

A Algorithm of Fast Digital Phase Modulation Signal Recognition

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Abstract

Classifications of digital modulation, the signal characteristics of the phase of the cycle will cause the phase wrapped and reduce overlapping signals in the received RF digital modulation automatically correct classification probability. According to the principle of minimum mean square, a phase modulation and phase modulation signal classification features as well as 2PSK and MPSK classification. The experiments illustrate the robust of the feature-based approach for automatic digital modulation classification and a higher recognition probability suitable to various pulse shaping filters, roll-off factors and symbol rates.

Keywords: digital phase modulation, modulation recognition

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1. Introduction

Automatic identification of the communication signal modulation types can be widely applied in signal monitoring, signal confirmation, interference identification, radio interception and electronic countermeasures. How to identify and monitor unknown communication signal modulation effectively? It is an important research topic in the military and civilian fields. The objects of modulation recognition are a variety of analog and digital signals; the methods to identify the unknown communication signal are modulation classification, multiple signals classification, or the same type of signal, but the different modulation order. Therefore, the nature of the modulation type identify automatically, signal sampling, extracting the key characteristics of the frequency, amplitude, phase, use of appropriate algorithms, processing capabilities through high-speed signal processing technology and computer operations, statistics, inferred, as a follow-up signal lays the foundation for the further processing of the demodulation, monitoring, interference[1].

In recent decades, a lot of automatic digital modulation classification methods were put out [1-4]. Broadly speaking there are two main categories: likelihood-based methods and feature-based methods [4].

Likelihood-based methods can achieve the best classification results, but higher computational complexity is not suitable for engineering applications; feature-based approaches cannot achieve the best classification results, but choosing the right features and the threshold can achieve approximately excellent classification results, and low computational complexity, and convenient to use.

2. Analysis of the phase modulation signal recognition

2.1. Structure of modulation recognition

Communication signal modulation recognition generally consists of three modules: data preprocessing, feature extraction and classifier-identification, as shown in Figure 1. The data preprocessing module provide the data for subsequent modules. Preprocessing task is the carrier frequency estimated, to in-phase and the orthogonal component decomposition. The feature extraction module is to extract the signal from the incoming data domain and transforms domain features. The time-domain characteristics include the instantaneous frequency of the signal, the instantaneous amplitude and instantaneous phase histogram or statistical parameters; transform domain features include a spectrum characteristic of the signal.

Classifier-identification module determines the appropriate decision rules and classifier structure.

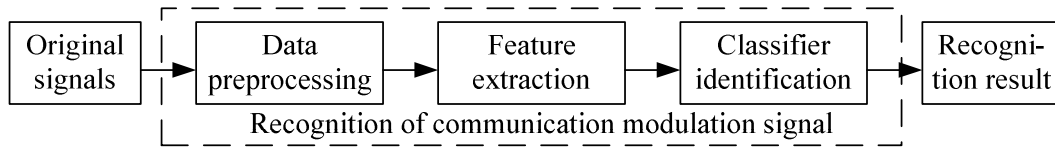


Figure 1 The basic structure of signal modulation recognition

Any modulation recognition algorithm is required to achieve the following functions:

- Identifying the modulation type automatically, and high recognition rate;
- Automatic adapting the SNR (signal to noise ratio) changes in a complex conditions;
- Wider range of modulation types can be identified automatically;
- Reducing the complexity of the algorithm, and achieving real-time analysis;

In these processes, the key steps are to automatically identify the modulation type of feature extraction and classification method, determining the identified effects and the complexity of the algorithm.

2.2. FSK(frequency shift keying) signal

2FSK/MFSK frequency shift keying modulation principle, the carrier frequency varies with the change of the digital baseband signal. 2FSK signal mathematical expression:

$$S(t) = m(t) \cos(\omega_1 t) + \overline{m(t)} \cos(\omega_2 t) \quad (1)$$

Where the ω_1 is the symbol 0 of a carrier angular frequency, is the symbols 1. $m(t)$ is normalized with a symbol value between 0 and 1, $\overline{m(t)}$ is the anti-code.

MFSK signal is 2FSK direct marketing. Its mathematical expression in general can be written as

$$S(t) = \cos(\omega_c t + a_n \Delta \omega_m t) \quad (2)$$

Where ω_c is the carrier angular frequency, ω_m ($m = 0, 1 \dots M-1$) is the corresponding offset. The normalized value is 0, 1, ... M-1 within a symbol.

The power spectrum of MFSK signal is the same as 2FSK, consists of two parts, the continuous spectrum and discrete spectrum, which continuous spectrum consists of two bilateral band spectrum overlays, discrete spectrum located in the position of the two carriers.

2.3. PSK(phase shift keying) signal

2PSK/MPSK phase shift keying signal modulation principle, the carrier phase changes with the change of the digital baseband signal.

2PSK mathematical expression can generally be written as

$$S(t) = m(t) \cos(\omega_c t) \quad (3)$$

where: $m(t)$, the normalized value is a symbol -1,1.

The MPSK mathematical expression for

$$S(t) = \cos(\omega_c t + \varphi_n) \quad (4)$$

Where: φ_n are phase parameters controlled by information, a finite number of discrete values normalized within a symbol.

If you make:

$$I(t) = \sum_n \cos(\varphi_n) \quad (5)$$

Then

$$Q(t) = \sum_n \sin(\varphi_n) \quad (6)$$

The MPSK another expression is as follows:

$$S(t) = I(t) \cos(\omega_c t) + Q(t) \sin(\omega_c t) \quad (7)$$

Under the same symbol transmission rate, power spectrum of MPSK signal and 2PSK are the same, bandwidth is also.

2.4. The characteristics of digital phase modulation signals

In digital communication systems, signal transmitter has a pulse shaping filter, Gaussian, Nyquist and root Nyquist(Rn) filter. Different filters are not the same as the instantaneous frequency, the instantaneous amplitude and instantaneous phase of the signal. Gaussian filter has little effect on the instantaneous frequency of the signal, the instantaneous amplitude and instantaneous phase, but the Nyquist and root Nyquist(Rn) filters have not, shown in Figure 2 (a). The 2FSK signal code through the graphical Gaussian pulse shaping filter, the instantaneous frequency is basically no change, after the graphical Nyquist filter (b), the instantaneous frequency has changed greatly. In theory PSK signal is a constant enveloping signal, the pulse shaping filter, the received PSK signal becomes non-constant enveloping signals [5].

2.5. Digital phase modulation and phase modulation signal recognition algorithm

Set the same component is $i(n)$, the orthogonal component is $q(n)$ of the receiver outputting IQ signal, then the zero-IF complex signal is

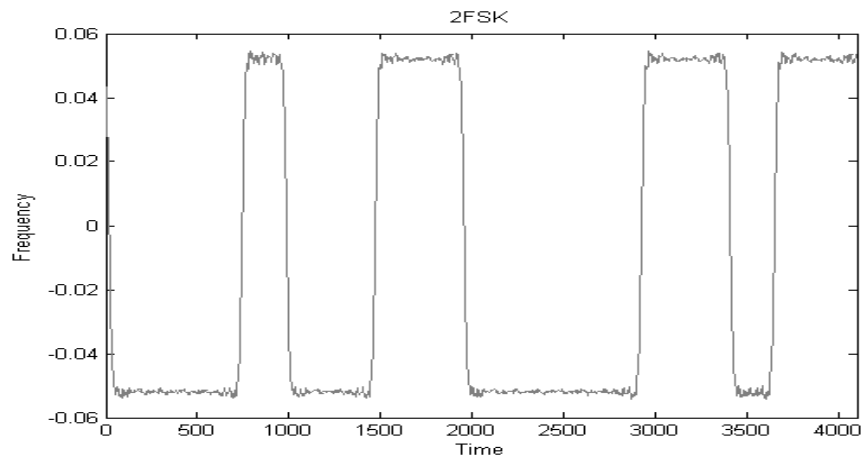
$$x(n) = i(n) + jq(n), \quad n = 0, 1, \dots, N-1 \quad (8)$$

Where N is the signal length, according to the definition of the complex signal, the instantaneous phase of the signal is as follows:

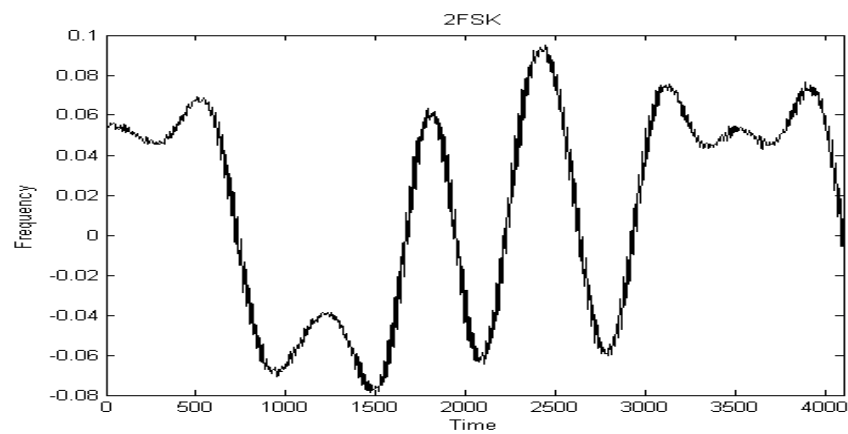
$$\phi(n) = \text{atan}\left(\frac{q(n)}{i(n)}\right) \quad (9)$$

Mathematically, atan is a periodic function of π , but the phase of the communication signal is a 2π cycle, where the value of $\phi(n)$ interval is $(-\pi, \pi]$. Function atan value change to the range $(-\pi, \pi]$ according to the symbol in the quadrant $q(n)$ and $i(n)$ [6]. That is,

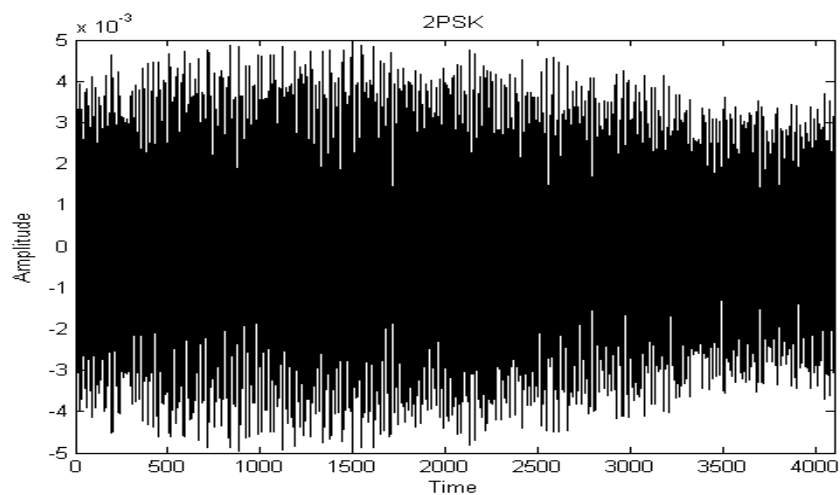
$$\phi(n) = \begin{cases} \theta(n) & , \quad q(n) \geq 0, i(n) \geq 0 \\ \theta(n) - \pi & , \quad q(n) < 0, i(n) < 0 \\ \theta(n) + \pi & , \quad q(n) > 0, i(n) < 0 \\ \theta(n) & , \quad q(n) < 0, i(n) > 0 \end{cases} \quad (10)$$



(a) Gaussian filter



(b) Nyquist filter



(c) PSK

Figure 2. Influence of pulse shaping filter on signal

Signal phase will increase linearly over time, when the phase exceeds thread ($\pm\pi$), a phase wrapped occurs, so the original phase of signal needs to be unwrapped. The method of phase unwrapped is as follows:

$$\phi_u(n:end) = \begin{cases} \phi(n:end) - 2\pi, & \text{if } \phi(n) - \phi(n-1) > \pi \\ \phi(n:end) + 2\pi, & \text{if } \phi(n) - \phi(n-1) < -\pi \end{cases} \quad (11)$$

Where $\phi_u(n)$ is the phase of unwrapped, although signal takes phase unwrapped, but there is linear phase component. This linear phase component will seriously affect the results of the ADMC. Eliminating the linear component can be solved by the minimum mean square error method[7].

First, assuming the predicted straight line as $\hat{y} = kx + b$, according to the principle of minimum mean square error, then

$$\min \left\{ \left\| \mathbf{Y} - \hat{\mathbf{Y}} \right\|^2 \right\} = \min \left\{ \left\| \mathbf{Y} - k\mathbf{X} + b\mathbf{I} \right\|^2 \right\} \quad (12)$$

Where $\mathbf{Y} = [y_1, y_2, \dots, y_N]^T$, $\mathbf{X} = [x_1, x_2, \dots, x_N]^T = [0, 1, \dots, N-1]^T$, \mathbf{I} is unit matrix, N is the signal of the sampling points.

According to the equation (12), obtaining k and b , the phase of the eliminated linear phase component is as follows:

$$\phi_m(n) = \phi_u(n) - (n-1) \times k, \quad n = 1, \dots, N \quad (13)$$

After $\phi_m(n)$ modulo π , then it becomes $\phi_{m|\pi}(n)$, MPSK signals can be distinguished by the variance of $\sigma_{mp|\pi}$, $\sigma_{mp|\pi}$ is variance of $\phi_{m|\pi}(n)$.

$$\phi_{m|\pi}(n) = \text{mod}(\phi_m(n), \pi) - \pi \quad (14)$$

$$\sigma_{mp|\pi} = \sqrt{\frac{1}{N} \sum_{n=1}^N \phi_{m|\pi}^2(n) - \left(\frac{1}{N} \sum_{n=1}^N \phi_{m|\pi}(n) \right)^2} \quad (15)$$

3. The results of the experiments

The hardware platform used in the experiment consists of the Agilent E4438C, the Marconi 2052 signal source, and the Agilent N8201A RF receiver module. The signal level is -30dBm and -60dbm, roll-off factor is 0.35 and 0.5, the filters are Gaussian, Nyquist and root Nyquist (Rn), the carrier frequency is 100MHz and 500MHz, and symbol rate is 2kbps and 20kbps. Each type of signal selected combination of parameters, the experiment signals are 4096 point, sampling by 1000 times, the experimental results can be shown in Table 1.

Table 1. Classification result of different rolloff-factor and pulse shaping filter

Type of signal	result of recognition								
	2PSK			4PSK			8PSK		
	G	N	Rn	G	N	Rn	G	N	Rn
2PSK (0.35)	200	200	200	0	0	0	0	0	0
2PSK (0.5)	200	200	200	0	0	0	0	0	0
4PSK (0.35)	0	0	0	200	200	200	0	0	0
4PSK (0.5)	0	0	2	200	200	198	0	0	0
8PSK (0.35)	0	0	0	0	0	0	200	200	200
8PSK (0.5)	0	0	0	0	0	0	200	200	200

Each PSK signal is divided into Gauss (G), Nyquist (N) and root Nyquist (Rn) pulse shaping filter in accordance with different filter, roll-off factor in the experiment sets for both 0.35 and 0.5,

each signal samples 200 point, total 3600 samples. The data are the recognition results of the PSK signal within three different pulse shaping filters and different roll-off factor (0.35 and 0.5) in Table 1. Comparing with ideal Gaussian white noise channel identification method, the model used in this paper is to identify the pulse shaping filter modulation signal through the air, the complex channel, is an engineering method, compared to other experimental results [8-9], the SNR of 0, 5, 10 dB, under the conditions of the program identification rate of 98% or more.

4. Conclusion

In this paper, a method to distinguish between phase-modulated and non-phase-modulated signal, and distinguish 2PSK and MPSK based on the characteristics of the IQ signal. Experimental tests prove that the method is robust to achieve a high recognition rate; under the emission level is generally greater than -70dbm with a variety of different pulse shaping filter, different symbol rate, and the different frequency bands communication band-limited signals.

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References

- [1] Nandi A K, Azzouz E E. Algorithms for Automatic Modulation Recognition of Communication Signals. *IEEE Trans on Communications*, 1998, 46(4): 431-436.
- [2] Azzouz E E, Nandi AK. Automatic Modulation Recognition of Communication Signals. Boston: Kluwer Academic Publishers, 1996.
- [3] Park C-S, Kim D Y. A Novel Robust Feature of Modulation Classification for Reconfigurable Software Radio. *IEEE Trans on Consumer Electronics*, 2006, 52(4): 1193-1200.
- [4] Dobre O A, Abdi A, Bar-Ness Y, Su W. Survey of Automatic Modulation Classification Techniques: Classical Approaches and New Trends. *IET Communications*, 2007, 1(2): 137-156.
- [5] Grimaldi D, Rapuano S, De Vito L. An Automatic Digital Modulation Classifier for Measurement on Telecommunication Networks. *IEEE Trans on Instrumentation and Measurement*, 2007, 56(5): 1711-1720.
- [6] Qu Jun-suo. A practical fast digital phase modulation signal classification and recognition method. *Journal of Chongqing University of Posts and Telecommunications*, 2011, 23(2): 190-194.
- [7] Guo Yong-ming, LI Ning, QU Jun-suo. A feature-based algorithm for automatic digital modulation classification. *Journal of Xi'an University of Posts and Telecommunications*, 2010, 15(1): 31-35.
- [8] Chen Ming, Zhu Qi. Research On Automatic Modulation Recognition Based on Improved Neural Network. *Journal of Chongqing University of Posts and Telecommunications: Natural Science Edition*, 2009, 21(6): 764-780.
- [9] Hu You-qiang, Liu Juan, et al. Digital modulation recognition based on instantaneous information. *The Journal of China Universities of Posts and Telecommunications*, 2010, 17(3): 52-59.