Journal of Communication Engineering & Systems
Volume 2, Issue 2, August 2012, Pages 32-41

# Review of Adaptive Modulation Recognition Techniques

**Prof. Akash I. Mecwan*, Prof. Piyush M. Bhatasana, Prof. Vijay G. Savani**
Electronics and Communication Department, Institute of Technology,
Nirma University, Ahmedabad, Gujarat, India

**ABSTRACT**

*Rising development in software defined radio (SDR) systems has led to rapid development in adaptive modulation recognition (AMR) techniques. Different order modulations combined with different coding schemes allow sending more bits per symbol, thus achieving higher throughputs and better spectral efficiencies by the use of AMR. The aim of this paper is to study various AMR techniques and compare their advantages and disadvantages to find the most efficient and feasible technique. In the techniques studied, various modulation schemes like M-PSK, GMSK, M-FSK, M-QAM, etc., are capable of being recognized and subsequently demodulated if required.*

***Keywords:*** *Software defined radio (SDR), adaptive modulation and recognition (AMR)*

****Author for Correspondence*** E-mail: akash.mecwan@nirmauni.ac.in

## 1. INTRODUCTION

Software defined radio reconfigurability in radio development is not such a new technique as one might think. Already during the 1980s, reconfigurable receivers were developed for radio intelligence in the shortwave range. These receivers included interesting features like automatic recognition of the modulation mode of a received signal or bit stream analysis.

Software defined radio can be defined as a radio in which some or all of the physical layer functions are software defined. In other words, the software is used to determine the specification of the radio and what it does. If the software within the radio is changed, its performance and function may change. The above discussion shows that the radio, which is reconfigurable to various standards, encoding techniques or modulation techniques, is called a software defined radio (SDR). It can also be defined as a radio system or a radio communication system where components that have been typically implemented in hardware (e.g., mixers, filters, amplifiers, modulators/demodulators, detectors, etc.) are instead implemented by means of software on a personal computer or embedded computing devices. To achieve this, the software defined radio technology uses software modules that run on a generic hardware platform consisting of digital signal processing (DSP) processors as well as general purpose processors to implement the radio functions to transmit and receive signals.

In modern communication era, one not only has voice and data but also multimedia services. This requires designing of more intelligent and alert communication systems which provide spectrally efficient and flexible data rates. Furthermore, the increased reliance on computer networking and the Internet has resulted in a wider demand for connectivity to be provided anywhere all the time, thus raising the requirements of higher capacity and highly reliable systems. Using the said SDR with programmable hardware, the reconfigurable radio can adapt various standards and provide benefits such as multi-functionality, compactness, ease of upgrades, etc. Thus, the problem of incompatibility of general hardware radio for different operations is eliminated.

Adaptive modulation recognition (AMR) involves estimation of any modulation scheme at the receiver generally followed by its demodulation using relevant technique. The receiver does not have any prior knowledge of the modulation scheme the transmitter is using. Based on certain parameters of the received bit stream, the modulation scheme is estimated.

In SDR systems, AMR is applied before demodulation which selects the appropriate demodulator based on the modulation scheme resulting in a receiver with multiple demodulators in a single SDR rendering it multi-functional and compact to implement on a reprogrammable device.

This paper compares various AMR techniques, and based on the advantages and disadvantages, comes up with the most feasible technique that can be implemented.

The work is organized as follows in Secs. 2, 3, 4 and 5 describing the following techniques respectively:

- Feedback-based
- Mean square error-based
- Least mean square difference-based
- Wavelet transform-based

Advantages and disadvantages are compared and results are shown in Sec 6 while Sec 7 concludes the work.

## 2. FEEDBACK-BASED TECHNIQUE

In this technique, a single software-defined circuit for BPSK, QPSK, 16QAM, and 64QAM or even future coming techniques based on SDR is used. When the basic system is successfully built and tested, a cognitive engine (CE) must develop to automatically direct the SDR to load and execute the appropriate profile based on the response obtained through the feedback. The CE refers to predefined polices, while continuously sensing the channel situation. Then, it performs its logic to pick up the suitable configuration to execute it in the SDR system. In the model, the receiver evaluates received packets (i.e., SNR or BER) to estimate the channel quality indictor (CQI) module, then feedback the transmitter to reconfigure itself for the next packet to be sent (Figure 1).

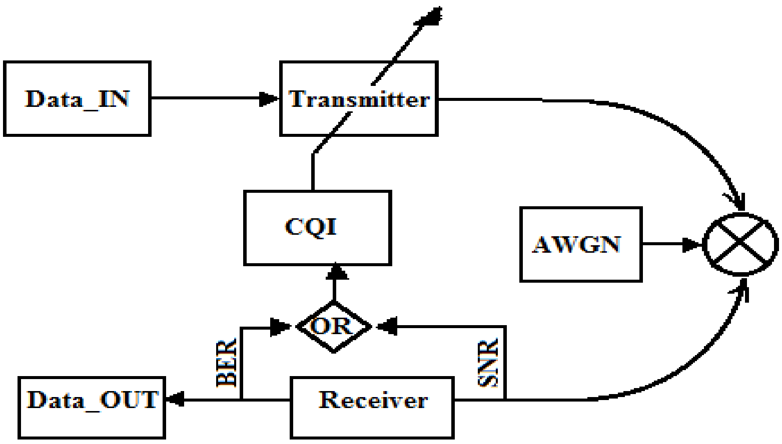


**Fig. 1:** *Adaptive System [1].*

## 3. MEAN SQUARE ERROR-BASED TECHNIQUE

This method recognizes the modulation scheme utilizing mean square error decision rule to recognize and differentiate M-ary PSK modulated signals in both AWGN and Rayleigh fading channels [2]. First of all, different PSK schemes are simulated after which the mean square error is calculated on whose basis appropriate a mean square error difference threshold is used to finally differentiate M-ary PSK schemes. The accuracy of this scheme is reported to be very high. The received band pass signal in the k-th signalling interval may be expressed as:

$$r(t, k) = S_m(t, k) + n(t, k)$$
$$k\,Ts < t < (k + 1)Ts$$

where $T_s$ is symbol duration, $S_m(t)$ is the message waveform corresponding to the M-PSK symbol $S_m$, $m = 1,2,3,\ldots M$. Assuming perfect carrier synchronization and timing recovery, we employ I-Q demodulation to get

$$r(k) = [r_I(k), r_Q(k)] = [s_{mI} + n_I(k), s_{mQ} + n_Q(k)]$$

These sequences of N signal samples are collected at demodulator output. Then it is checked how closely they match with the prototype ideal constellations. This degree of closeness is measured in terms of mean square error power defined as:

$$\mathrm{MSE(M)} = \left(\frac{1}{N}\right) \sum_{k=1}^{N} D_{k,M}^{2}\ , \ M = 2^q, q = 1, 2\ldots$$

where $D_{k,M} = \min\{|r(k) - s_m|\}$,
$m = 1,2,..M = \min\{|d_{k,M}|\}$

Now, lower-order PSK constellations are sub-sets of the higher order PSK schemes; therefore, when lower-order PSK symbols are transmitted, the received signal sequence r(k) will find a match not only with the corresponding constellation, it will also match with the higher-order constellation (with more or less the same degree of accuracy).

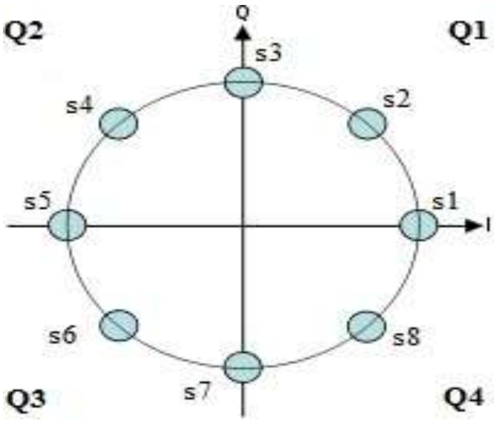


*Fig. 2:* *Constellation Diagram [2].*

**Case I: BPSK is transmitted**

Here, the received signal points will be scattered around the symbols S2 and S6 shown in Figure 2.

a. Majority of the points will be confined in the first and the third quadrants (Q1 and Q3) especially at high SNR. The contribution of these points towards MSE power will be the same in both BPSK and QPSK, i.e., MSE(2) = MSE(4), where r(k) = Q1 U Q3. However, this same set of points will result in a slightly lower MSE when matched to 8-PSK as some of these points will have closer match to 8-PSK symbols s1 or s3 and s5 or s7. Thus, MSE(8) < MSE(2), MSE(4), where r(k) = Q1 U Q3.

b. For a small fraction of the received points which lie in Q2 and Q4, their "match" with the BPSK prototype will be proper (the nearest symbols being s2 and s6) as compared to QPSK prototype (nearest symbols S4 and s8) and 8-PSK (nearest symbols s3, s4, s5 and s7, s8, s1). Thus, MSE(8) < MSE(4) < MSE(2), where r(k) = Q2 U Q4.

Finally, the observation of this case can be drawn as follows:

(i) when BPSK is transmitted, at any SNR, we shall find MSE(8) < MSE(4) < MSE(2).

(ii) At high SNR, the differences in MSE are negligibly small; only at low SNR, the differences are distinguishable.

**Case 2: QPSK is transmitted**

Now, r(k)s are scattered around the four symbols s2, s4, s6, s8. It follows that r(k) will match well with QPSK and 8-PSK prototypes while there will be large mismatch with BPSK prototype. Thus, MSE(2) > MSE(4) and MSE(8) at all SNR, MSE(8) ~ MSE(4) at high SNR and MSE(8) < MSE(4) at low SNR.

**Case 3**: **8-PSK is transmitted**

Following similar reasoning, one can conclude: MSE(2) > MSE(4) > MSE(8) at all SNR.

**Algorithm**

Step 1:

Compute: $D_{k,2} = |r(k)-S_m|$,

$m = 2$ if $r(k) = Q1$, $m = 6$ if $r(k) = Q3$

$D_{k,2} = \min\{|r(k)-s_m|\}$,

$m = 2, 6$ if $r(k) = Q2$ or $Q4$

$D_{k,4} = |r(k)-s_m|$,

$m = 2$ if $r(k) = Q1$, $m = 4$ if $r(k) = Q2$,

$m = 6$ if $r(k) = Q3$,

$m = 8$ if $r(k) = Q4$

$D_{k,8} = \min\{|r(k)-s_m|\}$,

$m = 1,2,3$ if $r(k) = Q1$, $m = 3,4,5$ if $r(k) = Q2$

$m = 5,6,7$ if $r(k = Q3$, $m = 7,8,1$ if $r(k) = Q4$

*Journal of Communication Engineering & Systems*
*Volume 2, Issue 2, August 2012, Pages 32-41*

Step 2: Compute $\mathrm{MSE(M)} = \left(\frac{1}{N}\right) \sum_{k=1}^{N} \mathrm{D}_{\mathrm{k,M}}^{2}$, M = 2,4,8

Step3: Compute Mean Square Error Difference (MSED)

$\mathrm{MSED}_{2-4} = \mathrm{MSE}(2) - \mathrm{MSE}(4)$

$\mathrm{MSED}_{4-8} = \mathrm{MSE}(4) - \mathrm{MSE}(8)$

Step 4: Decision rule

If $\mathrm{MSED}_{2-4} < \lambda_{2-4}$, BPSK is transmitted.

If $\mathrm{MSED}_{2-4} > \lambda_{2-4}$, then check if $\mathrm{MSED}_{4-8} < \lambda_{4-8}$, QPSK is transmitted.

If $\mathrm{MSED}_{4-8} < \lambda_{4-8}$, then 8-PSK is transmitted.

In order to determine the thresholds $\lambda_{2-4}$ and $\lambda_{4-8}$, distribution of MSED and frequency of its occurrence which is approximately Gaussian for both AWGN and Fading channels is plotted and thus the threshold value as the sum of the mean and standard deviation is set as $\lambda_{2-4} = \sigma_{2-4} + \mu_{2-4}$. This is shown in the Figure 3 of distribution of $\mathrm{MSED}_{2-4}$ at SNR = 10 dB in an AWGN channel.

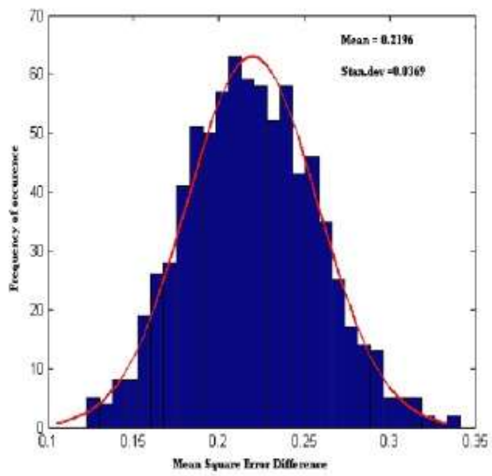


***Fig. 3:*** *Frequency of Distribution vs. Mean Square Error Difference Curve [2].*

Similarly, such distribution can be found out for $\mathrm{MSED}_{4-8}$, and thus one can find the corresponding threshold value. With these thresholds, the recognition capability is reported to be quite accurate for both AWGN and fading channels.

## 4. BASED ON LEAST MEAN SQUARE DIFFERENCE IN AMPLITUDE, PHASE AND FREQUENCY

The necessary pre-processing of the received signal and the signal itself is shown in Figure 4. The central frequency *fc* and the bandwidth *B* of a significant part of the received signal is estimated from a digitized and down-converted signal.

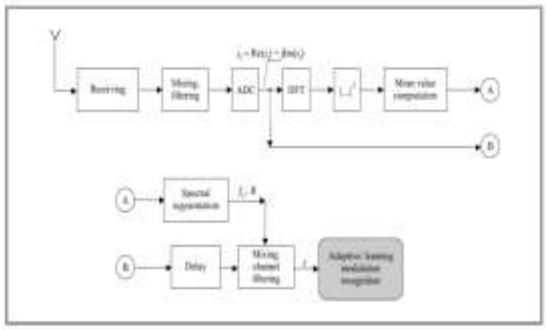

***Fig. 4:*** *Signal Analysis with Modulation Recognition [3].*

A complex baseband signal *z* is achieved by shifting the signal appropriately in frequency and subsequent filtering using the determined parameters. The signal *z* is fed to the modulation recognizer module. Figure 7 shows essential parts of the module. The next part of the processing moves in three branches: the upper two are associated with symbol rate recognition (SRR). The SRR is not used

*Journal of Communication Engineering & Systems*
*Volume 2, Issue 2, August 2012, Pages 32-41*

directly in this method; however, it is a feature that is modulation dependent and may be used in future for classification of waveforms. The SRR helps in estimating the symbol dwell time T. This value is required for calculating the values of phase difference values D within the third branch of the feature extraction module as shown in Figure 5. The coordinate resolving model finds the magnitude "a" and phase of the signal, before the signal is fed into the feature extraction module.

The feature extraction module gives four useful parameters: amplitude a, difference phase D, instantaneous frequency f and derivative of the instantaneous frequency *df*. The parameter T is used for calculating every pair of phase differences D. The instantaneous frequency f is used for detecting the simple frequency modulated signals like frequency shift keying and the derivative is used for recognizing the chirp signals. After the feature extraction process, the system is further divided into two branches: a learning branch and a testing branch. Using the learning branch, the system can adapt to various modulation types. For the purpose of learning, various parameters are accumulated in the learning phase and then properly scaled and written into histogram – one histogram for each of the four parameters. The histograms are then transformed to picture domains using DFT. These domains are called picture functions and they have an interesting property that the first part of the functions contains all the important information. In fact, the first quarter of the values of picture function is sufficient, as found experimentally. The picture function is stored and scaled as reference functions, for every modulation type a set of four functions. After finishing with the learning phase for all waveforms we are interested to classify, we change the switch following the feature extraction phase (Figure 5) to the lower position so that the testing phase can start. In the following comparison, the actual set of parameters is compared with the stored set and the set which matches the most with the actual set is the modulation type. The deviation measure used here is least mean square (LMS) and the fusion of the result obtained from individual parameter comparison is obtained by simply adding the LMS results from each comparison. The dashed lines in Figure 7 indicate the intention to search (in future) additional features of the wave using symbol rate recognition. Figure 5 shows the feature extraction module which extracts relevant parameter values for a, f, *df and* D.

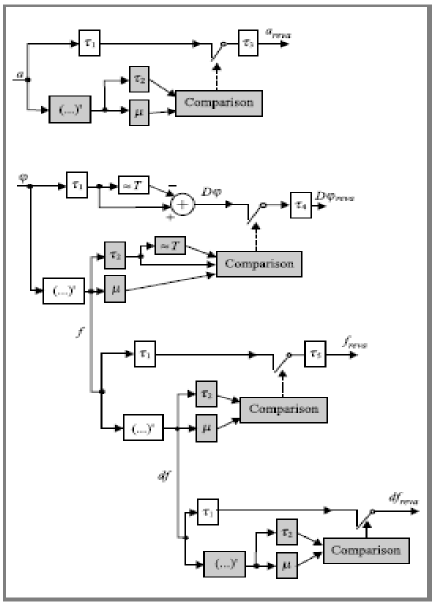


***Fig. 5:*** *Feature Extraction, Non-Synchronized [3].*

*Journal of Communication Engineering & Systems*
*Volume 2, Issue 2, August 2012, Pages 32-41*

## 5. WAVELET TRANSFORM-BASED TECHNIQUE

The algorithm is shown in Figure 6. The transmission of any signal is basically concentrated in the high frequency components. By extracting the coefficients using wavelet transform, the high frequency components can be withheld. After the extraction of the wavelet coefficients, a histogram is generated using them. Based on the number of peaks, the system identifies whether the modulated signal received belongs to the M-ary PSK and M-ary QAM group or GMSK and M-ary FSK group. After the major classification is done, the subsystem identifies individual modulation schemes based on the threshold values $T_1$ and $T_2$ derived in [6]. The equation for the threshold is given as

$$T_1 = \frac{\mu_{1,GMSK}\,\mu_{2,M\text{-}aryFSK} + \mu_{1,M\text{-}aryFSK}\,\mu_{2,GMSK}}{\mu_{2,GMSK} + \mu_{2,M\text{-}aryFSK}}$$

$$T_2 = \frac{\mu_{1,QPSK}\,\mu_{2,QAM} + \mu_{1,QAM}\,\mu_{2,QPSK}}{\mu_{2,QPSK} + \mu_{2,QAM}}$$

Here,

$\mu_1$ is the statistical mean of the process.

$\mu_2$ is variance of the process.

The thresholds are determined experimentally first, and then fed to the decision-making device.

Now, if $\mu_1 < T_2$, the received signal is QPSK or BPSK, else the signal is QAM; similarly, if the signal received has $\mu_2 > T_1$, the received signal is GMSK, else it is M-FSK [4–6]. The threshold between any two modulation schemes can be calculated through their mean and variance. When fed to the system, the decision between two waves can be taken.

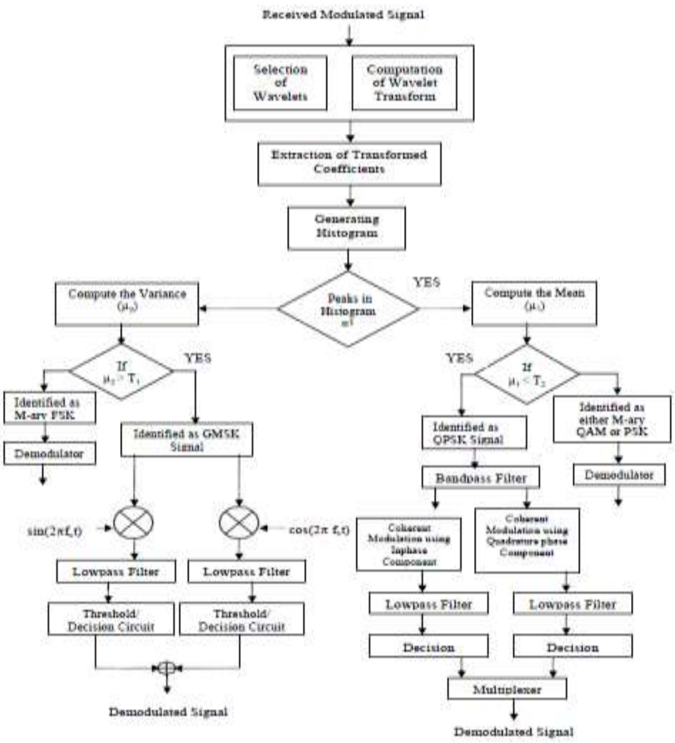


***Fig. 6:*** *Algorithm for Wavelet Transform Method [6].*

## 6. ADVANTAGES AND DISADVANTAGES

For the feedback-based technique, the advantage is that the scheme is simple and easy to implement and variety of modulation schemes can be used, but the disadvantages are far higher. One requires accurate channel estimation at the receiver and reliable feedback path through which receiver reports channel state information. But this cannot be achieved practically because the mobile channel is not constant but time varying; moreover it is not practical for the receiver to send channel state information to the transmitter via feedback.

For the technique based on the mean square error, the recognition accuracy is predicted to be quite high for even lower SNR values. But the main disadvantage is that the technique can distinguish only between M-PSK schemes and the phase of the transmitted signal needs to be known. In the third technique based on mean square difference, an extensive database is extracted from the memory bank (amplitude, phase, instantaneous frequency and differentiated phase) for modulation recognition. The chances of wrong detection are minimal, so the bit error rate will be very less. Also the process is easy to understand and results are quite reliable.

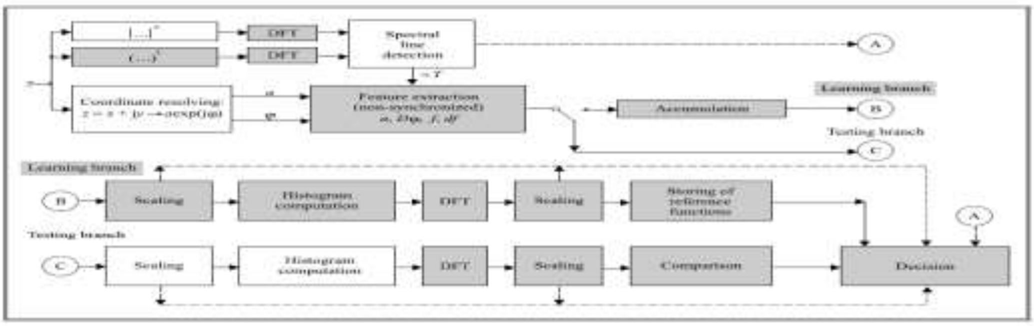


*Fig. 7:* *Adaptive Modulation Recognition, Non-Synchronized [3].*

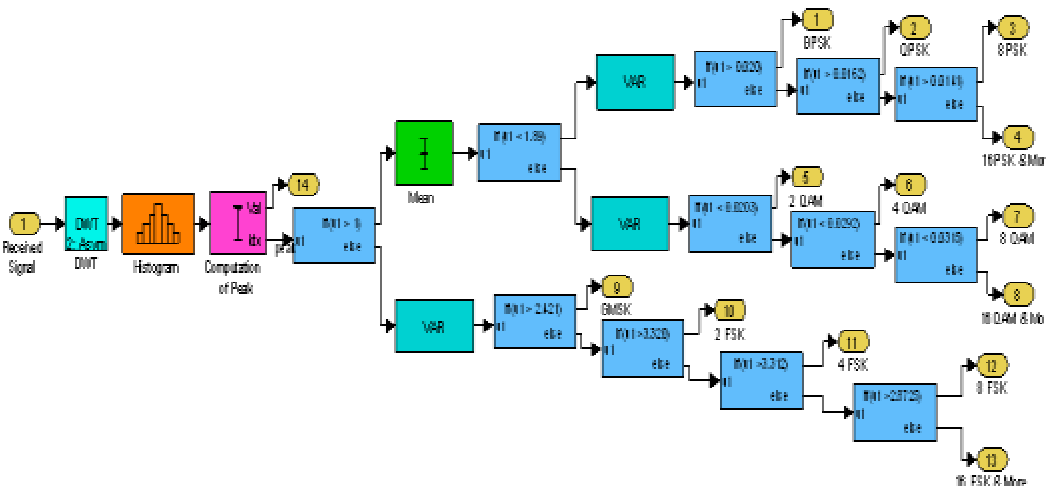


*Fig. 8:* *Simulink Model of the Proposed Algorithm [4].*

But this process requires large computations for detecting the phase changes, difference phase and instantaneous frequency of each frame which increases the time for detection of modulation technique and hence increases loss of bits. Also, it requires a large memory space for storing all the parameters for modulation detection, which is not feasible in practical scenarios.

Four parameters of the received signal are first computed and then compared with the stored parameters. This process is repeated for every frame received. It is a cumbersome process and requires very fast calculations in real time scenarios. Any miss match in synchronization will result in complete loss of information.

In the final method based on wavelet transform, wide variety of modulation schemes can be distinguished. Each technique has a distinct characteristic (phase, frequency, or amplitude) using which the discrete wavelet transform (DWT) can differentiate between modulation schemes. Other methods for modulation recognition do not provide such wide range of options in distinguishing the received modulation scheme.

Also, the flow of the process is simple and easy to understand and implement. The decision is dynamically taken for every frame. If the modulation scheme received changes, the receiver will automatically change the demodulator in the subsequent frame. Hence, the bit loss is minimal. The results at 12 dB SNR give a very high accuracy rate. The delay in selection of the scheme is negligible.
Disadvantages of this technique include the variation in the mean and variance values for different modulation schemes which may not obey threshold rule for low frame size, hence introducing detection error for the entire frame. If the threshold rule is not obeyed then the decision taken for the detection of the modulation technique is wrong for the entire frame, so the received bit stream is fed to the wrong demodulator. Thus, for all frames that do not obey the threshold rule, there will be an appreciation in the bit error rate. The system performance degrades rapidly at an SNR less than 12 dB.

## 7. CONCLUSIONS

Different techniques are studied and a comparison is drawn. The wavelet transform stands out as the most efficient method. The performance of the system is 100% at 12 dB SNR for all the modulation schemes involved. The results are better than other techniques and there is no restriction of the number of modulation schemes that can be detected, i.e., more modulation schemes can be added as per requirement. The design can be easily implemented on reconfigurable platforms like FPGA. The method detects a wide range of modulation schemes, which is a fetter in other schemes. Hence, the technique based on wavelet transform is the best amongst the ones studied and will be implemented by using the required software in future.

## REFERENCES

1. Sami H. O. Salih, Mamoun M. A. Suliman. *53rd International Symposium ELMAR-2011*. Sudan University of Science and Technology. Zadar, Croatia. 14–16 September 2011.
2. M. Vastram Naik, A. Mahata, R. Bhattacharjee et al. *International Conference on E-Business and Telecommunicalion Networks*. ICETE ‘05, UK. 2005.
3. Ferdinand Liedtke. *Journal of Telecommunications and Information Technology*.2004.
4. P. Prakasam and M. Madheswaran. *International Journal of Computer Theory and Engineering*. 1(5). December 2009.
5. P. Prakasam and M. Madheswaran. *IEEE–ICSCN*. 2007. MIT Campus, Anna University, Chennai, India. February 22–24, 2007. 507–511p.
6. P. Prakasam and M. Madheswaran. *International Journal of Information and Communication Engineering*. 5(1). 2009.