Signal Space based Adaptive Modulation for Software Radio

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***Abstract*-This paper describes an automatic modulation scheme recognition technique. The technique is designed for a real-time software radio using general-purpose processors and is based on modified pattern recognition and signal space approaches. It is robust and efficient with a processing time overhead that still allows the software radio to maintain its real-time operating objectives. Both digital and analogue modulation schemes can be identified. Tests and simulations using an AWGN channel show that the SNR threshold for correct analogue modulation scheme classification is approximately 6.5 dB.**

I. INTRODUCTION

The advent of realisable 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 minimising the need for dedicated hardware. At the physical level, flexible transceiver architectures enables greater maximisation of channel capacity. A software radio enables dynamic channel adaptation using standard adaptive equalisation techniques and allows the implementation of new creative algorithms but with the need for dedicated hardware reduced or eliminated.

This paper describes one application that exploits the flexibility of a software radio. The ability to automatically select the correct modulation scheme used in an unknown received signal is a major advantage in a wireless network. As the channel capacity varies, modulation scheme switching enables the baud rate to be increased or decreased thus maximising channel capacity usage. However, finite processor power limits the complexity of a software radio if real-time constraints are to be met. This paper proposes an automatic analogue modulation scheme recognition and scheme-switching algorithm adapted for general-purpose processor (GPP) based real-time software radio.

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 optimisation for maximisation of application efficiency, but with the x86 chipset, the degree of optimisation 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 addition of more complex signal processing techniques to the software radio system, while maintaining real-time objectives.

Automatic modulation recognition and scheme switching enables correct demodulation of a received signal without apriori modulation scheme knowledge. 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 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.

In section II, an outline of the existing modulation scheme recognition techniques is given. Section III describes a real-time analogue modulation scheme recognition technique developed and tested as part of the software radio, following on to define the digital modulation test signals used. Section IV outlines the operation of signal demodulation in software. Section V describes the proposed digital modulation scheme recognition algorithm. Section VI deals with the implementation issues, with results detailed in section VII.

II. BACKGROUND

Work in automatic modulation recognition has been carried out for a number of years, producing processor intensive techniques mainly restricted to non real-time operation. 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 by the signal envelope extraction method was described by Druckmann, Plotkin and Swamy [3]. A signal envelope and zero crossing based modulation recogniser was proposed by Callaghan, Pery, and Tjho**.** [4], but the accuracy of the recogniser is highly dependent on determining the exact intercepted signal centre frequency. This recogniser is capable of recognising 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 [5], 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 [6][7][8][9]. The technique proposed in [10] is a digital modulation scheme classifier based on a pattern recognition technique used in binary image word spotting problems. This classifier generalises the moment matrix technique to grey scale images, and the technique is used to discriminate between M-ary PSK and QAM signal space constellations. The technique described in [10] is processor intensive due to its image processing application and is more suited for offline work.

A successful algorithm for a software radio implementation must be robust and efficient but the processing overhead must not stop the software radio from maintaining its real-time operating objectives. As the techniques described here are in general very processor intensive, they are not suitable for software radio systems.

III. MODULATION SCHEME RECOGNITION

This modulation scheme recognition technique proposed in this paper builds on the work carried out on pattern recognition techniques and the signal space partitioning idea used in the Hellinger distance approach. The proposed technique addresses the problems of maintaining robustness against channel disturbances without introducing a significant processing overhead to the system. It is designed for sparse matrices of signal points, greatly reducing the required processing time.

This technique is a modified moment-based modulation recognition algorithm, designed for robust real-time software radio operation using a GPP. This technique incorporates signal plane partitioning, counting of received signal points within cells and pattern recognition approaches. The result is an improved technique with a minimal processing time overhead that still allows the software radio to maintain its real-time objectives.

The approach involves two main steps; determining whether the intercepted signal is modulated by an analogue or digital modulation scheme, and secondly, obtaining the specific modulation scheme parameters used.

A. *Differentiation between Analogue and Digital Modulation Schemes*

An intercepted signal may be one of two general classes: modulated using an analogue source or by using a digital modulation scheme. A measure of the frequency content of the signal enables an estimate of which general modulation class it belongs to. In general, the power spectral density (PSD) of a signal employing a digital modulation scheme will uncover power density peaks characteristic of fixed frequency offset and/or fixed phase values used. An audio modulated signal, for example, will generally result in a more even PSD distribution.

Let $\gamma_{\max f}$ be the maximum value (measured in dB) of the PSD of the normalised instantaneous frequency of the intercepted signal and is defined by:

$$\gamma_{\max f} = 20\log_{10}(\max\left|DFT(f_n(i))\right|) \qquad (1)$$

If $\gamma_{\max f} > t\gamma_{\max f}$, the received signal is an ASK2, ASK4, PSK2, or PSK4 modulated signal, otherwise, it is classed as an analogue modulation scheme signal [11].

B. *Analogue Modulation Scheme Recognition*

This second part of this section outlines how an estimate of what the modulation scheme used in an analogue modulated intercepted signal is made.

Consider the case where the received signal is either an AM or FM modulated signal. An estimate of which scheme is being used can be made by measuring the amplitude variations of the demodulated signal. An initial modulation scheme selection is made and the variation of the demodulated signal amplitude values is used as an indication of the validity of the scheme estimate.

If $\{AM, FM\}$ denotes the set of possible analogue modulation schemes, automatic modulation scheme selection is possible by determining σ_{da}.

σ_{da} is the standard deviation of the instantaneous amplitude for the nonweak signal segments and is defined by:

$$\sigma_{da} = \sqrt{\frac{1}{N_s}\left[\sum_{A_n(i)>a_t}^{N_s} A_n^2(i)\right] - \left[\frac{1}{N_s}\sum_{A_n(i)>a_t}^{N_s} A_n(i)\right]^2}$$

(2)

where $A_n(i)$ is the instantaneous amplitude value at time instants $t = i/f_s$ and $i = 1,3,\ldots,N_s$ and f_s is the sampling frequency, a_t is the instantaneous amplitude threshold level [12] [13].

A σ_{da} value below a threshold value denoted by $t\sigma_{da}$ indicates an FM scheme. If σ_{da} is above $t\sigma_{da}$, AM is selected.

We now follow on to consider intercepted signals modulated using a digital scheme.

C. *Digital Modulation Scheme Outline*

For the second case, where the received signal is digitally modulated, the details of the exact scheme used are unknown. This section describes a moment matrix based classifier applied to a sparse matrix of normalised received signal points.

For the QPSK modulated signal case:

Let $y(t)$ be the received passband signal where $y(t) = a_n A\cos(\omega_0 t + \theta) + b_n A\sin(\omega_0 t + \theta) + n(t)$ where $\omega_0 = 2\pi/Ts$, Ts is the sampling rate 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 where,

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

Finally, $y(t) = y_1\varphi_1(t) + y_2\varphi_2(t)$ where $\varphi_1(t) = \sqrt{\frac{2}{T_S}}\cos(\omega_0 t)$ and $\varphi_2(t) = \sqrt{\frac{2}{T_S}}\sin(\omega_0 t)$

Now that we have defined an input test signal, the following section outlines how the quadrature components of the signal are extracted.

IV. SOFTWARE DEMODULATOR

An intercepted signal is digitised using a high rate analogue to digital converter (ADC). The digital representation of the analogue signal is then passed to the processing stage. The objective of this stage is to obtain a low pass representation of the signal and extract the quadrature components, which are used to create the signal space diagram.

A practical synchronous receiver system used for demodulating DSB-SC signals is the Costas receiver [14]. The received signal is supplied to two coherent detectors, which have individual local oscillator signals that are phase quadrature with respect to each other. The coupled in-phase and quadrature detector outputs form a negative feedback system used to attain and maintain local oscillator phase lock with the input received signal.

Let $v_I(t) = \cos(\omega_V t + \hat{\theta})$ and $v_Q(t) = -\sin(\omega_V t + \hat{\theta})$ be the outputs of the NCO (Numerically Controlled Oscillator).

When the input signal $y(t)$ is applied to the Costas Loop receiver, the mixer stage output is $c_I(t) + jc_Q(t)$ where $c_I(t)$ is equal to $y(t)v_I(t)$ and $c_Q(t)$ is equal to $y(t)v_Q(t)$.

Let the quadrature information obtained from the integrate and dump demodulator component be $c_{I_{id}}$ and $c_{Q_{id}}$ where

$$c_{I_{id}} = \sum_{i=0}^{Ts} c_I(i) \text{ and } c_{Q_{id}} = \sum_{i=0}^{Ts} c_Q(i).$$

Decision-directed carrier recovery is a common method used in coherent demodulators, and in this case, $\tanh(\cdot)$ is used as a slicer. For the QPSK case, a slicer operates on both of the quadrature component legs. The received signal point and the decided signal point are represented as vectors on the complex plane. The NCO error signal $\hat{\theta}_{err}$ is the cross product of these two vectors.

$$\hat{\theta}_{err} = \tanh(c_{I_{id}})c_{Q_{id}} - \tanh(c_{Q_{id}})c_{I_{id}}$$

Phase lock is achieved when $\hat{\theta}_{err} < t\hat{\theta}_{err}$ where $t\hat{\theta}_{err}$ denotes the phase error threshold below which the PLL is declared 'locked'. The following section outlines how the quadrature components of the signal are used to construct a signal space diagram.

V. 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 [15]. Once the actual modulation scheme details have been obtained, a method used to monitor channel quality and estimate channel capacity, is to calculate the received signal point variations about the expected constellation diagram.

5.1. *Normalisation and Matrix Construction*

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 $c_{I_{id}}[1,...N] - jc_{Q_{id}}[1,...N]$, is replaced by

$$\frac{K\left[c_{I_{id}}[1,...N] - jc_{Q_{id}}[1,...N]\right]}{Max\{Mag\{c_{I_{id}}[1,...N] - jc_{Q_{id}}[1,...N]\}\}},\qquad (3)$$ where

K denotes the scaling factor.

Since all of the expected received signal points are equally likely, only one side of the signal space diagram will be used for modulation scheme classification in order to reduce the processing time. The valid set of signal points is reduced to $T = \{\Re\{c_{I_{id}}[1,...N] - jc_{Q_{id}}[1,...N]\} \geq 0\}$ where $T = T[\hat{N}]$.

The one-sided signal space diagram is constructed as follows:

Let $B = B(x,y)$ be a rectangular matrix where $(x,y) \in A = \{1,...n\} \times \{1,...2n\}$ and $K \bmod n = 0$.

The matrix values are obtained from the valid set of signal points, with an example shown in Fig. 1, using the transform $(x,y) = \left\{round(\frac{\Re\{T\}}{K}), round(\frac{\mathrm{Im}\{T\}}{K})\right\}$ (4).

Equations (3) and (4) are the results of algorithm optimisation work carried out on the techniques described in this paper.

For the set of $\hat{N}$ valid signal points, if the index $(x_2, y_2) = (x_1, y_1)$, then $B(x,y) = B(x,y) + 1$.

B. *Moment Matrix Calculations*

The eight-order moment I_{oxyz} is the sum of the moments about the origin of received signal point clouds where I_{oxyz} is defined by $I_{oxyz} = I_{ox} + I_{oy} + I_{oz} = 2\sum m_i r_i^8$ (5) [16] [17].

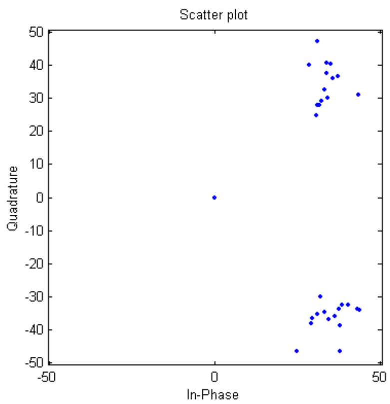

Fig. 1 Reduced Form Constellation Diagram for a QPSK modulated signal with AWGN (SNR 39 dB)

Let $t_{qpsk} I_{oxyz}$ denote the threshold above which, the modulation scheme estimation is QPSK and let $t_{bpsk} I_{oxyz}$ be the threshold below which, BPSK is the modulation scheme estimation. QPSK is chosen if $t_{qpsk} I_{oxyz} > t_{bpsk} I_{oxyz}$.

VI. IMPLEMENTATION

Automatic modulation scheme detection was implemented as part of a software radio system. Fig. 2 shows a block of a software radio receiver containing the recognition section. The software radio uses a layered approach, i.e. each function of the software radio forms a layer of the system. The software radio is a Win32 application, and runs on a 700MHz Pentium III processor.

The radio hardware is comprised of the antenna, RF amplifier, IF converter stage and bandpass filter. A 2MHz 12-bit ADC performs bandpass sampling of the amplified IF signal from the radio hardware. Bandpass sampling, where the received signal is not centred 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 [18]. 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 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.

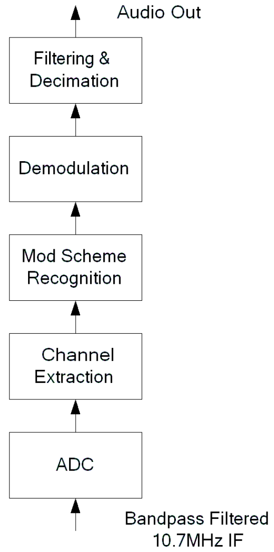

Fig. 2 Software receiver layer structure concept

For real-time receiver operation using a 100MHz 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. Code optimisation involved processing data 'in place' using optimised signal processing library functions and sin/cos lookup tables.

An AM/FM RF signal generator is used to generate a 2dBm audio modulated 10.7MHz test signal. This signal is connected to the ADC input via a coaxial cable. An audio output layer is placed on the top of the receiver stack, enabling demodulated data monitoring using the PC speakers.

VII. Results

We consider AM and FM cases only. The channel signal to noise ratio (SNR) is varied between 59dB and 2.9 dB, and σ_{da} is calculated, with a graph of σ_{da} versus SNR shown in Fig. 3. For the AM case, σ_{da} values are in the range (0.576,0.7098), σ_{da} values for the FM case are in the range (0.309, 0.6586). Based on these observations, $t\sigma_{da}$, the modulation scheme decision threshold estimate, is the mean of the AM and FM mean instantaneous amplitude standard deviation values. The mean of σ_{da} is determined over an observation period and low pass filtered. Low pass filtering is a basic means of minimising the effect of outliers and signal transients at the expense of introducing a modulation scheme switching delay to the system. For the example case shown in Fig. 3, for a channel with a SNR ≈ 50dB, a suitable $t\sigma_{da}$ value, may be taken as 0.5. As the channel SNR is decreased, differentiation between AM and FM modulation schemes fails at approximately 6.5 dB SNR.

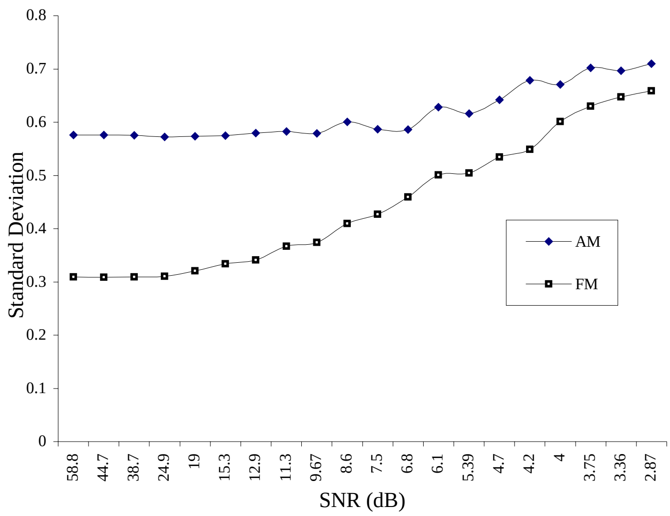

Fig.3. Graph of instantaneous amplitude standard deviation versus channel SNR

VIII. Conclusions

This paper described an automatic modulation-scheme recognisation technique for use in a software radio system. The technique is capable of determining the modulation scheme of a signal in the case when the modulation scheme class is not known *a priori*, and can be used for both digital and analogue modulation schemes. The modified moment based technique introduced here is a robust and lightweight scheme, and therefore requires a greatly reduced processing time over existing modulation detection schemes. The reduced processing time ensures that real-time operating objectives of the software radio system are met. Tests and simulations have shown that the SNR threshold for correct analogue modulation scheme recognisation is approximately 6.5dB. Further work in the area of facilitating dynamic threshold point for modulation scheme classification may help to increase the performance of the technique in the low SNR range.

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