

# A Plug and Play Module to identify type of Modulation

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**Abstract - The main goal of this work is to identify the type of modulation in an independent manner with the use of plug and play module. As, some of the applications of SDR involve military and relief operation (non-cooperative relay environments), wireless data is sent using a suitable modulation technique to provide suboptimal and optimal receivers, savings in power for a required bit error rate etc. The constraints like limited available frequency bandwidth, time-varying multipath fading, shadowing and mobility are considered in this work. Efficient techniques to overcome these limitations involve adaptively changing the modulation mode based on the channel quality information sensed by the receiver and fed back to the transmitter to the throttling of the window size by the sender. The receiver should identify the type of modulation the transmitter has sent**

**Keywords:** DPSK, Adaptive modulation

## Introduction

A wireless channel model can be considered both as a time-domain and frequency-domain channel with non-stationary and ergodic features. As a time-domain channel it is complex normal and independent across filter taps and users. Similarly in frequency-domain it has correlated channel gains across subcarriers. The general multidimensional linear channel model adequately represents a plethora of communication system models which utilize multidimensional transmit-receive signals for attaining increased rates and reliability in the presence of fading. Proposed method used to enhance feature extraction schemes, to optimize the number of feature vectors considered for identification and to combine deterministic and blind autonomous modulation identification schemes. The fundamental feature vectors are needed to be identified and used.

## Related works

HadiValipour, M et al., [2012] have proposed a new method for automatic digital modulation classification in presence of AWGN. In this method a set of three different types of features is extracted to be employed in recognition process. Classification is based on SVM as a powerful method for pattern recognition, and PSO to configure kernel parameters [1].

By referring to the paper by Jaspal Bagga & Neeta Tripathi [2011] in which the results of experiment carried out on a digitally modulated signals concludes that analysis using instantaneous and random features classification, we understood that: A receiver with completely non informative about the modulation scheme has inferior ability than a receiver used in a wireless Network with ability to detect the autonomous modulation scheme for detection. [2].

However based on the signal and Noise ratio (SNR) classification of modulated signals can be done in which noise interference is used as factor to classify the modulation scheme and is demonstrated in the paper published by Tao He &Zheng-Ou Zhou in 2008 [3].

Further the classification of M-array frequency shift keying signals can be done based on the analysis of complex envelope particularly the analysis of cyclostationary statistics modulated signals their periodicity was proved to be the better way to classify the modulated signal which was proposed and presented by Antti-VeikkoRosti& Visa Koivunen in 2000[4].

Prakasam, P &Madheswaran, M [2008] presented that by adopting the wavelet transform and statistical parameters the digital modulation identification model can be improved further. In this method the dynamic methodology was effectively used with the help of a priori unknown paparametersto identify the modulation schemes of M-ary PSK, M-ary QAM, GMSK and M-ary FSK modulations [5].

Liang Hong published paper in 1999, which deals with automatic identification of digitally modulated signal through transients in amplitude, frequency or phase was considered as a rapidly evolving field in the application areas like electronic warfare, surveillance and threat analysis was presented in this paper. [6].

### Adaptive modulation

The identification of modulation scheme can be done through either [12] Decision Theoretic approach in which likelihood functions were used for probabilistic and hypothesis testing of arguments or Statistical Pattern approach [10][11] in which prominent characteristics of the signals are extracted and analysed to find the modulation scheme. Other ways of identification of modulation scheme is through i) Spectral Processing ii) Instantaneous amplitude, Phase and Frequency parameters iii) Instantaneous Amplitude, phase and frequency histogram iv) Combination of all the above three types and v) Universal demodulators. The Decision theoretic approach uses the likelihood function where arguments to formulate the recognition problem and classification is performed. The drawback with this approach is its high computational complexity, difficulty in implementation and lack of robustness to model mismatch. The pattern recognition approach is based on feature extraction and recognition in which the prominent characteristics of the signal is extracted and analysed to indicate the modulation type. The important metrics used in this type of analysis are interference and system capacity. In a radio communication system the major bottle neck is the interference which causes the cross talks on voice channels and may results in severe problem like call dropping in the channels.

Normally Interferences may occur due to any of the following reasons,

- A mobile station working in the same base cell
- There may be a ongoing call in the neighboring cell
- If any base station operates in the frequency range of currently operating cell

The interferences are of two types

- i. Co-channel Interferences and
- ii. Adjacent Channel Interferences.

#### Co-channel Interference

It can be interference between signals from co-channel nodes [8]. The preferred way to reduce this interference is to separate nodes by minimum distance ‘D’ in a cell radius ‘R’ which results in a Co-channel reuse ratio is given by

$$Q = \frac{D}{R} = \sqrt{3N} \quad (1)$$

Where, N-cluster size for hexagonal geometry

where  $i_0$  = the number of Co-channel interfering nodes, the signal to interference ratio (SIR) is

$$\frac{S}{I} = \frac{S}{\sum_{i=1}^{i_0} I_i} \quad (2)$$

In (2) S represents the desired signal power from its base station and

$I_i$  is the interference power caused by  $i^{\text{th}}$  interference co channel station.

The average received power  $P_r$  at a distance ‘d’ from the transmitting antenna as per the poynting vector theorem

$$P_r = P_0(d/d_0)^{-n} \quad (3)$$

Hers  $P_0$  = received power

$d_0$  =s the distance from the transmitting antenna

$n$  = path loss exponent (ranges between 2 and 4)

Equation (3) can also be written as,

$$P_r(\text{dBm}) = P_0(\text{dBm}) - 10\log(d/d_0) \quad (4)$$

If all the interference base stations are equidistant from the desired base station, the signal to noise interference ratio is given by

$$\frac{S}{R} = \frac{(D/R)^n}{i_0} = \frac{(\sqrt{3N})^n}{i_0} \quad (5)$$

#### Channel support for Time slot based modulation schemes (TSMA)

Number of channels in Time slot based modulation schemes system is given by

$$N = \frac{m(B_{\text{tot}} - 2B_{\text{guard}})}{B_c} \quad (6)$$

Where m represents the maximum number of TSMA users on radio channel,  $2B_{\text{guard}}$  (one frequency band allocated at the low end and other at the high end) and  $B_c$  is the channel bandwidth.

#### Capacity of Users

Let the number of users be N. The signal to noise ratio is given by

$$\text{SNR} = \frac{S}{(N-1)S} = \frac{1}{N-1} \quad (7)$$

Where, S is the desired signal power and N-1 is the number of interfering users

The signal to noise ratio at the base station receiver can be represented in terms  $E_0/N_0$ .

$$E_0/N_0 = \frac{S/R}{(N-1)(S/W)} = \frac{W/R}{N-1} \quad (8)$$

Where, S is the signal power and R is the base band information bit rate. Bit energy to noise is obtained by dividing signal power by base band information bit rate R.

**Automatic Digital Modulation Recognition in Presence of Noise**

Automatic digital modulation[13] recognition is implemented in the presence of AWGN. This method employs three different types of features which are extracted to perform the recognition process[7]. They are Spectral based features, Statistical features and Wavelet based features. Hierarchical structure for classification of signals[16] is shown in Figure1. The first layer in the structure detects the incoming modulation type as either ASK, FSK, PSK or QAM. The next layer implements the corresponding classifiers to identify the modulation type's order (for example, 2ASK, 4ASK, 8ASK or 16ASK).

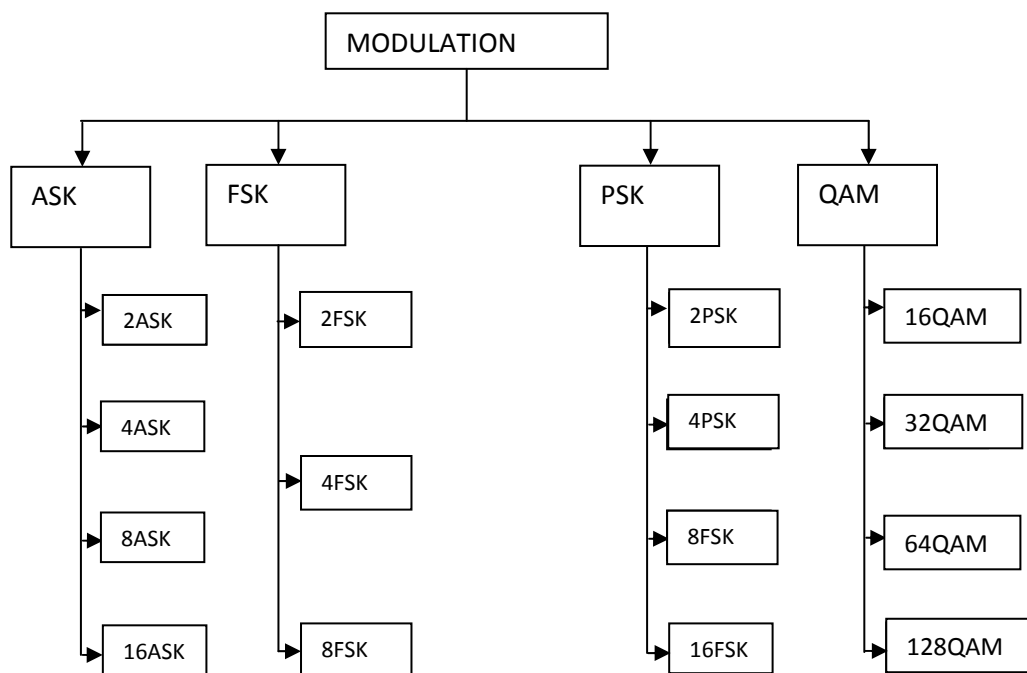


Figure 1: Modulation types using Hierarchical structure

Different types of digitally modulated signals (corrupted by AWGN) are simulated to measure the performance and suitability of the method. The following Table 1 shows the features and expressions for various types of modulations[14].

Table1. Various types of modulations with their features

Signal	Expression	Features
<b>Amplitude Modulation</b>		
Amplitude modulation	$A[1 + ms(t)] \cos \omega_c t$	Modulation and demodulation is simple Bandwidth of S(t) is B Hertz
Double sideband modulation	$s(t) \cos \omega_c t$	Modulation simple but demodulation is difficult. Power requirement is 50 % less than that of AM.
Quadrature amplitude modulation	$s_1(t) \cos \omega_c t + s_2(t) \sin \omega_c t$	Bandwidth requirement is less
Single sideband modulation	$\frac{1}{2} [s_1(t) \cos \omega_c t + s_2(t) \sin \omega_c t]$	Bandwidth requirement is less but difficult to demodulate
Vestigial sideband modulation	$\frac{1}{2} [s(t) \cos \omega_c t + js(t) \sin \omega_c t]$	Bandwidth requirement is less but difficult to demodulate. Only a half the sideband is used.
Phase modulation	$A \cos [\omega_c t + s(t)]$	Difficult to implement. Has distortion and noise durability.
Frequency modulation	$A \cos [\omega_c t + \int s(t) dt]$	Modulation and demodulation simple. Has distortion and noise durability.

Pulse Modulation			
Pulse modulation	amplitude	$\sum_{n=-\infty}^{\infty} [as(nT) + K] \pi(t - nT/\tau)$	Modulation and demodulation simple. Has distortion and noise durability.
Pulse modulation	duration	$\sum_{n=-\infty}^{\infty} \pi(t - nT/\tau(s)),$ $\tau(s) = as(nT) + K$	Amplitude is not affected by noise but sides of pulse are affected by noise.
Pulse modulation	position	$\sum_{n=-\infty}^{\infty} \pi(t - nT - \alpha/\tau(s)),$ $\alpha = as(nT)$	Pulses are equal height and width. Only one side of the pulse affected by noise.
Digitized form of Analog signal			
Pulse code modulation		BW=2nB, N=No. of quantization, B=BW of the analog signal	Modulation and demodulation simple so widely used. Sturdy against noise and distortion hence used for high quality transmission.
Binary Modulation			
Amplitude shift keying		$h(t) \cos \omega_c t, \quad h(t) = \begin{cases} 0 \\ 1 \end{cases}$	Modulation and demodulation simple. Sturdy against noise and distortion.
Frequency shift keying		$\cos[\omega_h(t)t], \quad \omega_h(t) = \begin{cases} 2\pi f_0 \\ 2\pi f_1 \end{cases}$	High energy requirement than ASK. Implementation complicated. Sturdy against noise and distortion.
Phase shift keying		$\cos[\omega(t) + \phi(t)], \quad \phi(t) = \begin{cases} 0 \\ \pi \end{cases}$	High energy requirement than ASK. Implementation complicated.

**DPSK modulation and Detection**

In DPSK modulation, in step1 data is differentially encoded by using comparator/decision block. In step 2, the differentially encoded data is phase encoded. The block diagram of DPSK transmitter and the DPSK pulse stream with noise are shown in figure 2 and figure 3.

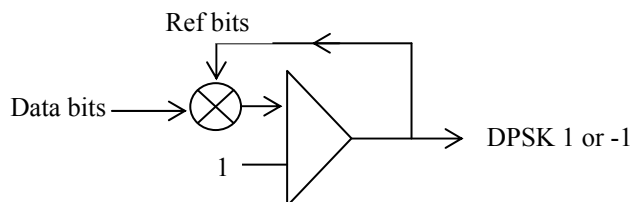


Figure 2 DPSK transmitter

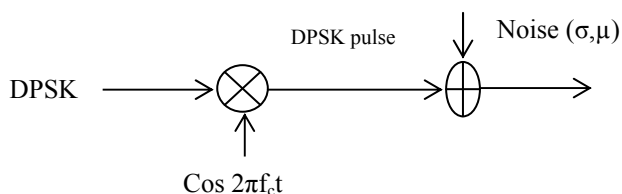


Figure 3 DPSK pulse stream with noise

**DPSK bit generation**

The table 2. Below is DPSK bit generation

Table 2. DPSK bit generation

$k_f$	$a(k_f-1)$	$ad(k_f-1)$ $ad[1]=1$	$a_r$	$a_d(k_f)$
2	$a[1]=1$	$ad[2]=1$	1	1
3	$a[2]=1$	$ad[3]=1$	1	1
4	$a[3]=-1$	$ad[4]=-1$	-1	-1
5	$a[4]=-1$	$ad[5]=1$	1	1
6	$a[5]=-1$	$ad[6]=-1$	-1	-1
7	$a[6]=1$	$ad[7]=-1$	-1	-1
8	$a[7]=-1$	$ad[8]=1$	1	1
9	$a[8]=1$	$ad[9]=1$	1	1

The Matlab tool is used to analysis the performance of DPSK transmitter. The pseudo code for DPSK bit generation is given by,

```

    -----
    For i=0 to 2:N+1
    ar=a(kf-1)*ad(kf-1)
    if ar equal to 1
    ad(kf)=1
    else
    ad(kf)=-1
    end if
    -----
    
```

**Simulated results**

The generated DPSK bits with 0.01 additive white noise is shown in figure 4.

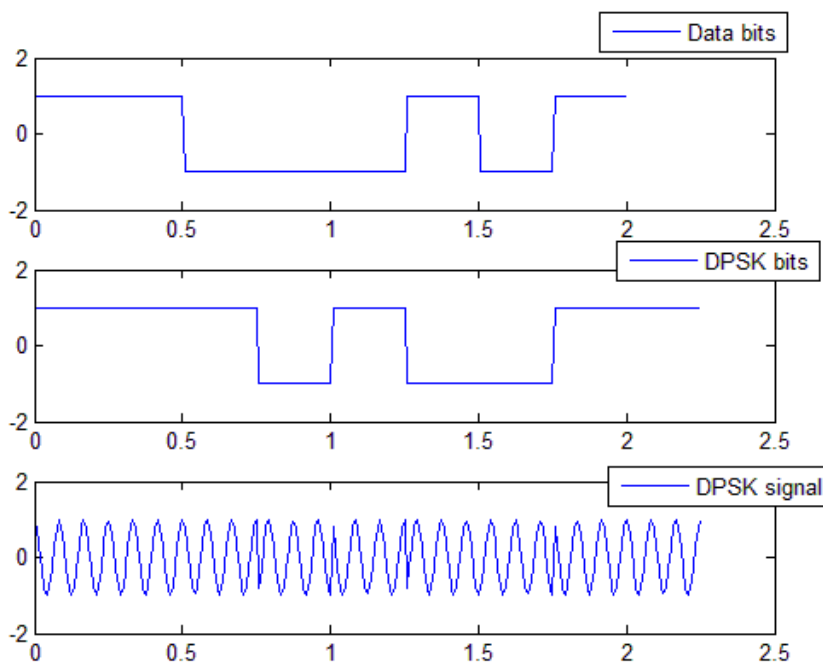


Figure 4 DPSK signal

**Information/ entropy analysis from extracted features**

**Information Calculation**

Table 3 shows the relation between the features extracted from the samples or frames and the DPSK modulation signal. Features related to each sample[13] are checked and the amount of information is calculated[17].

Table 3 Features of sample for DPSK modulation signal

Frame/sample	Feature F1	Feature F2	Feature F3	DPSK signal
1	Yes	Yes	1	Yes
2	Yes	Yes	6	Yes
3	Yes	No	5	No
4	No	No	4	Yes
5	No	Yes	7	No
6	No	Yes	3	No
7	No	No	8	No
8	Yes	No	7	Yes
9	No	Yes	5	No

Entropy for the DPSK signal from the above training samples can be calculated as

$$p(y)=4/9; p(n)=5/9;$$

$$H= -p(y) \log_2 p(y) -p(n) \log_2 p(n) =0.9911$$

The information gains for the two features using Boolean type F1 and F2 for these samples are calculated as follows,

**For feature F1**

Feature F1	No. of times DPSK signal is Positive	No. of times DPSK signal is Negative
Yes	3[i.e. samples 1,2 and 8]	1[i.e. sample 3]
No	1[i.e. sample 4]	4[i.e. samples 5,6,7,9]

Entropy for feature F1 is given as

$$H f1=4/9[-3/4 \log_2 3/4 - 1/4 \log_2 1/4] + 5/9[-1/5 \log_2 1/5 - 4/5 \log_2 4/5] =0.7616$$

Information gain for feature F1=Total Entropy- feature F1 Entropy

$$=0.9911- 0.7617 =0.2294$$

**For feature F2**

Feature F2	No. of times DPSK signal is Positive	No. of time DPSK signal is Negative
Yes	2[i.e. samples 1,2 ]	3[i.e. samples 5,6,9]
No	2[i.e. sample 4,8]	2[i.e. samples 3,7]

Entropy for F2 is given as

$$H f2=5/9 [-2/5 \log_2 2/5 - 3/5 \log_2 3/5] + 4/9 [-2/4 \log_2 2/4 - 2/4 \log_2 2/4] =0.9839$$

Information gain for feature F2=Total Entropy- feature F2 Entropy

$$=0.9911 - 0.9839 = 0.0072$$

**Performance metrics**

To achieve bit error rate of  $10^{-5}$  in both DPSK and coherent BPSK, the excess power margin is required in DPSK is 0.75 dB. Thus coherent BPSK requires lesser power. The required power margin for various bit error rate is shown in table 4.

Table 4 Power requirement to achieve bit error rate

To achieve required BER	Excess power margin required in DPSK in dB
$10^{-5}$	0.75
$10^{-4}$	0.9
$10^{-3}$	1.1
$10^{-2}$	1.6

Thus, if a system can achieve higher BER, then excess power margin required in DPSK is reduced[19] and in such cases DPSK can be tolerated compared to Coherent BPSK. This is shown in figure 5.

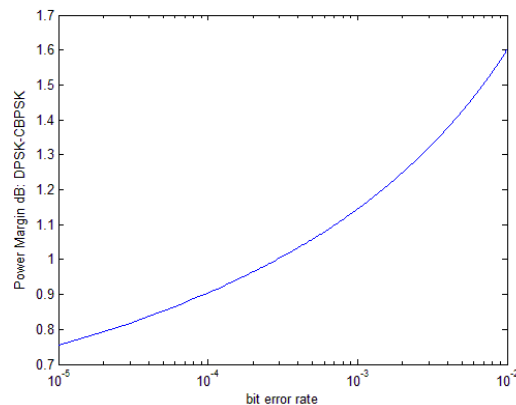


Figure 5 Power margin Vs Bit error rate

### Conclusion

In this paper, anticipative models where decisions are made based on the distribution and the actual realizations of the random quantities (adaptive models) are presented for MIMO system. A unique premodulation filter along with an adaptive code synchronization scheme is studied. Multistage recourse models with non-anticipative action for the 1<sup>st</sup> stage and recourse actions for the 2<sup>nd</sup> stage based on the realization of the random quantities are presented. Channel estimation models and self-healing noise filter modules are included to make the system robust. The proposed work shall assist in improved definitive prioritization among users and can trace boundary of capacity region with specified ratio. The proposed algorithms are adaptive with low complexity and more suited for online processing. Hardware implementation using embedded open source tools (Linux kernel) and Reconfigurable hardware is the future scope of this work.

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