LINK ADAPTATION AND CHANNEL PREDICTION IN WIRELESS OFDM SYSTEMS

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ABSTRACT

Link adaptation algorithms enhance throughput in OFDM communication systems by adjusting various transmission parameters based on channel state information. In wireless OFDM systems, however, changes in the channel due to Doppler spread and limits on the reverse link bandwidth make it difficult to fully adapt to current channel conditions. Therefore in this paper we study link adaptation based on time-frequency blocks. We propose an algorithm that adapts the modulation scheme in a time-frequency block to maximize throughput while maintaining a target average block error rate. We describe a channel prediction algorithm and comment on the channel conditions where it improves performance. Simulations show that it is possible to optimize the size of the time-frequency block to maximize throughput given a fixed frequency of link adaptation control messages.

1. INTRODUCTION

Orthogonal frequency division multiplexing (OFDM) is an attractive modulation technique for high-bandwidth wireless applications since it dramatically simplifies equalization of intersymbol interference channels. Using link adaptation (LA), it is possible to improve throughput in wireless OFDM systems by adjusting the transmission parameters such as the modulation scheme, power allocation and/or code rate to the current channel state information [1]. Ideally, link adaptation algorithms would adapt at every time instant in frequency to the instantaneous channel realizations. Unfortunately, limitations in the feedback bandwidth (critical in frequency division duplex (FDD) systems) and variation of the channel due to Doppler spread makes ideal adaptation difficult in practice.

In this paper we consider the problem of adaptively selecting the optimal modulation scheme (or transmission mode) for wireless systems that use OFDM modulation (see [1] and the references therein). To dramatically reduce the amount and frequency of required mode change messages (MCM) we propose an algorithm for LA that adjust the modulation scheme for time-frequency blocks of tones. Other papers have proposed to use subband aggregation to take advantage of the coherence bandwidth in frequency [1] [2] however these algorithms still require a high frequency of MCM to account for temporal variations. Prediction [3] or robust allocation [4] can be used to improve tolerance to delay and to reduce the frequency of MCM but they are only useful when the averaging window is less than the coherence time of the channel. We derive expressions for the Average Subband Error Rate (ASER) as a function of both the subband width and temporal window length. Using these expressions, we derive algorithms for maximizing the throughput based on a target OFDM subband error rate threshold. Finally, we demonstrate how channel prediction improves performance when the channel is slowly varying.

A general discussion of time, space, and frequency adaptation was presented in [5] where it is proposed to collect channel statistics over time and frequency to map the SNR to the BER information. Compared with [5], we do not consider spatial adaptation but we provide exact expressions for the ASER and thus do not need to construct extensive look-up tables. Compared with [2][3], we do not have restrictions on the size of the subband or the length of the averaging window. Naturally, performance improves when more rapid feedback is available. Further, unlike prior work [5][2][3], we show how it is possible to optimimize the dimensions of the time-frequency block based on a target average number of MCM per second.

This paper is organized as follows. In Section 2 we presents a brief overview of the system and comment on the simulation parameters. In Section 3 we derive the OFDM symbol error rate, and a low-complexity approximation, as a function of the per-tone symbol error rate. Preliminary results are summarized. In Section 4 we show how the performance of the proposed algorithm can in some cases be improved through linear prediction of the future channel coefficients.

2. SYSTEM DESCRIPTION

Consider an OFDM communication system that employs adaptive modulation. Let there be N total tones and a cyclic prefix of length L_p. Let $h(l, t)$ $l = 0, 1, \ldots, L_s$, $L_s \leq L_p$, denote the block time-varying impulse response where l is the lag and t is the OFDM symbol index in time. We assume that the channel is constant for the duration of each given OFDM symbol but can vary symbol to symbol. Equivalently this implies that T_s, the OFDM symbol period, is much less than the channel coherence time T_c. This allows us to ignore the effect of inter carrier interference. Let $s(n, t)$ be the symbol transmitted on tone n during block t. Assuming perfect synchronization and sampling, the equivalent input-output relationship for an OFDM system can be written as

$$y(n,t) = \sqrt{E_s} H(n,t) s(n,t) + v(n,t) \qquad (1)$$

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where $H(n,t) = (1/\sqrt{N}) \sum_{l=0}^{L_s} h(l,t) \exp(-j2\pi nl/N)$, $v(n,t)$ is complex additive white gaussian noise with distribution $\mathcal{N}(0, N_o/2)$ per dimension, and $y(n,t)$ is the received signal. For purposes of simulation we generate $h(l,t)$ using a uniform power delay profile with $L_s \le L_p$ sample-spaced independent complex Gaussian multi-paths with power spectrum satisfying the Clarke spectrum. We refer to [1] for details on the system description.

To reduce the amount of feedback required, we divide the channel into time-frequency blocks. To index the temporal window, let $\mathcal{T}_m$ denote the set of integers that indicate the OFDM symbols in the m^{th} window. The length of the temporal window is $T_w = |\mathcal{T}_m| T_s$ where T_s is the duration of the OFDM symbol. We say that a window is short if $T_w < T_c$ otherwise we say the window is long. To index the tones, let $\mathcal{N}$ denote the set of integers, with elements in $[0, N-1)$, that indicates the usable subcarriers in the given OFDM symbol. For purposes of adaptive modulation we assume that $\mathcal{N}$ is divisible into S subbands whose index sets are denoted $\mathcal{S}_1, \mathcal{S}_2, \ldots, \mathcal{S}_S$. For simplicity we assume that the width of the k^{th} subband is $|\mathcal{S}_k| = |\mathcal{N}|/S$. We say that a subband is narrow if $|\mathcal{S}_k|$ is less than the coherence bandwidth of the channel.

In this paper we are interested in finding an algorithm to determine the modulation mode in each time-frequency block. To minimize the amount of feedback we do not optimize the transmitted power distribution although we do allow for *mode 0* for which no bits are transmitted. We also assume the system is uncoded, that is, we do not take forward error correction coding or time-frequency interleaving. We are currently working on the best way to incorporate these issues.

Given these assumptions, with ideal channel knowledge, the mode selection problem is to determine the modulation scheme in the m^{th} window $\{s(n,t)\}_{n \in \mathcal{S}_k, t \in \mathcal{T}_m}$ given $\{H(n,t)\}_{n \in \mathcal{S}_k, t \in \mathcal{T}_m}$ for each subband k. If ideal channel knowledge is not available then we will predict the modulation mode for the m^{th} window given $\{H(n,t)\}_{n \in \mathcal{N}, t \in \mathcal{T}_k}$ for $k = k_0, \ldots, k_1$. We can incorporate delay in the feedback by allowing $k_1 < k$. Allowing $k_1 - k_0 + 1$ previous windows improves channel prediction at the expense of an increase in complexity. Note that due to the fact that fading channels are often oversampled [6], we may not use all the previous channel observations in the predictor.

To make the discussion concrete, we will present our results in the context of the IEEE 802.11a system which uses $N = 64$ tones (with 48 active tones) over $20MHz$ bandwidth and carrier frequencies in the UNII band around $5GHz$. In this system the OFDM symbol duration is $3.2\mu s$ while the cyclic prefix duration is $800ns$ thus $T_s = 4\mu s$. The terminal mobility which can be supported by the standard IEEE 802.11a is up to $10m/s$. Therefore, we assume that the Doppler is $170Hz$ which corresponds to a coherence time $T_c = 5.8ms$. We will choose from the following constellations BPSK, 4-QAM, 16-QAM, and 64-QAM modulations in our link adaptation algorithm.

3. TIME-FREQUENCY ADAPTIVE MODULATION IN OFDM SYSTEMS

In this section we will now describe a general approach for adaptive modulation in wireless OFDM systems. The goal of our proposed adaptive modulation techniques is to maximize the throughput while maintaining a minimum ASER.

3.1. OFDM Symbol Error Rate Computation

Let $P_s(\gamma, H, M_s, t)$ denote the error rate of the s^{th} subband as a function of the SNR γ, the channel H, the size of the constellation M_s (2, 4, 16, 64 in our case), and the block t. Recognizing that one or more per-tone symbol errors cause a subband error, the subband error rate is written

$$P_s(\gamma, H, M_s, t) = 1 - \prod_{n \in \mathcal{S}_s} \left(1 - P(\gamma |H(n,t)|^2, M_s)\right) \quad (2)$$

$$\approx \sum_{n \in \mathcal{S}_s} P(\gamma |H(n,t)|^2, M_s) \quad (3)$$

where $P(\gamma |H(n,t)|^2, M_s)$ is the per-tone symbol error probability. The approximation in (3) results from expanding the product in (2) and throwing out the higher-order terms which are typically much smaller. For M-QAM we can use [7]

$$P(\gamma, M) \le 4\left(1 - \frac{1}{\sqrt{M}}\right) Q\left(\sqrt{\frac{3\gamma}{M-1}}\right) \quad (4)$$

while for BPSK we can use the exact expression $P(\gamma, 2) = Q(\sqrt{2\gamma})$.

Since we propose adaptation of time-frequency blocks in this paper, we propose to use the subband error rate averaged over the temporal window. Thus we define the ASER in the m^{th} window as

$$\bar{P}_s(\gamma, H, M_s, m) = \frac{1}{|\mathcal{T}_m|} \sum_{t \in \mathcal{T}_m} P_s(\gamma, H, M_s, t) \quad (5)$$

$$\approx \frac{1}{|\mathcal{T}_m|} \sum_{t \in \mathcal{T}_m} \sum_{s=1}^{S} P(\gamma |H(n,t)|^2, M_s) \quad (6)$$

When (3) is used to simplify the calculation of (6), we refer to this as the *approximated* ASER. Otherwise when (2) is used to compute (5) we refer to this as the *exact* ASER. Clearly the approximate approach reduces the complexity of the calculation since it requires much fewer multiplies.

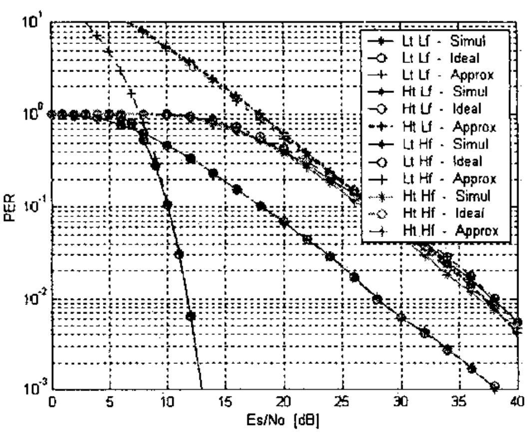

Figure 1: *Near-exact*, *approximated* and *simulated* ASER for 4 QAM modulation, for different channel conditions (H_t/L_t: High/Low time-selectivity; H_f/L_f: High/Low frequency-selectivity)

To illustrate the impact of choosing either the exact or the approximate method for calculation, Fig. 1 shows the evolution of the ASER for different channel conditions for the case of 4-QAM modulation. It is evident that for most of the channel conditions the approximated ASER curves match the simulated ones, except when the channel is flat in frequency (L_f) and highly time-selective (H_t). In fact, in this case, when the SNR drops down (deep-fade) all the tones will experience at the same time high per-tone symbol error probability. Therefore, the first order terms in (3) are not small enough to describe the probability of error of the s^{th} subband given by (2). Consequently, to use the approximation of the ASER for highly time-selective channels, we have to decrease the size of the averaging window so that the channel is not highly time-selective in the observation window. Similarly for higher-order modulation schemes, we find that for threshold values of $P_{thresh} < 0.1$ the approximate ASER can be used instead of the exact one as long as the size of the averaging window is small enough (i.e., $T_w/T_c < 0.5$).

3.2. Adaptation Algorithm

Given the results in (2)-(6) we are now ready to formulate a rule for choosing the optimum modulation scheme that maintains an ASER target threshold P_{thresh}. In this section we assume that we have perfect knowledge of the channel in the time-frequency block of interest. We deal with the problem of obtaining and predicting this channel in the next section. Following the approximation in (6), we will choose the largest modulation scheme such that average subband error probability is less than P_{thresh} for each subband.

> **Windowed modulation selection.** Given a window of channel observations $\{H(n,t)\}_{n\in\mathcal{N},t\in\mathcal{T}}$ and γ, for each $s = 1, 2, \ldots, S$, choose the largest M_s such that $\bar{P}_s(\gamma, H, M_s, m) \leq P_{thresh}$.

The link adaptation algorithm uses windowed modulation selection to generate a MCM for every time-frequency block. The control messages for every subband in a given window are then conveyed back to the transmitter. In a time-division duplex (TDD) system, due to the reciprocity of the channel, the transmitter simply uses the modulation scheme predicted by the receive channel. In an FDD system, on the other hand, each MCM must be conveyed back to the transmitter through a reverse link control channel. Limitations on the speed at which the modulation can be adapted in TDD systems, and restrictions on the bandwidth of the reverse link control channel in FDD systems, motivate the desire the minimize the frequency of MCMs.

When the frequency of MCMs is fixed, or equivalently the average number of MCMs per second, a decision needs to be made about whether to allocate more subbands and longer averaging windows or fewer subbands and shorter averaging windows. In the former case we adapt to the average behavior of the channel in frequency while in the later case we adapt to the average behavior of the channel in time. Clearly this choice depends on the coherence time, the coherence bandwidth, the MCM frequency, and other relevant system parameters. To illustrate the tradeoff in adaptive modulation with a fixed number of MCMs, consider the simulation results illustrated in Fig. 2 and Fig. 3. In both simulations we computed the spectral efficiency for each window as $B_w T_w^{-1} \sum_s R_s(1 - \bar{P}_s)$ where R_s is the total number of bits transmitted on subband s, B_w is the bandwidth, and $\bar{P}_s$ is obtained from (5) or (6) in that window. In Fig. 2 we illustrate the spectral efficiency for the adaptive windowed algorithm for a fixed frequency of mode change messages $MCM_{av} := S/T_w$. We set $MCM_{av} = 1/20$ (one control message every twenty OFDM symbols) and considered low frequency selectivity of the channel, with the number of channel taps equal to 2. It is possible to see that a good tradeoff is 8 subbands and averaging window of the order of $T_w/T_c = 0.08$.

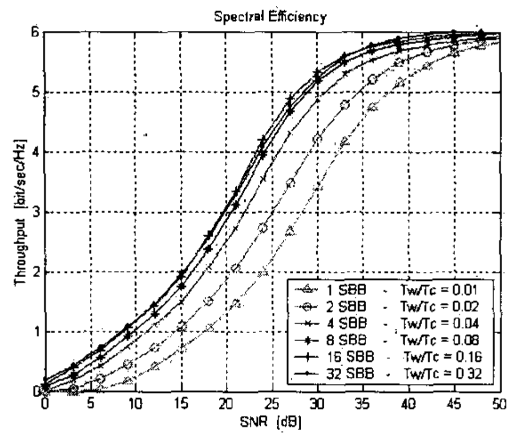

Figure 2: Spectral efficiency: tradeoff between averaging window (T_w) and size of each subband (B_w); fixed number of MCM_{av}

In Fig. 3 we illustrate the throughput assuming $S = 4$ for different choices of T_w. As expected, shorter averaging windows ($T_w/T_c < 0.5$) improve throughput since we are able to more closely track the channel. A side benefit is that, with this choice of window length, we can use the approximate ASER expression. With the same Doppler as before, the corresponding MCM_{av} will be 10^{-2}, 10^{-3} and 10^{-4} for the different choices of T_w. These numbers underline how loading algorithms based on time averaging window are able to smoothly tradeoff system performance with the frequency of signalling messages, as compared with instantaneous LA.

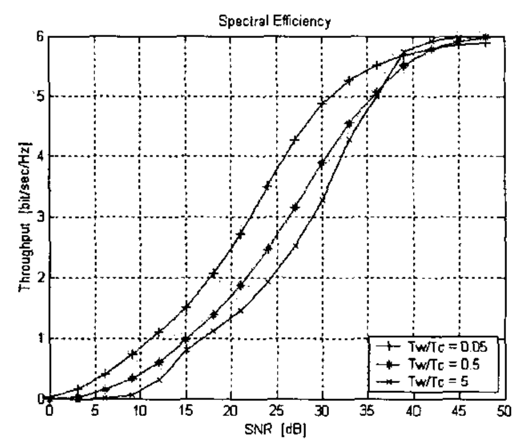

Figure 3: Spectral efficiency: comparison of the Throughput for different averaging windows (T_w/T_c) and number of MCM_{av}

4. CHANNEL PREDICTION IN OFDM SYSTEMS

The windowed adaptation algorithm described in the previous section assumes that the channel is known perfectly in the window of interest. Of course, due to delay in the feedback path, this assumption is unrealistic. It has been observed that channel prediction can solve this problem for short windows [3] therefore in this section we briefly summarize some the improvements due to prediction.

Due to space limitations, we provide only a brief description of our adaptive algorithm. Essentially, following the work by [6], we downsample the channel data to improve the predictor performance. Then, for each tone we estimate the coefficients of a linear predictor using the Burg algorithm, which is stable even when the stationarity condition of the channel is not guaranteed inside the observation window [8]. To estimate the future channel coefficients we apply one-step prediction reusing the past channel estimates. The prediction order of the filter is chosen based on the Minimum Descriptor Length (MDL) cost function.

To illustrate the benefits of prediction, we compared the performance with and without prediction in Fig. 4 and Fig. 5. We obtained these results by downsampling the channel per each tone by a factor of 10. The observation period used to collect the channel coefficients and estimate the prediction coefficients was about $2.5xT_c$. The prediction horizon, for which low estimation error of the future channel coefficients is guaranteed, is T_c. At a velocity of $10m/s$ we estimated the prediction order in the range $[60, 80]$, provided by the MDL function. In Fig. 4 the performance gain due to prediction is expressed in terms of throughput. It is possible to see that for high SNR this gain increases up to $5dB$ while for lower SNRs the gain is marginal. In Fig. 5 we compare the performance of the LA algorithm in terms of BER for the case of prediction to the case in which the out of date CSI is used to predict the next transmission mode. While the benefits of prediction are not always significant in terms of throughput, using the predicted channel offers a significant reduction of BER. With prediction, the BER has the flat behavior that is expected when we are loading based on subband error rate. With outdated channel estimates, this mismatch causes fading that reduces BER performance. Comparing Fig. 5 and Fig. 4, this shows that LA algorithm can also provide significant improvement in BER, even though the throughput may not be significantly different. Similar results were also observed in [3].

The fact that we considered only short windows was not an oversight. When $T_w > T_c$ the channel changes too much in a given window to reliably predict it's future values. This is not a severe limitation; when the window is long the performance is really determined by the subband error rate averaged over the channel statistics. Since the parameters of the channel change on a much slower basis, we can reliably predict the future mode from the previous mode [5]. Of course prediction can still be used to adapt to other parameters of channel that vary faster such as the Doppler or delay-spread, but we reserve this for future study.

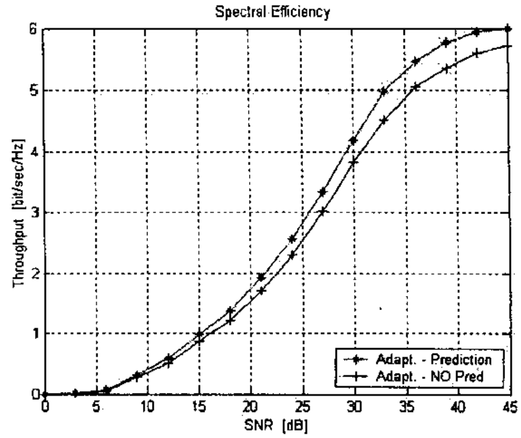

Figure 4: Throughput with and without prediction.

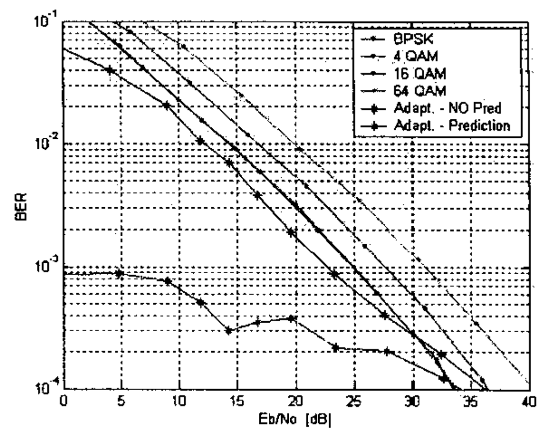

Figure 5: BER with and without prediction. Comparison with fixed mode transmission.

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