Statistical Cross Layer Adaptation in Fast Fading Mobile Satellite Channels

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Abstract—Link adaptation in mobile satellite channels is difficult because of the large propagation delay, the frequent signal blockage and the variation of the channel statistics. In this paper, we propose a cross-layer link adaptation strategy that exploits statistical, long-term CSI to increase the throughput when some retransmissions are available; this increase is obtained while meeting a target outage probability constraint. Our strategy takes as inputs the estimated packet error rates of the available modulation and coding schemes (MCS) and their rates; as an output, it returns the optimum sequence of MCS to be used. Results will also show that simple information acquisition strategies can still provide very good results.

I. INTRODUCTION

In the last few years there has been an increasing interest in enabling high capacity mobile satellite communications at an affordable cost. This gain in spectral efficiency can be achieved by adopting some techniques being used in terrestrial communications. An aggressive frequency reuse, for example, dramatically improves the throughput performance by resorting to the use of smart multiuser detection techniques [1]. Link adaptation or adaptive coding and modulation (ACM) enables the possibility of dynamically adjusting the modulation and coding scheme (MCS) according to channel conditions so both reliability and throughput can be increased.

Link adaptation usually requires some sort of channel state information (CSI) at the transmitter (CSIT) to correctly select the transmission parameters. CSIT can be obtained in *open-loop* or *closed-loop*. In an open-loop scheme, CSIT can be obtained by exploiting channel reciprocity in time division duplex systems, and even channel statistical reciprocity in frequency division duplex. In the closed-loop case the receiver effectively measures the channel state, and feeds back some sort of channel quality indicator to the transmit side.

The nature of the land mobile satellite channel (LMS), however, makes the use of traditional link adaptation and CSIT acquisition procedures difficult to apply. First, the large propagation delay, inherent to the satellite scenario, makes it troublesome to obtain non-outdated CSIT in both open-loop and closed-loop, specially for the forward link [2]. Second, the

received power is expected to suffer frequent blockage of the line of sight (LOS) propagation path, thereby making the use of power margins very inefficient. Third, the available channel models (see, for instance, [3], and [4] and references therein) have quite involved analytical formulations, thus potentially disabling the possibility of deriving optimum strategies, as done with the Nakagami and Rayleigh channels, for example [5].

Even in this challenging scenario, quality of service (QoS) constraints coming from the data-link layer must be enforced, in particular packet error ratio (PER) and maximum delay. In order to avoid a drastic reduction of the throughput, retransmissions must be allowed, at least for some types of traffic. With this, the PER requirement can be relieved due to the diversity introduced by the repeated access to the channel. Despite the absence of reliable channel state information, we will prove that an adaptive automatic repeat request (ARQ) scheme can still benefit from the available statistical information of the channel.

The notion that physical layer (PHY) reliability and ARQ should be jointly considered to increase the throughput is clearly detailed in [6], and also in additional works such as [7], [8] for different scenarios. If rate-compatible codes are used [9], then Hybrid ARQ (H-ARQ) can perform better by combining the redundancy of different retransmissions. For simplicity, in this work we will assume basic MCS such that rateless strategies cannot be implemented. The uncertainty of the channel will be addressed in this work by using different MCS for successive retransmissions, so more aggressive (higher code rate) MCS can be used for the first transmissions. In a way this is an evolution of classical ARQ schemes for those cases when rate-compatible codes are not the preferred solution. Our approach is valid for any channel distribution as opposed to previous works such as [7], [8], specifically designed for Rayleigh and Nakagami block fading channel models, respectively, and with CSIT available. Moreover, CSIT reduces to the knowledge of the outage probability of the different MCS, which can be obtained just by observing the ACK/NAK information at the data link layer.

The remaining of the paper is structured as follows: Section II introduces some initial considerations concerning link adaptation; Section III presents the system model; in Section IV we present the proposed algorithm for link adaptation; Section V details the simulation results; finally, Section VI contains the conclusions.

II. ADAPTATION CONSIDERATIONS

Consider a satellite communication link with a physical layer with ACM and data link layer with ARQ. At the transmit side, and for each transmitted packet, the PHY layer selects an MCS, and the remote data link layer responds with an ACK if the packet is correctly received. If the receiver is not able to decode the packet, it transmits an NAK and the transmit data link layer triggers a retransmission with a potentially different MCS. In this paper we assume perfect decoding of NAK and ACK packets, as they are usually transmitted with a highly robust MCS.

The packets are divided into M different classes of traffic, $\mathcal{K}_1, \ldots, \mathcal{K}_M$, each one with a maximum delay constraint $d_1, \ldots, d_M$ and an outage probability constraint $p_1, \ldots, p_M$. In this scenario, the outage probability constraint means that the probability of a packet being delayed more than d_i (or not transmitted at all) has to be less than p_i, $i = 1, \ldots, M$. The delay includes the queue waiting time, the propagation delay, and the retransmission delay. These constraints are related to the reliability required by higher layers (TCP, for example), or to the intrinsic time-dependent nature of the data being transmitted (voice, for example). This problem seems to be intractable due to the coupling between scheduling and MCS selection.

In a satellite scenario, the main delay contributions are expected to be the propagation and retransmission delay due to the long communication distance, specially in scenarios where the queue is *not very busy*. If the waiting time can be considered to be negligible with respect to the propagation delay, then the scheduling and MCS selection tasks can be decoupled by the transformation of the delay constraint d_i into a number of transmissions t_i constraint, meaning that the probability of a packet not being correctly delivered in t_i transmissions ($t_i - 1$ retransmissions) is less than p_i. The maximum number of transmissions can be obtained by

$$t_i = \left\lfloor \frac{d_i - \delta}{D} \right\rfloor \tag{1}$$

with δ a (usually small) value taking into account the waiting time, and D the propagation delay.

III. SYSTEM MODEL

Consider a narrowband fading channel model, such that the received signal at the n-th time instant can be written as

$$y_n = h_n x_n + w_n \tag{2}$$

with h_n the complex channel value, x_n the transmitted sample, and w_n is a zero-mean circularly symmetric complex Gaussian radom variable with variance σ^2, i.e., $w_n \sim \mathcal{CN}\left(0, \sigma^2\right)$. We assume that each codeword $\mathbf{x}_k = \left[x_{kN}, x_{kN+1}, \ldots, x_{(k+1)N-1}\right]$ has a constant length N, such that it sees the channel states

$$\mathbf{h}_k \triangleq \left[h_{kN}, h_{kN+1}, \ldots, h_{(k+1)N-1}\right]. \tag{3}$$

Each codeword $\mathbf{x}_k$ is the result of applying forward error correction coding and constellation mapping to a stream of bits; we consider a finite set of available codes $\mathcal{C} = \{\mathcal{C}_1, \ldots, \mathcal{C}_K\}$,

MCS	r	γ_{th} (dB)	MCS	r	γ_{th} (dB)
QPSK 1/4	0.357	-1.5	QPSK 2/3	1.261	4
QPSK 1/3	0.616	-0.3	QPSK 3/4	1.390	4.9
QPSK 2/5	0.745	0.6	QPSK 4/5	1.476	5.6
QPSK 1/2	0.831	1.9	QPSK 5/6	1.562	6.1
QPSK 3/5	1.132	3.1	QPSK 8/9	1.691	7.1

each one characterized by a certain mutual information threshold $i_1, \ldots, i_K$ and spectral efficiency $r_1, \ldots, r_K$. The set of MCS considered in this paper is shown in Table I. For simplicity, we have restricted our analysis to the QPSK constellation, although the proposed methodology can be extended to other modulations.

A. Abstracting from the PHY

We assume that N is long enough so that a codeword is successfully decoded whenever its mutual information is above its MCS threshold i_k, and discarded otherwise. The average mutual information can be expressed as

$$\mathcal{I}\left(\mathbf{y}_k, \mathbf{x}_k\right) = \frac{1}{N} \sum_{i=kN}^{(k+1)N-1} \Theta\left(\frac{|h_i|^2}{\sigma^2}\right) \tag{4}$$

with $\Theta\left(\gamma\right)$ the mutual information over a gaussian channel with SNR γ and input restricted to a certain constellation [10] $\{X_i, \ldots, X_M\}$

$$\Theta\left(\gamma\right) = 1 - \frac{1}{M \log_2 M} \\ \times \sum_{m=1}^{M} \mathbb{E}_w\left[\log_2\left(\sum_{k=1}^{M} e^{-\frac{|X_m - X_k + w| - |w|^2}{1/\gamma}}\right)\right] \tag{5}$$

with $w \sim \mathcal{CN}\left(0, \frac{1}{\gamma}\right)$.

For convenience, and in the fashion of the physical layer abstraction literature [11], we will work with the *effective SNR* instead of the mutual information; for the k-th codeword it reads as

$$\gamma_{\text{eff},k} \triangleq \Theta^{-1}\left(\frac{1}{N} \sum_{i=kN}^{(k+1)N-1} \Theta\left(\frac{|h_i|^2}{\sigma^2}\right)\right) \tag{6}$$

that is the SNR of an additive white Gaussian noise channel with the same mutual information as the faded channel $\mathbf{h}_k$. Equivalently, we transform the mutual information thresholds into SNR thresholds by $\gamma_{\text{th},j} = \Theta^{-1}\left(i_j\right)$.

B. Channel model

Although the proposed adaptation technique is applicable to any channel distribution, we will assume a 3-state Fontan [3] channel model for the simulations, with a resulting pdf given by

$$f_{|h|}(x) = \sum_{i=1}^{3} p_i \frac{x}{b_0 \sqrt{2\pi d_{0,i}}} \\ \times \int_0^\infty \frac{1}{z} \exp\left(-\frac{(\log z - \mu_i)^2}{2d_{0,i}} - \frac{x^2 + z^2}{2b_{0,i}}\right) I_0\left(\frac{xz}{b_{0,i}}\right) \, \mathrm{d}z \tag{7}$$

which is a mixture of Loo pdf, with i an index that is spanning the 3 different states of the Markov chain, p_i the probability of the i-th state, and $d_{0,i}$, μ_i and $b_{0,i}$ the parameters of the Loo distribution in the i-th state. The pdf $f_{\mathbf{h}}(\mathbf{h})$ of the vector channel seen by a codeword depends on the transition between states, the mobile speed, the space correlation of shadowing and fading, and lacks of a closed form expression. We assume that the delay is long enough such that two channel samples separated by a round trip time (RTT) span are approximately independent, so

$$f_{\mathbf{h}}(\mathbf{h}_k | \mathbf{h}_{k-d}) \approx f_{\mathbf{h}}(\mathbf{h}_k) \tag{8}$$

with d the RTT expressed in codewords. This will represent the usual channel evolution unless the mobile terminal speed is very low; in such a case independence cannot be assumed , and the corresponding correlation could be exploited for adaptation purposes.

Under this assumption, only statistical CSIT can be available, so the probability of error of the k-th codeword when using code $\mathcal{C}_j$ with CSI delayed d codewords is

$$P_{j,k} = \Pr\left[\gamma_{\text{eff},k} < \gamma_{\text{th},j} \,|\, \mathbf{h}_{k-d}\right] = \Pr\left[\gamma_{\text{eff},k} < \gamma_{\text{th},j}\right]. \tag{9}$$

Since $\gamma_{\text{eff},k}$ is stationary, $P_{j,k}$ does not depend on the time instant k, and we can define

$$P_j = \Pr\left[\gamma_{\text{eff}} < \gamma_{\text{th},j}\right]. \tag{10}$$

Our objective is to find the optimum sequence of MCS to use in the t available transmissions, in such a way that we maximize the average throughput while meeting an outage constraint. To do so, we will use as inputs the codeword rates and their outage probabilities for the 3-state Fontan channel; although we will start by assuming perfect knowledge of the pdf of γ_{eff} and, as a consequence, of P_j in (10), we will also show that it is also possible to obtain good results by estimating them in a simple way.

IV. LINK ADAPTATION

A. Problem statement

The objective of link adaptation is to maximize the throughput guaranteeing the data-link layer QoS, expressed in terms of packet error rate and maximum delay. For a maximum delay or, equivalently, for a maximum number of transmissions t given by (1), our design variables are the MCS indices to use on each attempt $\{s_1, \ldots, s_t\} \in [1, \ldots, K]^t$. These values must satisfy a PER value given by p_0, so

$$\prod_{i=1}^{t} P_{s_i} \leq p. \tag{11}$$

Each bit waiting to be transmitted can be in one of t different states denoted by M_i, $i = 1, \ldots, t$, with i the number of maximum attempts before being discarded. In state M_i the data still has i transmissions available. Decoding will fail with a probability P_{s_i}, and data will move onto the next state, M_{i-1}, with $i-1$ transmissions left; on the other hand, if the transmission is successful –with probability $(1 - P_{s_i})$– the system moves back to the first state. The Markov chain
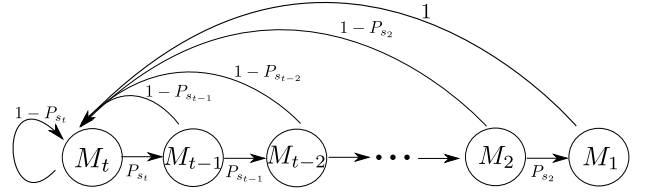

Fig. 1. Markov model representing the transitions between states.

representing this transmission procedure is depicted in Figure 1.

With this, we have that the average spectral efficiency (ASE) is given by

$$r = \sum_{i=1}^{t} \mathrm{P}\left[M_i\right] r_{s_i} \left(1 - P_{s_i}\right) \tag{12}$$

where the probability for a given bit to belong to state i is easily obtained from the diagram in Figure 1 as

$$\begin{aligned}
\mathrm{P}\left[M_t\right] &= 1/P \\
\mathrm{P}\left[M_{t-1}\right] &= P_{s_t}/P \\
\mathrm{P}\left[M_{t-2}\right] &= P_{s_t} \cdot P_{s_{t-1}}/P \\
&\vdots \\
\mathrm{P}\left[M_1\right] &= P_{s_t} \cdot P_{s_{t-1}} \cdot \ldots \cdot P_{s_2}/P.
\end{aligned} \tag{13}$$

For notational convenience, we have defined $P \doteq 1 + P_{s_t} + P_{s_t} \cdot P_{s_{t-1}} + \ldots + P_{s_t} \cdot P_{s_{t-1}} \cdot \ldots \cdot P_{s_2}$. Note that the outage probability of the last state P_{s_1} does not affect the probability of the different states, but only the average spectral efficiency of the last retransmission and the overall outage probability. The input bit rate is supposed to be such that these probabilities remain stationary with time.

Based on the previous considerations, the choice of the MCS indexes can be recast as the following optimization problem:

$$\begin{aligned}
\text{maximize} \quad & \sum_{i=1}^{t} \mathrm{P}\left[M_i\right] r_{s_i}(1 - P_{s_i}) \\
\text{subject to} \quad & \prod_{i=1}^{t} P_{s_i} \leq p_0.
\end{aligned} \tag{14}$$

This problem seems hard to solve, at least for the general case of different rates being assigned to subsequent retransmissions. We will now present some simplifications to this problem.

B. Simplifying the optimization problem

There are some optimality properties that can be exploited to avoid the need for an exhaustive search over the whole set of codes. Intuitively, for example, it seems plausible to expect higher rates at earlier attempts and more robust MCS for the subsequent retransmissions. Next we will prove this for $t = 2$, with the conjecture, supported by practical optimizations, that this is also the case for any t.

Proposition 1: If $t=2$, then the solution to (14) meets the following two propositions:

I) if $\exists j \neq s_i$ such that $r_{s_i}(1 - P_{s_i}) < r_j(1 - P_j)$ then $P_{s_i} \leq P_j \;\forall i=1, 2$

II) $r_{s_2} \geq r_{s_1}$ (or, equivalently, $P_{s_2} \geq P_{s_1}$).

Proof: The first proposition is easy to prove for $i = 1$. Let us assume (s_2^*, s_1^*) is optimum, and denote $T_x \triangleq r_x (1 - P_x)$, so we can write the objective function as

$$r = \frac{T_{s_2^*} + P_{s_2^*} T_{s_1^*}}{1 + P_{s_2^*}}. \tag{15}$$

It is easy to see that if there exists j such that $T_j > T_{s_1^*}$ and $P_j < P_{s_1^*}$ the outage constraint will be met if it was met for (s_2^*, s_1^*) and the objective function will increase, so (s_2^*, s_1^*) cannot be optimum.

Next, assume that (s_2^*, s_1^*) is optimum with $P_{s_1^*} > P_{s_2^*}$. This implies that $T_{s_1^*} > T_{s_2^*}$ from the previous property. Also, we have that the outage probability of (s_2^*, s_1^*) is the same as that of (s_1^*, s_2^*). Therefore, to prove that (s_2^*, s_1^*) is not optimum it suffices to prove that

$$T_{s_1^*}\theta_{s_1^*} + T_{s_2^*}(1 - \theta_{s_1^*}) \geq T_{s_2^*}\theta_{s_2^*} + T_{s_1^*}(1 - \theta_{s_2^*}) \tag{16}$$

where $\theta_{s_2^*} \triangleq 1/(1 + P_{s_2^*})$ is the probability of a bit being transmitted in the first state if the order is (s_2^*, s_1^*), and $\theta_{s_1^*} \triangleq 1/(1 + P_{s_1^*})$ is the probability of a bit being transmitted in the first state if the order is (s_1^*, s_2^*). Note also that $\theta_i \geq (1 - \theta_i)$, and $\theta_{s_2^*} \geq \theta_{s_1^*}$. With this, we have that

$$
\begin{aligned}
T_{s_1^*}\theta_{s_1^*} + T_{s_2^*}(1 - \theta_{s_1^*}) &= \theta_{s_1^*}\left(T_{s_1^*} - T_{s_2^*}\right) + T_{s_2^*} \\
&\overset{(i)}{\geq} \left(1 - \theta_{s_1^*}\right)\left(T_{s_1^*} - T_{s_2^*}\right) + T_{s_2^*} \\
&= T_{s_1^*} + \theta_{s_1^*}\left(T_{s_2^*} - T_{s_1^*}\right) \\
&\overset{(ii)}{\geq} T_{s_1^*} + \theta_{s_2^*}\left(T_{s_2^*} - T_{s_1^*}\right) \\
&= T_{s_1^*}\left(1 - \theta_{s_2^*}\right) + T_{s_2^*}\theta_{s_2^*}
\end{aligned}
\tag{17}
$$

where (i) is due to $T_{s_1^*} > T_{s_2^*}$ and $\theta_{s_1^*} > (1 - \theta_{s_1^*})$, and (ii) is due to $T_{s_1^*} > T_{s_2^*}$ and $\theta_{s_1^*} > \theta_{s_2^*}$. With this, we prove that it is better to transmit first with the higher rate code, so (s_2^*, s_1^*) is not optimum if $P_{s_1^*} > P_{s_2^*}$. Now, we have to prove proposition I for $i = 2$. This is easy to prove since

$$\frac{\mathrm{d}r}{\mathrm{d}P_{s_2}} = \frac{T_{s_1} - T_{s_2}}{(1 + P_{s_2})} \leq 0 \tag{18}$$

as $T_{s_1} \leq T_{s_2}$, and

$$\frac{\mathrm{d}r}{\mathrm{d}T_{s_2}} = \frac{1}{(1 + P_{s_2})} \geq 0 \tag{19}$$

so decreasing the error probability of s_2 while increasing T_{s_2} will increase the objective function.

This lemma proves that for $t = 2$ we only have to take into account those MCS where an increment in the outage probability implies an increment in the throughput, and that the MCS are of non-increasing rate. Note that this result is quite general, since the only assumption for the relationship between code rate and error probability is that higher rates imply higher error probabilities. ∎

Conjecture 2: For all t, the solution to (14) meets the following two propositions:

I) if $\exists j \neq s_i$ such that $r_{s_i}(1 - P_{s_i}) < r_j(1 - P_j)$ then $P_{s_i} \leq P_j \ \forall i = 1, \dots, t$

II) $r_{s_{i+1}} \geq r_{s_i}$ (or, equivalently, $P_{s_{i+1}} \geq P_{s_i}$) $\forall i = 1, \dots, t - 1$.

C. Optimization algorithm

With the above conjecture, the optimization algorithm could be written as follows: in a first step, we will obtain the subset of K' MCS that meet Condition 1; then, we will perform the optimization over them using Condition 2.

Require: MCS set sorted by descending rate, $i < j \Leftrightarrow r_j \leq r_i \ \forall r_i, r_j \in \{r_i\}_{i=1}^{M}$

$T_0 \leftarrow 0$
for $i = 1$ to K **do**
  if $r_i(1 - P_i) < T_{i-1}$ **then**
    Delete $\mathcal{C}_i$
  else
    $T_i = r_i(1 - P_i)$
  end if
end for

We now compute all the valid combinations of MCS:

$i \leftarrow 0$
for $a_1 = 1$ to K' **do**
  for $a_2 = 1$ to a_1 **do**
    $\vdots$
    for $a_t = 1$ to a_{t-1} **do**
      $F_i = \{\mathcal{C}_{a_1}, \mathcal{C}_{a_2}, \dots, \mathcal{C}_{a_t}\}$
      if $\prod_{j=1}^{t} P_{a_j} \leq p_0$ **then**
        $G_i = \sum_{j=1}^{t} \mathrm{P}\left[M_{a_j}\right] r_{a_j}(1 - P_{a_j})$
      else
        $G_i = \infty$
      end if
    end for
  end for
end for
return $F_{\mathrm{opt}} = \arg\max_{F_i} G_i$

The complexity of this algorithm is less than that of a brute-force search over all the possible combinations, which would result into K^t combinations; the complexity is now given by the following proposition.

Lemma 1: The modified algorithm only needs to check $\binom{K'-1+t}{t}$ combinations.

See Appendix A for the proof.

The reduction in complexity with respect to the brute force solution is quite remarkable, reaching almost an order of magnitude for $t = 3$ and $K' = 20$.

D. Knowledge of P_i

So far we have been implicitly assuming that the transmitter will know the pdf of γ_{eff} in order to obtain P_k perfectly. But, indeed, the proposed link adaptation algorithm can be easily implemented without this knowledge, in an online fashion. A similar idea was introduced in [12] for link adaptation in MIMO-OFDM systems. Here, the transmit side would estimate directly P_k by just observing the ACK/NAK of the ARQ protocol:

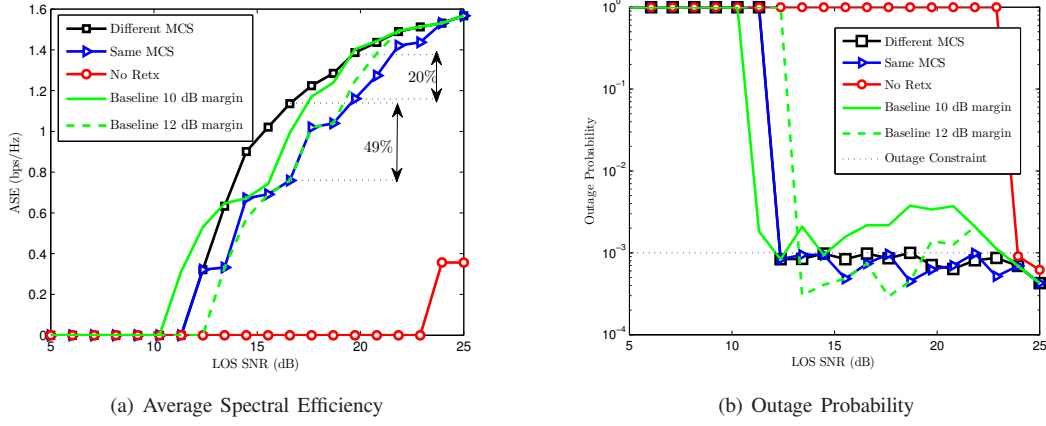$$\hat{P}_{k,n_{k+1}} = \frac{n_k}{n_k + 1}\hat{P}_{k,n_k} + \frac{1}{n_k + 1}a_k \tag{20}$$

(a) Average Spectral Efficiency

(b) Outage Probability

Fig. 2. Average spectral efficiency and outage probability. Receiver speed: 1 m/s

(a) Average Spectral Efficiency
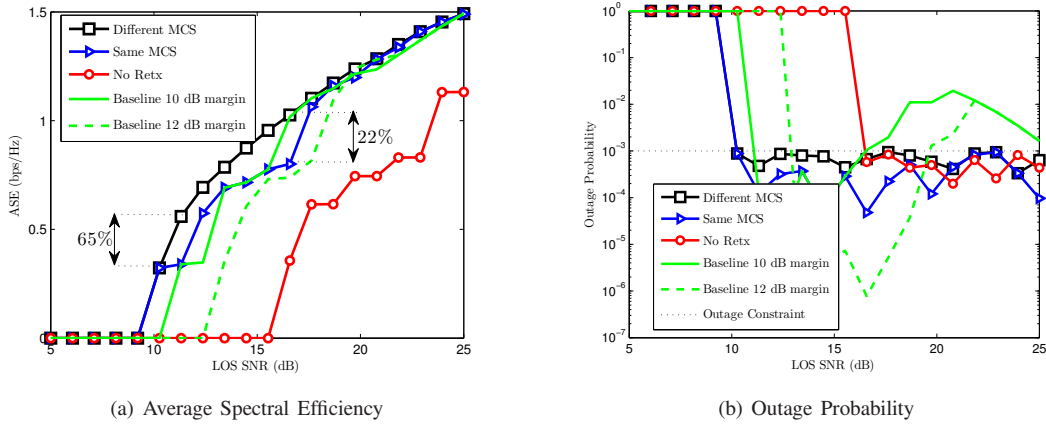

(b) Outage Probability

Fig. 3. Average spectral efficiency and outage probability. Receiver speed: 5 m/s

with n_k the number of ACK/NAK obsevations for MCS c_k, $a_k = 1(0)$ if an ACK (NAK) is received, and $\hat{P}_{k,n_k}$ the estimation of P_k after n_k packets are received.

A transmitter selecting the MCS following the adaptation algorithm and updating the PER estimations following (20) can be easily stuck in a suboptimal solution. If some of the optimum MCS have not been *explored*, the corresponding PER estimation will have a large variance and it can cause the adaptation algorithm not to select that MCS, thus disabling the possibility of improving the estimation. To overcome this issue, we set a fraction ϵ of the packets to be scheduled with a random MCS, so we can improve the estimation of the PER in case an MCS is seldom selected. In our simulations we selected $\epsilon = 0.01$.

V. SIMULATION RESULTS

We have simulated the performance of the proposed ARQ scheme; we simulated a 3-transmission strategy (2 retransmissions) with an outage probability of 10^{-3}. The chosen central frequency is $f_c = 1550 \, \text{MHz}$ with a symbol period $T_{\text{symb}} = 1/(33.6) \, \text{ms}$ and a codeword length of 2688 symbols. The MCS for optimization are shown in Table I. In this first approximation, we assume perfect empirical knowledge of the probability density function of γ_{eff}. We compare the results

with the case of not allowing retransmissions at all ($t = 1$) with the same outage probability constraint, that could be the constraint in interactive applications, for example.

Results are shown on Figures 2-3 for speeds of 1 m/s and 5 m/s. The results are compared with the case of using the same MCS for every retransmission (i.e., an outage probability of 0.1 for each transmission), and with a baseline that consists on selecting the MCS by applying a margin to the measured average SNR value. The baseline method is a simplified version of link adaptation algorithms currently used in satellite communications, and the margin tries to capture both the mismatch between effective SNR and average SNR, as well as the CSI error induced by long propagation delays. It can be seen that our approach outperforms the fixed MCS in all scenarios, specially in the low speed ones due to the lower time diversity. Note that in some cases the fixed MCS is the solution to our problem, leading to the same throughput and outage probability. In the case of not allowing retransmissions the throughput is severely reduced due to the smaller diversity. In fact, for the low speed case, the small diversity introduced by the channel variations causes the system to be unable to meet the outage constraint for almost every SNR. The baseline method can be seen to have an irregular behavior, not being able to meet the outage constraint in many cases. In the low

TABLE II
OPTIMUM MCS EVOLUTION.

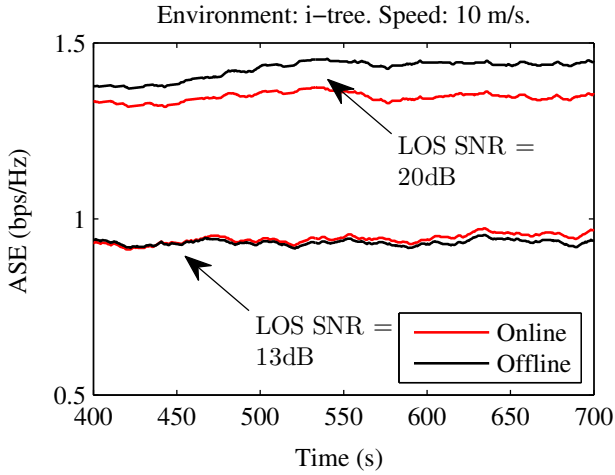| LOS SNR (dB)	$\mathcal{C}_{s_3}$	$\mathcal{C}_{s_2}$	$\mathcal{C}_{s_1}$	$\mathcal{C}_{\text{same MCS}}$
23.9474	QPSK 8/9	QPSK 8/9	QPSK 8/9	QPSK 8/9
21.8421	QPSK 8/9	QPSK 8/9	QPSK 3/4	QPSK 5/6
19.7368	QPSK 8/9	QPSK 8/9	QPSK 2/5	QPSK 2/3
18.6842	QPSK 8/9	QPSK 5/6	QPSK 1/4	QPSK 2/3
17.6316	QPSK 8/9	QPSK 5/6	QPSK 1/4	QPSK 3/5
16.5789	QPSK 5/6	QPSK 2/3	QPSK 1/4	QPSK 1/2
15.5263	QPSK 3/4	QPSK 3/5	QPSK 1/4	QPSK 2/5
13.4211	QPSK 3/5	QPSK 1/4	QPSK 1/4	QPSK 1/3

Fig. 4. Evolution of the average spectral efficiency with online adaptation, and comparison with the offline approach.

speed scenario the baseline algorithm outperforms in terms of throughput the proposed statistical approach, but this comes at the cost of not meeting the outage constraint.

The solution for 1 m/s for the different points is shown on Table II, where it can be seen that the MCS rate decreases with the transmission index, and that there can be a huge variation between the first and last transmissions. In the simulations we solved the problem by brute force, so these results agree with Conjecture 2.

In Figure 4 we show the evolution of the spectral efficiency with time (averaged over a window of 700 packets) when online adaptation is used following (20). We can see that in the high SNR case the offline optimization outperforms the online approach due to the perfect knowledge of the error probabilities P_k. For the lower SNR case both approaches offer approximately the same performance. For the sake of simplicity, we used *Stop-and-Wait* ARQ for the simulations, so only one packet is transmitted every RTT, although the method can be also applied for more efficient protocols like *Selective Repeat ARQ* [13].

VI. CONCLUSIONS

In this paper we presented a cross-layer approach to link adaptation in mobile satellite links. Only statistical information about the channel is assumed, and ARQ is exploited to obtain time diversity. Unlike previous approaches, and even when the

same statistical CSI is observed, different MCS are used for the subsequent retransmissions. We conclude that the use of different MCS leads to a throughput increment, and that simple ACK/NAK information can be exploited to obtain statistical CSI. Future work will address environment (statistical CSI) changes, convergence of the online adaptation algorithm, and channel correlation modeling.

APPENDIX A
PROOF OF LEMMA 1

Proof: From the algorithm description, we have that the number of combinations will be given by

$$
\begin{aligned}
N &= \sum_{a_1=0}^{K'-1} \sum_{a_2=0}^{a_1} \cdots \sum_{a_{t-2}=0}^{a_{t-3}} \sum_{a_{t-1}=0}^{a_{t-2}} \sum_{a_t=0}^{a_{t-1}} 1 \\
&= \sum_{a_1=0}^{K'-1} \sum_{a_2=0}^{a_1} \cdots \sum_{a_{t-2}=0}^{a_{t-3}} \binom{a_{t-2}+2}{2} \\
&= \sum_{a_1=0}^{K'-1} \binom{a_1+(t-1)}{t-1} = \binom{K'-1+t}{t}
\end{aligned}
\tag{21}
$$

where we have used $\sum_{j=0}^{k} \binom{n+j}{n} = \binom{k+n+1}{n+1}$. ∎

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