International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 1 Issue 9, November- 2012

A Review On Adaptive Modulation Recognition Techniques

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Abstract: Software defined radio is the booming technology in the modern era. The main focus of the SDR is to decode/demodulate the received signal. In the modern transmitter the signal coding/modulation changes with the present transmitting situations. The receiver must be able to demodulate any received signal irrespective of the information from the transmitter about the modulation scheme being transmitted. This technique is a blind modulation technique. The paper discusses such basic techniques for adaptive modulation detection. The pros and cons of all the techniques are listed and final conclusion based on the same is drawn.

Key words: FSK, QAM, Adaptive Modulation, SNR

Introduction

The heart of the SDR system is adaptive modulation recognition (AMR). AMR involves estimation of any modulation scheme at the receiver generally followed by its demodulation using relevanttechnique. 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. Adaptive modulation systems improve rate of transmission, and/or bit error rates, by exploiting the channel information that is present at the transmitter. Especially over fading channels which model wireless propagation environments, adaptive modulation systems exhibit great performance enhancements compared to systems that do not exploit channel knowledge at the transmitter.

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. AMR is used in wide range of civil and military applications such as spectrum surveillance, threat evaluation, interference identification mobile communication etc. The basic techniques for the AMR are discussed here with their advantages and disadvantages.

Feedback Based Technique

In the feedback based 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. It is diagrammatically explained in the figure 1 shown below.

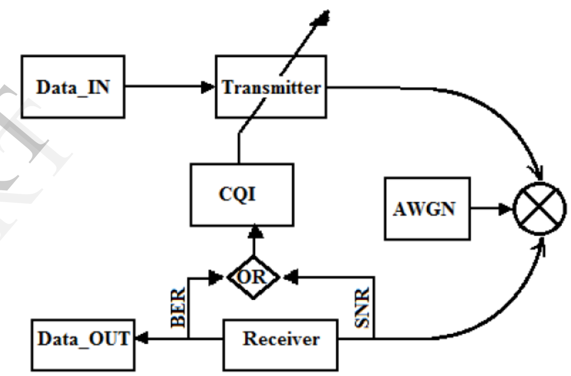

Figure 1Adaptive System [1]

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 Mean Square Error is calculated on whose basis appropriate 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-thsignaling interval may be expressed as as

$$\mathbf{r(t, k) = Sm(t, k) + n(t, k)}$$
$$\mathbf{kTs < t < (k + 1)\ Ts}$$

whereTs is symbol duration, Sm(t) is the message waveform corresponding to the M-PSK symbol Sm, m=1,2,3,…M. Assuming perfect carrier synchronization and timing recovery, we employ I-Q demodulation to get

r(k)=[rI (k), rQ(k)] =[smI + nI(k), smQ + nQ(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:

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

$$Where, D_{kM} = min\{|r(k)s_m|\}, m = 1,2,..M$$

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).

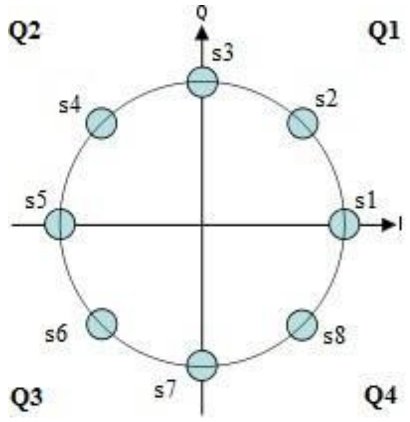

Figure 2. Constellation Diagram

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) ϵ 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) ϵ 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) ϵ 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.

Similarly such distribution can be found out for MSED4-8, 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.

Least Mean Square Error Difference Based Technique
The third technique studied is based on least mean square error difference in amplitude, phase and frequency. This procedure has an ability to adapt to mostly all possible schemes dynamically and the ability to recognize even continuous phase modulation (CPM) signals like Gaussian Minimum Shift Keying (GMSK) too. The necessary pre-processing of the received signal and the signal itself is shown in Figure 3.. 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.

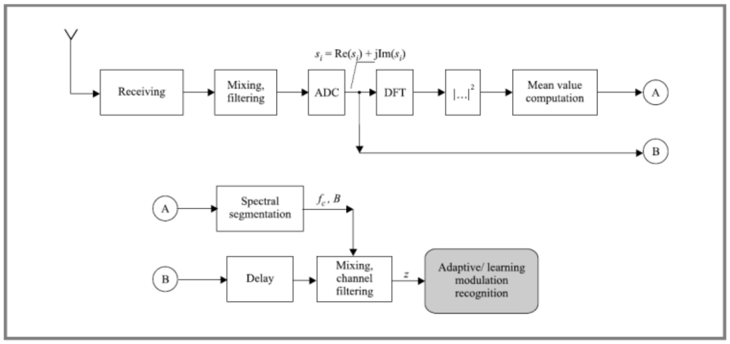

Figure 3. Signal Analysis with Modulation Recognition [3]

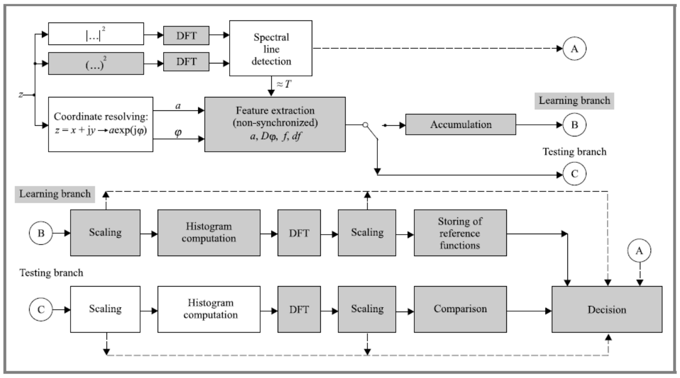

Figure 4.Adaptive modulation recognition, non-synchronized [3]

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 4 shows the 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 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 Dω within the third branch of the feature extraction module as shown in Figure 4. The coordinate resolving model finds the magnitude 'a' and phase ω 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 Dω, instantaneous frequency f and derivative of the instantaneous frequency *df*. The parameter T is used for calculating every pair of phase differences Dω. 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 the 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 theyhave 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 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 5 indicate the intention to search (in future) additional features of the wave using symbol rate recognition. The Figure5 shows the feature extraction module which extracts relevant parameter values for a, f, *df and* D*ω.*

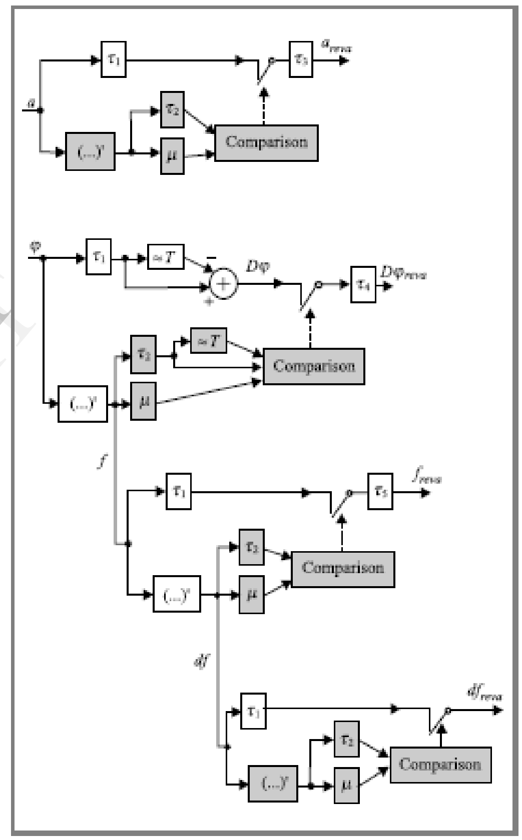

Figure 5.Feature extraction, non-synchronized [3]

In this technique 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. 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

ISSN: 2278-0181
Vol. 1 Issue 9, November- 2012

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 required very fast calculations in real time scenarios. Any miss match in synchronization will result in complete loss of information.

Conclusion

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's not practical for the receiver to send channel state information to the transmitter via feedback. Hence this technique is ruled out for practical implementation. 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. The best of the discussed method is least mean square error difference based technique where with low SNR good results can be obtained but large database or memory is require.

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