumps, the amplitude of *yn* decreases rapidly.Comparing the binary baseband signals of AK with AK, we find that, in addition to certain delays, the baseband signals are approximately equal. That is, the demodulation output is basically error free. The system can run correctly and get the correct waveform. It can be seen that the baseband signal is correctly restored.

TABLE I. The meanings of waveforms

Waveforms	Meanings
ak	baseband signal
bk	Difference code
msk	MSK signal
nmsk	MSK signal containing noise
yn	output signal
fabs(yn)	median filtering
via filter	the envelope of yn
ck	Decision level
clock	clock signal
ak'	Demodulated baseband signal

In this paper, the MSK demodulation system based on single frequency tracker is analyzed by error analysis, provided that the total number of symbols is sufficient. Multiple simulations ensure the accuracy of the results. The MSK demodulation based on single frequency tracker is compared with the traditional coherent demodulation to observe their bit error rate and analyze the performance of the system [14].The bit error rate curve is shown in Fig 7.

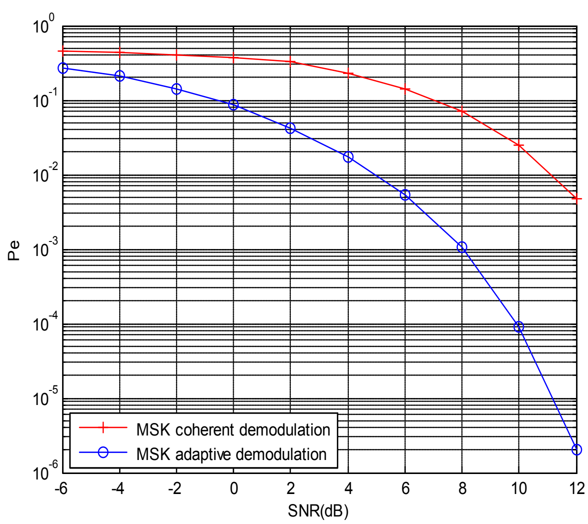

Fig.7 The bit error rate curve

Observe the error rate curve of the two demodulation methods, it can be seen that the bit error rate of MSK demodulation system based on single frequency tracker is lower. Single frequency tracker has better anti - noise performance than coherent demodulation system.

IV. CONCLUSION

This paper presents a method based on single frequency tracker to improve the performance of communication systems and reduce the bit error rate of the system. According to the characteristics of various signals to design a different single frequency tracker, the system can effectively track the signal changes, filter out the noise during transmission. The algorithm used in this paper is LMS algorithm, and other more efficient and convenient algorithms can be used to improve the performance of the system.

REFERENCES

[1] Dzung D, Von H T, Benninger H, et al. Modulation and demodulation of a combination of analog and digital signals: EP, EP1395012[P]. 2004.

[2] Wagner C, Wittum G. Adaptive Filtering[J]. NumerischeMathematik, 1997, 78(2):305-328.

[3] Sun W, Sweeting M N, Hodgart M S. Investigation of MSK and FSK modulation schemes for single channel communications system using DSP techniques on-board low Earth orbit microsatellite[C]// Tenth International Conference on Digital Satellite Communications. IET, 1995:9-16 vol.1.

[4] Pitchaiah T, Sridevi P V, Rao S K. Adaptive noise cancellation using LMS algorithm in Monte Carlo simulation[C]// International Conference on Electronics and Communication Systems. IEEE, 2015:368-372.

[5] Cioffi J M. The fast adaptive ROTOR's RLS algorithm[J]. IEEE Transactions on Acoustics Speech & Signal Processing, 1990, 38(4):631-653.

[6] Meng L, Zhu J, Zhao X, et al. Simulation for all-digital BPSK modulation and demodulation system[J]. Journal of Zhejiang University of Technology, 2003.

[7] Ding G D, Xiang X, Zhang J L, et al. Direct Spread Spectrum Signal Detection in Gauss White Noise[J]. Journal of Air Force Engineering University, 2003.

[8] Gronemeyer S, Mcbride A. MSK and Offset QPSK Modulation[J]. IEEE Transactions on Communications, 1976, 24(8):809-820.

[9] Hagmann W. Method for the transmission of digital data: US, US4774716[P]. 1988.

[10] Falconer D D. Adaptive filter theory and applications[M]// Analysis and Optimization of Systems. Springer Berlin Heidelberg, 1980:161-188.

[11] Widrow B, Mccool J M, Larimore M G, et al. Stationary and nonstationary learning characteristics of the LMS adaptive filter[J]. Proceedings of the IEEE, 1977, 64(8):1151-1162.

[12] Douglas S C, Zhu Q, Smith K F. A pipelined LMS adaptive FIR filter architecture without adaptation delay[J]. IEEE Transactions on Signal Processing, 1998, 46(3):775-779.

[13] Kaleh G K. Differentially coherent detection of MSK[C]// IEEE International Conference on Communications, 1988. ICC '88. Digital Technology - Spanning the Universe. Conference Record. IEEE, 1988:730-734 vol.2.

[14] Lambrette U, Mehlan R, Meyr H. Comparison of Demodulation Techniques for MSK[J]. 1996, 112:232-243.