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Adaptive Modulation with Moments based Signal-to-Noise Ratio Estimator

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ABSTRACT

Data-Aided Signal-to-Noise-Ratio (SNR) estimation is considered for time selective fading channels whose time variation is described by a polynomial time model. The inherent estimation accuracy limitations associated with the problem are quantified via a CramerRao Bound analysis. A maximum likelihood type class of estimators is proposed and its exact, non-asymptotic performance is computed. The standard, constant channel SNR estimator performance is determined in the presence of channel polynomial order mismatch. Simulations results are presented which verify the effectiveness of the technique as well as its performance advantage over previously proposed methods.

I. INTRODUCTION

Modern wireless communication systems often require knowledge of the receiver signal-to-noise-ratio (SNR). For example, SNR estimates are typically employed in power control, mobile assisted hand-off, adaptive modulation schemes, as well as soft decoding procedures, e.g., [1]-[2]. Variety SNR estimators have appeared in the literature, e.g. [3] and references therein. One well-known approach is the maximum likelihood (ML) estimator of [4]-[5]. All of these estimators are based on the assumption that the channel is constant throughout the observation period. In many application, e.g. mobile communication, this assumption is not true [6]. In this paper, we will deal with the problem of SNR estimation in time selective channels.
Consider a discrete time, complex baseband representation of a received communication signal, subject to flat fading:

yn = anhn + wn n = 1 · · · N, (1)

where at the time index n, yn is a received signal, an is a known transmitted signal, hn is a slowly time varying channel gain, and wn is a realization of a zero mean white Gaussian random process of variance σ 2 w. These samples can be conveniently represented in N × 1 column vector form

y = Ah + w ------- (2)

Where y ≡ [y1 · · · yN] T, h ≡ [h1 · · · hN] T, w ≡ [w1 · · · wN] T , and T denotes the transpose operator. The N × N matrix A is a diagonal matrix with Ann = an. There are different statistical models in the literature for the channel h. The most common is Clarke’s model, e.g. [6], in which the process hn is a realization of a zero mean complex normal random process, giving rise to Rayleigh amplitude fading with the following correlation function

$$\mathrm{E}[h_n h^*_{n-l}] = \sigma_h^2 J_0\left(2\pi \frac{f_d}{f_s} l\right), \quad \text{------ (3)}$$

Where E [·] denotes the expectation operator, [·] ∗ denotes the conjugate operator, fd is the maximum Doppler frequency, fs is the sampling rate, and J0 (·) is the zero’th order Bessel function. Most data aided applications choose N such that fd fs N 1, i.e. the channel is highly correlated and almost constant during the observation interval. The channel is modeled as hn = α0 where α0 is a constant channel gain. As will be seen in the sequel, even small channel variations from this model, can dramatically degrade the performance of traditional constant channel SNR estimators.

II. ADAPTIVE MODULATION WITH PERFECT CHANNEL STATE INFORMATION

The extensive literature of adaptive communication systems extend from as far back in the late 1960's [11]. After that, due to lack of technical depth in estimation methods and practical constraints in hardware advancements, the new found interest started to gain momentum approximately two decades thereafter. Goldsmith discussed some results on AMC with fading channels, utilizing trellis and turbo-coded MQAM modulation for flat-fading [12]. Goldsmith and Varaiya, showed that Shannon capacity of the communication link can be achieved by adapting the rate and power of transmitter in their work [13].

This was further developed to practical variable-rate variable power adaptive modulation communication is derived and analysed. The adaptive system's improvement in spectral efficiency is fulfilled while satisfying BER and power boundaries. It is also shown that the technique has up to 10 dB gain over constant rate variable power modulation whereas an even bigger 20 dB gain over non-adaptive modulation. Following the discoveries of adaptive modulation scheme's performance gain, a proposal to implement AMC in cellular wireless standards was given in [14], specifically in the CDMA2000 standard. Furthermore, AMC is also imparted in WCDMA high speed downlink packet access (HSDPA), IEEE 802.11x family standards and also WiMax standards for wireless internet access.

The system model and related representations are described henceforth. In this literature, complex signals are dealt with discrete sampling. Practical limitations for AMC system prior designing requires a feasible feedback path and a channel with slow varying conditions because the feedback must be able to keep up. This feedback frequency can be adjusted so that the link will always have updated information. However, with too frequent signaling between receiver and the transmitter can also mean inefficient use of bandwidth. Assume a discrete channel model, so that the sampled symbol at the receiver's matched filter output given by

$$r_k = \sqrt{g_k}\, x_k + n_k \,, \; k = 1, \ldots, N \qquad \text{---- (4)}$$

Where k x represents the transmitted complex symbols, k g is stationary and ergodic channel gain, and k n are independent and identically distributed Gaussian noise with . With samples r0, rσunknown variance 2 2 1 the–,rN Λ1, system is predicted to estimate the SNR value. Adaptive modulation requires CSI which can be derived at the receiver by evaluating pilot symbols embedded in the transmitted signal. These pilot symbols are known at both ends. While pilot symbols provide a straight forward way to conduct channel acquisition, they consume transmit power and bandwidth, which in turn reduces spectral efficiency of the overall system.

Hence, we propose a NDA estimator which is responsible to inform transmitter of the channel state. The symbols are drawn from a finite constellation which is known to the receiver and has I different amplitudes, having i th amplitude Ai and probability Pi (i = 1, ..., I). The constellation pth moment is denoted by

$$M_p = E\{|x_k|^p\} = \sum_{i=1}^{I} P_i A_i^p \qquad \text{---- (5)}$$

III. ADAPTIVE M-PSK MODULATION

In this proposed, there is a delay- and error-free feedback path to the transmitter. This feedback is responsible for returning CSI of SNR estimates. Denote the instantaneous SNR received at time k. The κγLet received signal is also assumed to have ideal coherent phase. Since gk is stationary, then it is also independent on time k,).γWe denote this distribution p (The receiver adapts the modulation constellation according to SNR received from the feedback. Given a finite set of constellations available, 1 –0 { } = N defines the γi i where the constellations are associated. One γrange of constellation is assigned to an instantaneous SNR region of , but when an SNR value drops– N ≤ i ≤1 + i γ i γ[,)(0 1) isγ, the communication link stops transmitting. 0 0γbelow called the cut-off SNR. This means that if the channel reaches an extensive degrades in quality, the channel should 6 dB is the cut-off value as it is–not is used. In this paper, decided by the performance if the SNR estimator. This {4, 8, 16}. M i.= 2, where ≤ i ≤paper considers only 0 In this paper, Spectral efficiency (SE) and average BER is evaluated. SE, denoted as R / B, equals the average data rate over unit bandwidth. At point in time, 2 M bits/symbol is sent across the channel= γ( ) log [ ] k where M-ary PSK is used, this also equals the spectral efficiency for a fixed M. In typical cases, spectral efficiency is influenced by BER and SNR. The Shannon-Hartley equation (15), describes just that and BER is also a factor because it relies on the theories of mutual information and entropy, which are not discussed in this general formula. C is the channel capacity in unit bps

$$\frac{C}{B} = \log_2\left(1 + \frac{S}{N}\right) \qquad \textbf{---- (6)}$$

The M 2M 4 moments NDA SNR estimator performs very well from 5 dB onwards for BPSK, 8-PSK, 16-PSK and 32-PSK. The estimator's performance is mainly acceptable.

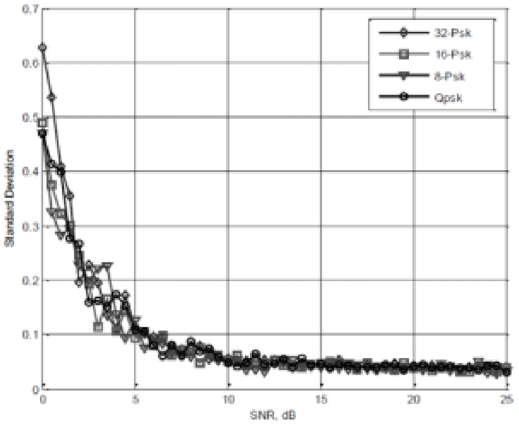

Fig.1 Standard deviation of SNR estimator

However, estimation values below 5 dB exhibit a minimal level of standard deviation from the exact value in Fig. 1. In Fig. 2, at –10 dB the estimators show an error of approximately 4 dB from the exact value. From that point onwards, its performance gradually improves with better SNR values. It is also noted that the M 2M 4 does not discriminate between all types of M-psk modulation. The performance is about the same for all where they perform poorly in low SNR conditions and then from –5 dB SNR onwards shows good accuracy. In real world implementation, the error of the estimator in low SNR will affect adaptive M-psk modulation which in reality the channel conditions are bad, but the estimator gives a better SNR feedback value to the transmitter. In the end, that can lead to a wrong decision hence contribute to a higher BER.

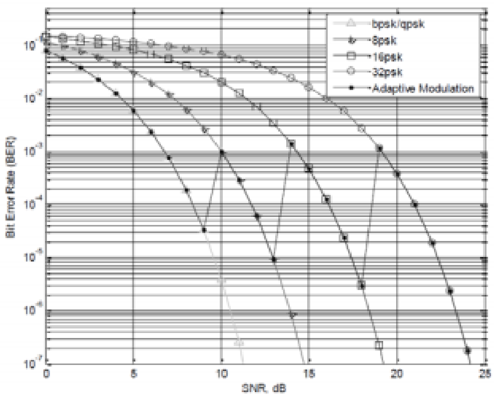

Fig.2 BER performance of adaptive M-PSK

The simulation on adaptive M-PSK modulation is conducted based on a quality of service (Quos) constraint, which is the BER. In the simulations, a BER of 10-3 is set to demonstrate how the link changes according to feedback.

Comparing the BER performance between the fixed M-PSK modulation methods with adaptive modulation in Fig. 4, we see that in low SNR values (6 – 9 dB), adaptive modulation has a better BER performance compared to the other higher order of modulations. For adaptive modulation, there is advantage in having low error rate at low SNR conditions and also higher bit rate performance at favourable channel conditions. Besides having the best performance compared to other types of estimators as shown in [10], the complexity of the M 2M 4 estimators is a factor to consider. Advantageously, simulation implementations for this estimator are not complex, therefore it computes rapidly even over long transmission duration of symbols.

IV. CONCLUSION

In this paper, second- and fourth-order moments based SNR estimator proves to be reliable, simple and accurate. Envelope-based SNR estimators are less complex yet produce excellent performance for M-psk constellations. This qualifies the estimator to complement the adaptive system described. We have shown that for a certain range of low SNR environment, the BER performance is much better in adaptive modulation compared to fixed modulation schemes. We also compared the spectral efficiency of said system with the theoretical bound. However, there are many constraints to take account before considering it as a practical implementation. This paper provides analytical approach towards adaptive modulation's capabilities in the context of measuring overall performance with valid assumptions. For future work, the adaptation parameters could include coding for better resilience against fading channels as forward error correction (FEC) coding has proven effective in noisy channels. Using different coding rates can introduce more types of transmission strategies, hence more suitable approach to different CSI. Furthermore, the balance between using FEC which consumes higher forward bandwidth against the spectral advantage of AMC is an interesting topic to venture.

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