Software Radio Modulation Scheme Recognition Techniques For ISI Channels

Keith E. Nolan, Linda Doyle, Philip Mackenenn, Max Ammann

Networks and Telecommunications Research Group, Trinity College, Dublin 2, Rep. of Ireland
1 DIT, Kevin St., Dublin 8, Rep. Of Ireland.

Abstract—This paper describes an automatic digital modulation scheme recognition technique. The technique is designed for a real-time software radio using general-purpose processors and is based on a modified pattern recognition approach. It is 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

The advent of realizable software radio allows the implementation of creative transceiver designs, which can dynamically adapt to the communications channel and user applications. Instead of dedicated hardware designed to carry out a rigid set of objectives, software implementations of hardware devices are entirely flexible regarding their functionality. An ideal use for this flexible architecture is in the area of wireless networks, where a node may adapt to its environment and user objectives. Used in a packet-switched wireless network, software radio systems can perform multi-mode modulation and demodulation on a per-packet basis offering greater control over spectrum usage and minimizing the need for dedicated hardware. At the physical level, flexible transceiver architectures enable greater maximization of channel capacity. A software radio enables dynamic channel adaptation using standard adaptive equalization techniques and allows the implementation of new creative algorithms but with the need for dedicated hardware reduced or eliminated.

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. We consider the case of Gaussian Minimum Shift Keying (GMSK) over an intersymbol interference (ISI) channel.

The GPP approach makes use of x86 chipsets for rapid application development, large amounts of program memory, and is relatively inexpensive compared to inflexible dedicated hardware. Conventional DSP processors rely on assembly language optimization for maximization of application efficiency, but with the x86 chipset, the degree of optimization possible using high-level languages such as C and C++ is much greater [1]. Software radio system performance closely tracks the advances in new higher speed processor technology allowing the dynamic addition of more complex signal processing techniques to the software radio system.

Automatic modulation scheme recognition enables correct classification of a received signal without apriori modulation scheme knowledge enabling the selection of the correct demodulation scheme. 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.

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-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 MODEL

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

\[
y(t) = a_n A \cos(\omega_0 t + \theta) + b_n A \sin(\omega_0 t + \theta) + n(t) + \phi(t)
\]

where \( \omega_0 = 2\pi/T_s \), \( T_s \) 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, \( \theta \) is the unknown phase of the received

\[
P[a_n = +1] = P[a_n = -1] = P[b_n = +1] = \frac{1}{2}
\]

\[
y(t) = y_1(t) + y_2(t) \quad \text{where} \quad y_1(t) = \frac{2}{T_s} \cos(\omega_0 t) \quad \text{and} \quad y_2(t) = \frac{2}{T_s} \sin(\omega_0 t)
\]

The channel model is a standard AWGN channel with a two-sided PSD of \( N_0/2W \) 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

\[
s(t) = \frac{2E_s}{T_s} \cos \left( \omega_0 t + \frac{a_i [t - (i-1)T_s] \theta_h}{T_s} + \sum_{j=1}^{i-1} a_j + \phi_0 \right)
\]

for \( (i-1)T_s \leq t \leq iT_s \), 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

$$ g(t) = \frac{1}{\sqrt{2\pi\sigma^2_t}} \exp\left\{-\frac{t^2}{2\sigma^2_t}\right\} \otimes \frac{h\pi}{T} \text{rect}\left(\frac{t}{T_s}\right) $$

(4)

$$ = \frac{h\pi}{2T} \left[ \text{erfc}\left(\frac{t/T_s - 1/2}{\delta/2}\right) - \text{erfc}\left(\frac{t/T_s + 1/2}{\delta/2}\right) \right] $$

(5)

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

$$ \text{erfc}(x) = \frac{2}{\sqrt{\pi}} \int_x^{\infty} e^{-\eta^2} d\eta. $$

(6)

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

$$ s(t) = \sum A_n Cos(\omega t + \frac{1}{\lambda} t^2 - (i-1)T_s + \varphi) + \eta $$

(7)

where $h = \pm \frac{\nu}{\lambda} \cos \varphi_n$, with $\nu$ being the speed of the mobile station and $\lambda$ the signal wavelength, $\varphi$ 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 $N_0/2W$.

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 $c_{id}$ and $C_{id}$ 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 $\tau_n$ is sequentially adjusted using an algorithm that may be described as $\tau_{n+1} = \tau_n + \alpha[k(z_n, \tau_n)]$ (8) where $\alpha$ is a constant step size, $E[k(z_n, \tau_n)] = \frac{\partial[\ln Q(\tau)]}{\partial \tau}$ (9), where

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

(10)

The variates $c_{id}$ and $C_{id}$ are scaled in order to improve the numerical stability of the moment matrix calculations.

The set of N demodulated signal points $c_{id}[1...N] - jC_{id}[1...N]$, is replaced by

$$ K\left[ c_{id}[1...N] - jC_{id}[1...N] \right] / \text{Max}\{\text{Mag}[c_{id}[1...N] - jC_{id}[1...N]]\} $$

(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

$$ T = \left\{ c_{id}[1...N] - jC_{id}[1...N] \right\} $$

(12)
B. Moment Matrix Calculations

The eight-order statistical moment $I_{oxyz}$ is the sum of the moments about the point $[x = \text{Max}(\Re(T)), y = 0]$ of the signal space diagram representation of received signal point clouds. $I_{oxyz}$ is defined by

$$I_{oxyz} = I_{ox} + I_{oy} = 2 \sum m_i r_i^a$$

where $m_i = 1$ for all received signal points and $r_i = \text{mag}(T_{i,j})$.

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 radio processing 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.

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.
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 8th 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.

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.

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.

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.

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 SNR = 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.

ACKNOWLEDGMENTS

K. E. Nolan thanks Dr. Anil Kokaram, Trinity College Dublin, for his helpful comments regarding statistical moments.

REFERENCES


