Modulation Scheme Classification for 4G Software Radio Wireless Networks

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***Abstract*-This paper describes software radio and a software radio technique for automatic digital modulation scheme recognition. The modulation recognition technique is designed for a real-time software radio using general-purpose processors and is based on a modified pattern recognition and phase classifier approach. The techniques presented in this paper are designed to be robust and efficient with a processing time overhead that still allows the software radio to maintain its real-time operating objectives. Experimental results for GMSK recognition over a fading, ISI channel with AWGN and random phase noise are presented. One test scenario, where an intercepted signal is either BPSK or QPSK modulated, using an AWGN channel with random phase noise shows that correct modulation scheme classification is possible to a lower-bound channel SNR of approximately 9dB.**

1. INTRODUCTION

There has been a dramatic move towards digitising RF signals as close as possible to the antenna, thus replacing analogue components with Digital Signal Processors. Wireless systems are gravitating towards minimal radio hardware designs using flexible architecture software radio. This method can greatly simplify the design of a radio system since typical hardware radio components are replaced by software designs. Future generations of wireless networks will take advantage of software realizations of hardware devices because of the highly re-configurable nature of such systems. For example, a hardware implementation of a FM receiver typically uses an integrated circuit to perform signal demodulation. Conversion of this receiver to some other modulation scheme e.g. AM, requires a physical change to the circuit, which can be difficult or even impossible. Two separate receivers, one AM and one FM, are therefore required. This is an expensive and space-consuming exercise. The alternative is to convert the hardware demodulation functions to software realizations. Conversion of this software receiver to other modulation schemes is greatly simplified because a change in the software algorithm is all that is required. The radio hardware is simplified because a common hardware RF front-end may be used. As outlined in Figure 1, this RF front-end amplifies the signal from the antenna, performs signal filtering if required and then digitizes the filtered received signal using an analogue to digital converter. Once the received signal has entered the software-processing core, passband to base-band conversion and the rest of the radio functions are software processes.

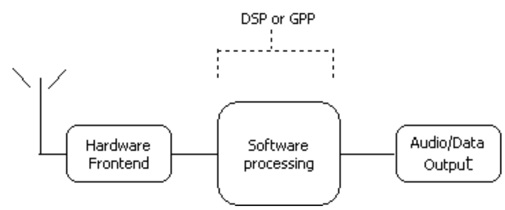

Fig. 1 Overview of a software radio system

The signal-processing engine can either be a dedicated DSP or a General-Purpose Processor (GPP). DSP implementations offer fully deterministic function execution times, usually high code optimization options (optimization by hand is possible for optimal performance) but can be very expensive, lack the robustness required for mobile work and are have highly constrained memory capabilities for waveform storage and buffering. The approach presented in this paper has been to use General Purpose Processors (GPPs) instead of dedicated DSP hardware. General-purpose processors offer large amounts of program and user memory, floating-point operation, better high-level language development environments and are less expensive compared to dedicated DSPs. The rapid development in processor technology has meant that the performance of some GPPs outstrips that of dedicated DSPs. According to a recent processor speed survey carried out by Berkeley Design Technology, a Pentium III processor outperforms the Texas Instruments TMS320C55xx, TMS320C62xx, TMS320C67xx, Analog Devices ADSP-21xx/219xx, ADI-Intel MSA/Analog Devices ADSP-2153x and the Motorola DSP563xx/DSP568xx. [1]

Software radio technology is expected to play an important role in the development of Fourth Generation (4G) wireless communication systems. Software radio and software-defined radio allows a very flexible radio management system resulting in a gain in radio resource and spectrum utilization. One major advantage resulting from re-configurable radio terminal deployment is that as the network topology changes, the terminal can adapt to the users requirements, change modes of operation, channels, access methods and request new software upgrades if required without user intervention. Future wireless communication systems will no longer be static, but each individual node in the network will have to acquire a dynamic route to the intended destination. A dynamic network such as this is not feasible with existing fixed architecture radio systems.

Automatic selection of the correct modulation scheme used in an unknown received signal is a major advantage in a wireless network. As the channel capacity varies, switching between modulation schemes enables the baud rate to be increased or decreased in order to maximise the channel capacity usage and reduce the bit error rate (BER). However, finite processor power constrains the complexity of a software radio if real-time objectives are to be met. This paper proposes an automatic digital modulation scheme recognition technique adapted for general-purpose processor (GPP) based real-time software radio. The basic design outline is shown in figure 2. We consider the cases of BPSK and QPSK using an Additive White Gaussian Noise (AWGN) channel with random phase noise, and Gaussian Minimum Shift Keying (GMSK) over an intersymbol interference (ISI) channel subjected to Doppler, multipath propagation effects, random phase noise and AWGN.

Automatic modulation scheme recognition is a desirable application for a software radio system. Using or combining pattern recognition, cluster analysis, phase and square law modulation scheme classifiers, it enables correct classification of a received signal without apriori modulation scheme knowledge. Using this knowledge, the correct demodulation scheme can be selected automatically. A software receiver implementation facilitates a much more flexible and relatively inexpensive application design. This is important for dynamically changing the function of the radio, and for reacting to changes in the intercepted signal such as a change of modulation scheme employed, for example. Signal space representation of quadrature components of an intercepted signal is a graphical means of monitoring channel quality variations. Constellation diagrams are commonly used to assess the underlying signal structure of a signal. The position and spread of the received signal point clouds are affected by the channel signal to noise ratio (SNR), channel distortion resulting in ISI and fading due to a moving signal source or receiver. The mean excursion of received signal points about ideal signal points on a constellation diagram may be used as a metric to determine whether the employed modulation scheme can be supported over the time-varying channel.

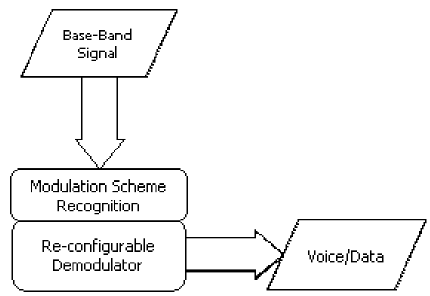

Fig. 2. Overview of a re-configurable demodulator using automatic modulation scheme classification

This paper is presented as follows: section 2 outlines some of the background research in this area. In section 3, the QPSK and GMSK signal models used are described. The modulation scheme classifier is described in section 4, with an overview of its implementation in section 5. Test results are summarized in section 6.

2. BACKGROUND

Modulation scheme recognition is increasing in importance and extensive work has been carried out for a number of years. For software radio realizations, robust but lightweight techniques are required in order to minimize the resulting processing time overhead. Recently published modulation recognition algorithms include a decision-theoretic approach and pattern recognition approach used to discriminate between digitally modulated signals [2]. Modulation scheme recognition using the signal envelope extraction method was described by Druckmann, Plotkin, and Swamy. [3]. Lopatka and Pedzisz also adopted this approach incorporating fuzzy classification for 4DPSK, 16QAM and FSK schemes [4]. Experimental results showed that the lower-bound SNR for correct identification was 5dB but the sampling frequency considered was 8kHz. Callaghan, Pery, and Tjho proposed a signal envelope and zero crossing based modulation recognizer**.** [5], but the accuracy of the recognizer is highly dependent on determining the exact intercepted signal centre frequency. This recognizer is capable of recognizing amplitude modulation (AM), carrier wave (CW), frequency shift keying (FSK) and frequency modulation (FM) but requires

SNR ≥ 20dB for correct modulation scheme recognition. A pattern recognition approach for both digital and analogue recognition was proposed by Jondral [6], which can classify AM, ASK2, SSB, PSK2, FSK2 and FSK4 modulation scheme types. Key features of the received signal; instantaneous amplitude, frequency and phase are used to discriminate between modulation scheme types. Hellinger distance parameter estimation is a modulation classification technique that can automatically overcome a moderate degree of noise model distortion, with improved robustness and efficiency [7][8][9][10]. The technique proposed in [11] is a digital modulation scheme classifier based on a pattern recognition technique used in binary image word spotting problems. This classifier generalizes the moment matrix technique to grey scale images, and the technique is used to discriminate between M-ary PSK and QAM signal space constellations. Aiello, Grimaldi and Rapuano proposed an artificial neural network (ANN) based classifier due to the pattern recognition qualities of ANN [12]. This was implemented using a dedicated DSP and focused on GMSK recognition. Polydoros and Kim proposed a BPSK/QPSK classifier named the quasi Log-Likelihood ratio [13], derived by approximating the likelihood-ratio functionals of phase-modulated signals in white Gaussian noise and reported on ‘per-survivor processing’ for unknown ISI environments [14]. These techniques were reported to be significantly better than conventional square-law classifier or phase-based rules. Real-time modulation scheme recognition techniques have been proposed by Boudreau, Dubuc, Patenaude, Dufour, Lodge and Inkol using dual processors [15] and also by Hsue, Soliman [16] using three processors and parallel processing approaches. This paper builds on the work carried out by Nolan, Mackenzie, Doyle and O’Mahony [17].

3. SIGNAL MODELS

Firstly, we choose to define the QPSK signal model but consider both BPSK and QPSK cases for the experimental work.

Let $y(t)$ be the received passband signal where

$$y(t) = a_n A\cos(\omega_0 t + \theta) + b_n A\sin(\omega_0 t + \theta) + n(t) + \varphi(t)$$

(1) where $\omega_0 = {}^{2\pi}\!/_{Ts}$, Ts is the sampling period of the ADC, A is the amplitude of the received signal, $[a_n, b_n]$ is the set of transmitted symbols, θ is the unknown phase of the received signal and $P[a_n = +1] = P[a_n = -1] = P[b_n = +1] = P[a_b = -1] = \frac{1}{2}$

$$y(t) = y_1\varphi_1(t) + y_2\varphi_2(t) \quad \text{where} \quad \varphi_1(t) = \sqrt{\frac{2}{T_S}}\cos(\omega_0 t)$$

and $\varphi_2(t) = \sqrt{\frac{2}{T_S}}\sin(\omega_0 t)$ (2)

The channel model is a standard AWGN channel with a two-sided PSD of $N_0/2\frac{W}{H_Z}$ and also subjected to random phase noise $\varphi(t)$, where $\varphi(t)$ is uniformly distributed in the range $[0,2\pi]$.

Several wireless telecommunications systems use the GMSK modulation scheme, i.e. GSM European cellular telephone system and CDPD (Cellular Digital Packet Data) in the USA [18]. For Continuous Phase Modulation (CPM), let $s(t)$ denote the modulated signal, where $s(t)$, during the i^{th} bit time as

$$s(t) = \sqrt{\frac{2E_s}{T_s}} Cos\left[\omega_c t + \frac{a_i[t-(i-1)T_s]\pi h}{T_s} + \pi h\sum_{j=1}^{i-1} a_j + \theta_0\right]$$

(3)

for $(i-1)T_s \le t \le iT_s$, where a_i is the i^{th} data element, T_s is the symbol period and h is the modulation index. This signal has continuous phase at the symbol boundaries for any value of h. The continuity of phase introduces memory into the system, which can result in intersymbol interference. Demodulation entails the examination of a sequence of received symbols using maximum likelihood techniques (MLSE) or the Viterbi algorithm rather than bit-by-bit demodulation.

GMSK is a CPM signal with an infinite width pulse $g(t)$ given by

$$g(t) = \frac{1}{\sqrt{2\pi}\,\delta T_s}\exp\left\{-\frac{t^2}{2\delta^2 T_s^2}\right\} \otimes \frac{h\pi}{T} rect\left(\frac{t}{T_s}\right) \quad (4)$$

$$= \frac{h\pi}{2T}\left[erfc\left(\frac{{}^{t}\!/_{T_s} - {}^{1}\!/_{2}}{\delta\sqrt{2}}\right) - erfc\left(\frac{{}^{t}\!/_{T_s} + {}^{1}\!/_{2}}{\delta\sqrt{2}}\right)\right] \quad (5)$$

where $h = 0.5, \delta = \sqrt{\ln 2}/(2\pi BT_s)$, ($BT_s = 0.3$, typically), and $erfc(\cdot)$ is the complementary error function defined by

$$erfc(x) \equiv \frac{2}{\sqrt{\pi}}\int_x^{+\infty} e^{-u^2} du. \quad (6)$$

In the case of multi-path propagation, Doppler and fading effects, the signal model to be considered is

$$s(t) = \sum_n A_n Cos\left[\omega_c t + \frac{a_i[t-(i-1)T_s]\pi h}{T_s} + \pi h \sum_{j=1}^{i-1} a_j + \theta_0 + \varphi_n + 2\pi \wedge f_n t\right]$$

(7)

where $\wedge f_n = \pm \frac{v}{\lambda} \cos \alpha_n$, with v being the speed of the mobile station and λ the signal wavelength, φ_n the phase uniformly distributed in the range $[0, 2\pi]$, A_n the signal amplitude evaluated according to the Rayleigh distribution. The signal is also subject to standard AWGN with a two-sided PSD of $N_0/2 \frac{W}{H_Z}$.

We now proceed to describe the modulation scheme recognition classifier

4. DIGITAL MODULATION SCHEME RECOGNITION

The amplitude and phase information of a set of received signal points is commonly represented graphically on a constellation diagram. Constellation diagrams are used for digitally modulation schemes where each transmitted symbol has a specific phase and/or amplitude value. These schemes range from BPSK to 768-QAM (V.34), and beyond. Once the actual modulation scheme details have been obtained, one method used to monitor channel quality and estimate channel capacity is to calculate the statistical moments of the received signal point variations about the expected constellation diagram. For channels with relatively high signal to noise ratios (SNR), the signal-space approach allows the obvious features of the received signal points to be measured, mainly the amplitude and phase variations of the signal.

A. Normalisation and Matrix Construction

Consider the case where all possible transmitted symbols are equiprobable Let $c_{I_{id}}$ and $c_{Q_{id}}$ denote the outputs of the integrate and dump section of a Costas Loop demodulator. Symbol timing is estimated using early-*late timing recovery* where the sampling time τ_n is sequentially adjusted using an algorithm that may be described as $\tau_{n+1} = \tau_n + \alpha[k(z_n, \tau_n)]$ (8) where α is a constant step size, $E[k(z_n,\tau_n)] = \frac{\partial[\ln Q(\tau)]}{\partial \tau}$ (9), where

$$Q(\tau) = E\left\{\exp\left[-\frac{1}{\eta}\int_0^{NT_s}\left(\left[s(t) - \sum_n a_n h(t - nT_s + \tau)\right]\right)^2 dt\right]\right\}$$

(10)

The variates $c_{I_{id}}$ and $c_{Q_{id}}$ are scaled in order to improve the numerical stability of the moment matrix calculations. The set of N demodulated signal points $c_{I_{id}}[1,\ldots N] - jc_{Q_{id}}[1,\ldots N]$, is replaced by $\frac{K\left|c_{I_{id}}[1,\ldots N] - jc_{Q_{id}}[1,\ldots N]\right|}{Max\{Mag\{c_{I_{id}}[1,\ldots N] - jc_{Q_{id}}[1,\ldots N]\}\}}$, (11) where K denotes the scaling factor.

Since the transmitted symbols are equally likely, and the expected signal space representation of the received signal points is approximately symmetrical, only one side of the signal space diagram is used for modulation scheme classification as shown in Fig.1. Reducing the number of received signal points reduces the processing time required for modulation scheme at the expense of data for signal classification.

identity estimation. The valid set of signal points is reduced to $\mathrm{T} = \{\Re\{c_{I_{id}}[1,\ldots N] - jc_{Q_{id}}[1,\ldots N]\} \geq 0\}$ where $T = T[\hat{N}]$. (12)

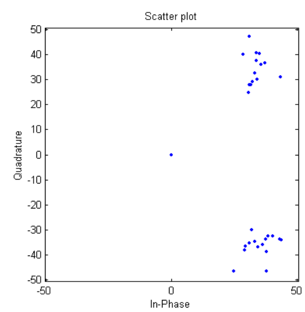

Fig. 3. Reduced Form Constellation Diagram for a QPSK modulated signal with AWGN (SNR 39 dB)

B. Moment Matrix Calculations

The eight-order statistical moment I_{oxyz} is the sum of the moments about the point [$x = Max(\Re(T))$, $y = 0$] of the signal space diagram representation of received signal point clouds. I_{oxy} is defined by $I_{0xy} = I_{0x} + I_{0y} = 2\sum m_i r_i^8$ (13) where $m_i = 1$ for all received signal points and $r_i = mag(T_{i,j})$.

5. IMPLEMENTATION

Automatic modulation scheme detection was implemented as part of a software radio system implemented on a general- purpose processor platform. Fig. 2 shows a basic block diagram of a software radio receiver illustrating how the recognition section is integrated into the radio. The software radioprocessing core uses a sequential processing block approach; each layer can be added or removed as required resulting in very flexible software radio architecture. The software radio is a Win32 application, and runs on a single 700MHz Pentium III processor.

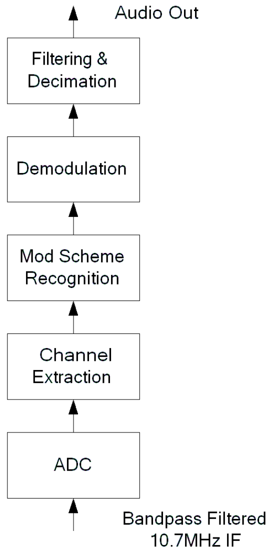

Fig. 4. Software receiver basic layer structure concept

The radio hardware is comprised of the antenna, RF amplifier, IF converter stage and bandpass filter. A 20MHz 12-bit ADC performs bandpass sampling of the amplified IF signal from the radio hardware. Bandpass sampling, where the received signal is not centered about zero Hz, not only reduces the speed requirement of ADCs below that necessary with low pass sampling, digital memory required to capture a given time interval of a continuous signal is also reduced. Similar to a communications protocol stack, the structure of the software radio can be reconfigured by adding/removing function layers [19]. The bottom software radio layer is the link to the air-interface (ADC) and the top of the software radio stack is connected to the main communications stack or to an audio output layer. Data transfer from the ADC PCI card FIFO to PC memory is performed using a circular buffer structure. Data sample blocks are passed to the processing layers using Windows messaging functions. Each block (or layer) runs as a separate execution thread and only processes the message data payload passed to the layer. This is a flexible radio architecture allowing dynamic addition or removal of stack components. From the experimental work, it was found that, for real-time receiver operation using the standard PCI bus, it is necessary to decrease the ADC sampling rate to 2MHz resulting in an under-sampled IF signal. An additional bandpass filter is required after the ADC to reduce aliasing. Data processing ‘in place’ using optimized signal processing library functions is required to minimize excessive data movement within program memory and thus reduces the overall software radio processing time.

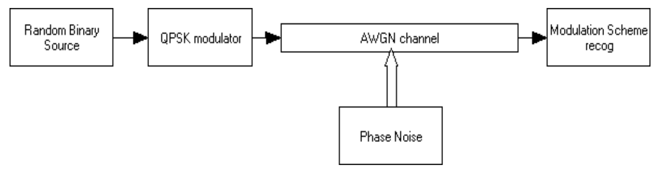

Fig.5. QPSK signal model (AWGN channel with random phase noise)

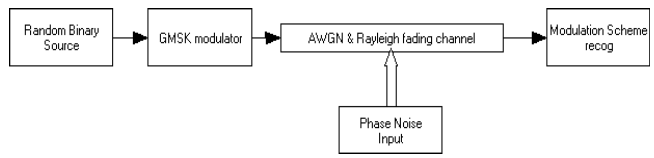

Fig. 6. GMSK signal model (Rayleigh fading, AWGN channel with random phase noise)

6. Experimental results

In this section, we firstly focus on the QPSK and BPSK tests and simulations. The test transmitter and channel model for the QPSK case is shown in Fig.3. For both cases, the channel SNR is reduced from 100dB to 10dB and the random phase noise input has a power of 0.01W. Fig. 5 is a graph of the 8^{th} order statistical moments versus the channel SNR for a sample set of 100 received signal points. Using a threshold based decision metric, if the intercepted signal is of either class, then the moment method can identify the scheme used to a lower-bound SNR of approximately 9dB.

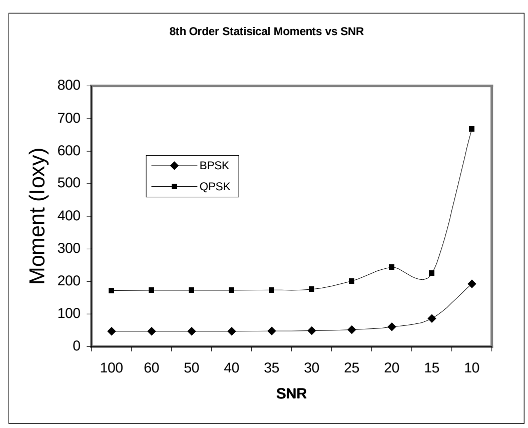

Fig. 7. Graph of the channel SNR versus the 8^{th} order Moment Ioxy

We now consider the case where the intercepted signal is a GMSK scheme. The transmitter and channel model is outlined in Fig. 4. Intersymbol interference significantly affects the received signal point distribution and therefore is expected to significantly reduce the effectiveness of the proposed moment-based classifier. For purposes of multipath propagation and Doppler effects measurements, this signal model was simulated using MATLAB.

The simulations carried out consider the cases where the transmitted pulse is spread out over two symbol periods, three symbol periods and finally, over four symbol periods. As for the previous case, the channel SNR is reduced from 100dB to 10dB and is also subjected to random phase noise and Rayleigh fading.

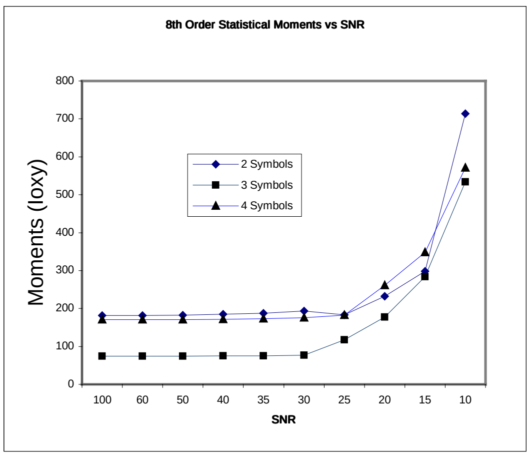

Fig. 8. Graph of the channel SNR versus vs. the 8th order Moment for GMSK over a fading, AWGN, ISI channel with random phase noise. Three cases are tested: where the transmitted pulse is spread over two, three and four symbol periods.

From the experimental results shown in fig. 6, the level of intersymbol interference significantly affects the statistical moment measurements. For high SNR channels, the moment measurements are approximately constant but as the channel SNR approaches 20dB, the classifier begins to fail.

7. CONCLUSIONS

This paper described an automatic modulation-scheme recognition technique for use in a software radio system implemented on a general-purpose processor platform. The technique is capable of determining the modulation scheme of a signal in the case when the digital modulation scheme class is not known *a priori* but is either BPSK or QPSK schemes. Tests and simulations involving an intercepted signal, which is modulated using BPSK or QPSK, show that correct modulation scheme identification is possible even at a low channel $\mathrm{SNR} \approx 9dB$. Tests using the GMSK scheme show that the classifier is significantly affected by pulse spreading over ISI channels.

Further work on this technique will expand the range of identifiable digital modulation schemes and increase the robustness of the classifier to increase its effectiveness for low SNR channels.

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