A Finite State Modeling for Adaptive Modulation in Wireless OFDMA Systems

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***Abstract*—WiMAX is considered one of the most prominent technologies for providing a Broadband Wireless Access (BWA) in a metropolitan area. Among several strategies, the adaptive modulation techniques has been selected for providing an efficient non line of sight (NLOS) coverage. Adaptive modulation allows the system to adapt the modulation scheme according to the channel conditions in order to enhance the system performance. This paper deals with the proposal of a state model to be used for the performance comparison of two different adaptation algorithms based on the maximization of different functional costs suitable for use in WiMAX system with an OFDMA physical structure. The algorithms performance has been derived and compared in terms of error rate and throughput. The algorithms show a significant improvement of the system performance compared with the static case, i.e., no adaptive modulation; in particular each algorithm is suited for satisfying different QoS requirements.**

I. INTRODUCTION

The consumer interest in multimedia applications and, hence, the increasing demand of high data rate services lead to a grow for research and development in wireless communications. Wireless systems have the capacity to address broad geographic areas without the costly infrastructure required to deploy cabled links. In particular IEEE 802.16 family of standards [1], [2], supported by the WiMAX commercial consortium, concerns the Physical and MAC (Medium Access Control) layers specifications for a Broadband Wireless Access (BWA) communication protocol.

While many technologies currently available for fixed broadband wireless can only provide line of sight (LOS) coverage, the technology behind WiMAX has been optimized to provide excellent NLOS coverage by using performance-enhancing technologies. Among these Orthogonal Frequency Division Multiplexing (OFDM) has been demonstrated as an efficient way to mitigate the adverse effects of frequency-selective multi-path fading by transmitting signals over a number of flat-faded narrow-band channels that are relatively easy to equalize.

The inherent multi-carrier nature of OFDM also allows the use of adaptive modulation according to the behavior of the narrow-band channels, in order to improve system capacity, peak data rate, and/or coverage reliability. To exploit fully the advantages of OFDM in wireless systems, dynamic allocation techniques need to be devised in order to efficiently use resources such as bandwidth, power as well as modulation schemes to increase the spectral efficiency. In particular, in this paper we have considered an OFDMA system in which modulation is set separately for each sub-channel based on channel conditions to optimize the use of network resources and enabling a flexible use of resources that can support nomadic or mobile operation.

Adaptive modulation allows the WiMAX system to select the most appropriate modulation scheme depending on the propagation conditions of the communication channel, e.g., during good propagation conditions a high order modulation scheme is used in order to increase the data rate transmission while during a signal fade, the system select a lower order modulation scheme to maintain the connection quality and link stability without increasing the signal power.

To enhance throughput in future wireless data communication systems, adaptive modulation have been studied and advocated at the physical layer, in order to match transmission rates to time-varying channel conditions; see e.g., [3]–[5], and references therein. The link adaptation algorithms can be designed to maximize the overall network throughput or to achieve target error performance [6]. The first approach may be appropriate for best-effort services but does not meet QoS requirements in term of error performance. The throughput-based scheme typically yields a higher throughput but the error-base approach provides lower error probability.

In this paper two adaptation methods for the WiMAX system has been considered. Our aim is to develop some techniques that maximize the system performance in terms of some QoS metric; in particular our attention has been focused on error probability and throughput. In this sense, among the adaptation methods shown in the following, the first aims to maximize the link throughput, while the second is designed in order to select a target error probability; we have also refined the second technique by considering to achieve the same performance in terms of error probability as in a QPSK constellation for a given SNR value.

II. SYSTEM MODEL

In this section we describe an OFDMA system for mobile wireless MANs, based on a TDD (Time Division Duplexing)

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transmission. The frame structure used in our simulations is one of the possible frame structures shown in IEEE 802.16e standard [2].

In OFDM communication systems, the available spectrum is accessed by a large number of subcarriers. Data symbols are efficiently modulated on these carriers by resorting to the use of the inverse fast Fourier transform (IFFT) in transmission and the fast Fourier transform (FFT) at the receiving end. We focus here, on an OFDM-based wireless communication system where U users communicate with the base station, and N_i carriers are assigned to the i-th user. We have considered a frame duration ($T_F = 8$ ms) and a 1024 FFT for a 10 MHz channel according with the SOFDMA (Scalable OFDMA) concept [7]: the subcarrier spacing is a fixed parameter in all the systems with different bandwidth. Moreover the 1024 OFDM carriers are divided into 32 subgroups, leading to a multiuser factor of 32 users.

In the OFDMA system under consideration, data symbols from one user are not modulated on all the subcarriers, but different users are assigned different subcarriers. The subcarriers of each user are interleaved with the subcarriers assigned to other users by using suitable permutation formulas defined in the IEEE 802.16e standard [2]. We suppose to perform the link adaptation in the downlink. The information bit streams are mapped into a complex value c_k with $k = 0...N_{\mathrm{FFT}} - 1$ using one of the three possible Gray constellations: QPSK, 16QAM and 64QAM. The symbols belong to different communications, with different users. Each subcarrier is mapped independently from the others. The information sequence result to be:

$$s(n) = \sum_{k=0}^{N_{\mathrm{FFT}}-1} c_k \cdot e^{j\frac{2\pi}{N_{\mathrm{FFT}}}nk} \quad n = 0, ..., N_{\mathrm{FFT}} - 1 \qquad (1)$$

We consider a frequency selective multipath fading channel with discrete impulse response $h(n)$ shorter than the cyclic prefix, so that, by removing the cyclic prefix at the receiving end we avoid ISI. By using the cyclic prefix, the convolution between the transmitted signal and $h(n)$ is a cyclic convolution, hence, the received signal can be defined as:

$$\hat{r}(n) = \hat{s}(n) \otimes h(n) + n(n) \qquad (2)$$

where $n(n)$ is a Gaussian random term with zero mean and variance $N_0/2$ introduced by the communication channel. The signal $r(n)$ is obtained by down-converting the received signal and removing the cyclic prefix. Then the signal is processed by the FFT block. The FFT output for the k-th subcarrier, before equalization, is:

$$\begin{aligned} Z(k) &= \frac{1}{N_{\mathrm{FFT}}} \sum_{n=0}^{N_{\mathrm{FFT}}-1} r(n) \cdot e^{-j\frac{2\pi}{N_{\mathrm{FFT}}}nk} \\ &= S(k) \cdot H(k) + N(k) \end{aligned} \qquad (3)$$

where $S(k)$, $H(k)$ and $N(k)$ are the samples in the frequency domain of the transmitted signal, channel impulse response and AWGN contribution on the k-th subcarrier, respectively. The channel coefficients can be written as:

$$H(k) = \alpha(k)e^{\phi(k)} \qquad (4)$$

where $\alpha(k)$ and $\phi(k)$ are the attenuation and the phase rotation of the channel impulse response at the k-th subcarrier, respectively.

The modulation is adapted on a frame basis. We have assumed to have a TDD structure that is more suitable for highly asymmetrical services such as new IP based multi-rate, multi QoS services. The TDD frame begins with the Downlink subframe and a guard time of 5 µs precedes the Uplink part; then a guard time of 5 µs separates a frame from the next one. Due to an odd number of symbols in the frame, as stated in the IEEE 802.16e standard, we have resorted to 40 OFDMA symbols for the Downlink subframe and 39 OFDMA symbols for the Uplink subframe, achieving a quasi equal division. The extension to more asymmetrical division to the TDD structure does not impact to the adaptive algorithm behavior.

III. OFDMA-AM System

In this section the proposal of an Adaptive Modulation (AM) system exploiting the OFDMA capabilities will be described, considering that the modulation order of each complex symbol c_k associated to the corresponding k-th subcarrier $(0 \leq k \leq 1023)$ could be based on the physical channel state, even if, as explained in Section IV, the subcarriers will be considered grouped between them, as foreseen in the SOFDMA approach [7]. For each subcarrier, the channel frequency-response can be considered reasonable non selective; this is one of the main reasons why OFDM is a good solution for multipath frequency-selective fading channels. The *water-filling interpretation* [8] is the basis for our subcarrier by subcarrier adaptive modulation.

By assuming the channel as reciprocal in frequency, after estimating the channel response in the uplink of the current TDD frame, its behavior can be used in the following frame. The estimation of the complex channel coefficients $\hat{H}(k)$ is done by the BS in the Uplink subframe. The BS will use the updated modulation order in the Downlink subframe and also the MSs will map each subcarrier, in Uplink, with the same constellation used for Downlink.

It is important to stress that even if we assume a prefect knowledge of the channel impulse response at the receiver we take into account the delay introduced by the channel estimation algorithm. The modulation is adapted on the basis of the channel state at the previous frame and it can introduce a performance loss in time varying channels as the one considered in this paper.

In this paper two AM techniques based on the channel state information will be introduced. The first, called *Maximum Throughput* (MT) aims to maximize the system throughput without any explicit constraint on target SER (Symbol Error Rate). The second technique, aiming to respect a maximum preset target Symbol Error Rate (SER) imposed on the basis of a target QoS level, is named *Target SER* (TSER). The target value of the SER can be fixed for every SNR (Signal to Noise

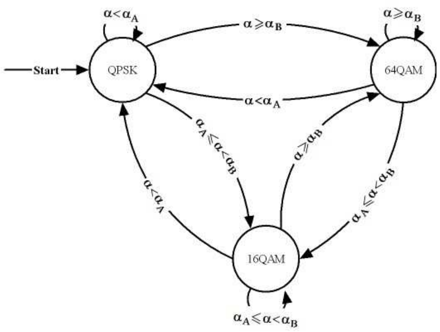

Fig. 1. Moore's three state machine.

Ratio) or can vary with it. In the second case the modulation can be adapted in order to respect the theoretical SER of the QPSK modulation thus obtaining the Minimum SER (MSER) algorithm.

A. AMC block

All the considered techniques use the same system structure. The difference among *Maximum Throughput*, *Target SER* and *Minimum SER* is the thresholds calculation, because they are the inputs for the AMC block controlling the adaptive modulation. In that sense the proposed approach can represent a framework to develop in order to meet different systems requirements and parameters.

The AMC system is modeled as a Moore's three state machine, as represented in Fig. 1. Each adaptation algorithm is basically characterized by two thresholds, representing the changing events between different modulation schemes. By assuming to compare the actual channel attenuation factor $\alpha(k)$ to the thresholds, it is possible to estimate the modulation order to be used in the following frame. In Fig. 1 the two thresholds are represented by α_A and α_B, where α_A is the lower threshold and α_B is the upper threshold. The QPSK state is always used at the beginning of the communication supposing that the BS has no informations about the channel at the initial state. The modulation order is chosen separately for each subcarrier. In the following we refer to the generic channel attenuation factor α omitting the subcarrier index k.

The main difference between the adaptation algorithms described in the following is how the two threshold are calculated; their value strongly influences the behavior of the adaptation algorithm in some performance measurement.

B. Maximum Throughput *technique*

The *Maximum Throughput* is designed in order to achieve the maximum throughput value at a certain channel attenuation factor. The throughput is defined as the average number of correct bit that are received in a symbol time. This technique has no degree of freedom and the thresholds are found by comparing the throughput of the considered modulation schemes [9]. The throughput (η) can be expressed as a function of the channel attenuation factor α for a fixed $\overline{\mathrm{SNR}}$ (the

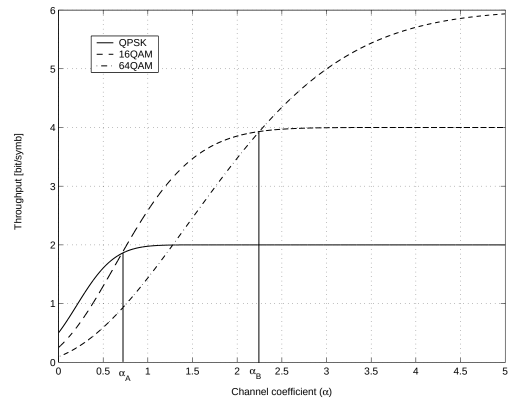

Fig. 2. Thresholds definition for Maximum Throughput technique

average signal to noise ratio), as:

$$\eta(\overline{\mathrm{SNR}}, \alpha, M) = \log_2(M) \cdot [1 - \mathrm{SER}(\overline{\mathrm{SNR}}, \alpha, M)], \quad (5)$$

where M = 4, 16 or 64 corresponds to the constellation order. We can express the theoretical SER expression for an even QAM constellation [8] in function of a certain SNR value for different α values and modulation order as:

$$\mathrm{SER}(\overline{\mathrm{SNR}}, \alpha, M) = \frac{4(\sqrt{M}-1)}{\sqrt{M}} \cdot Q\left(\sqrt{\frac{3\overline{E_s} \cdot \alpha^2}{(M-1)\,N_0}}\right) - \left[\frac{2(\sqrt{M}-1)}{\sqrt{M}} \cdot Q\left(\sqrt{\frac{3\overline{E_s} \cdot \alpha^2}{(M-1)\,N_0}}\right)\right]^2 \quad (6)$$

The *Maximum Throughput* algorithm aims to maximize the total link throughput by interpolating the throughput curves of the used modulation at their maximum value. In this sense the attenuation factor thresholds are those values where the throughput curves for different modulation order have the same value, with the aim of selecting, for each frame, the modulation scheme that maximize the total throughput. Hence, the thresholds for maximizing the throughput are such as:

$$\eta(\overline{\mathrm{SNR}}, \alpha_A, 4) = \eta(\overline{\mathrm{SNR}}, \alpha_A, 16) \quad (7)$$

$$\eta(\overline{\mathrm{SNR}}, \alpha_B, 16) = \eta(\overline{\mathrm{SNR}}, \alpha_B, 64) \quad (8)$$

By doing so this technique allows to select the most efficient scheme in terms of throughput for a certain α value. This technique is more devoted to such services that request the maximum achievable data rate, with a lower sensibility to the SER; in this sense this technique is best suited for video streaming or VoIP services. In Fig. 2 it is shown how the threshold for state change are defined; in particular herein it is supposed to have a SNR=8dB. The figure shows the behavior in terms of throughput for different values of attenuation factor.

C. Target SER *technique*

The *Target SER* technique has been introduced as the method for achieving a target QoS level in terms of error

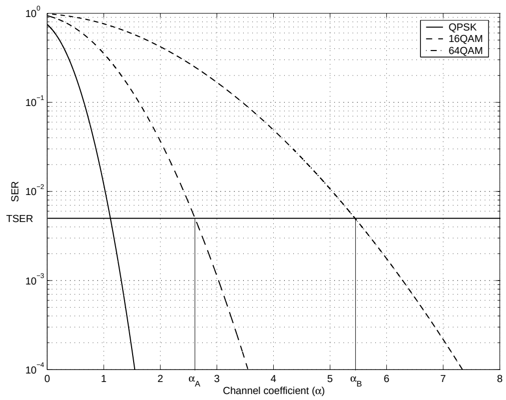

Fig. 3. Thresholds definition for Target SER technique

probability. The *Target SER* technique has one degree of freedom, represented by the imposed target SER. For each value of SNR we can find the SER value as a function of the channel attenuation factor α, and fix two target attenuation factor values α_A and α_B related to the target SER value, where α_A is related to the minimum attenuation factor requested for respecting the target SER with the 16QAM modulation order and α_B is related to the minimum attenuation factor requested for respecting the target SER with the 64QAM modulation order. This technique is more devoted to services requesting a strict respect of a certain maximum SER value, like best effort traffic. The α_A and α_B switching values are those that assure the target SER, for a certain SNR and a certain modulation. The α_A and α_B can be found by selecting:

$$\text{SER}(\overline{\text{SNR}}, \alpha_A, 16) = \text{SER}_{\text{target}} \tag{9}$$

$$\text{SER}(\overline{\text{SNR}}, \alpha_B, 64) = \text{SER}_{\text{target}} \tag{10}$$

In Fig. 3 it is shown how the thresholds for state change are defined; in particular herein it is supposed to have a SNR=8dB and a target SER equal to $5 \cdot 10^{-3}$. The figure shows the behavior in terms of symbol error probability for different values of attenuation factor.

D. Minimum SER *technique*

This technique is derived from the previous TSER technique. Differently, in this case the Target SER is not a fixed value but is changed with the SNR. In particular, the algorithm for the thresholds calculation is based on the consideration that we want to have the same performance results as for the QPSK with a certain SNR. By recalling the theoretical SER expression in (6), the thresholds for switching the modulation are obtained by solving:

$$\text{SER}(\overline{\text{SNR}}, \alpha_A, 16) = \text{SER}(\overline{\text{SNR}}, \alpha = 1, 4) \tag{11}$$

$$\text{SER}(\overline{\text{SNR}}, \alpha_B, 64) = \text{SER}(\overline{\text{SNR}}, \alpha = 1, 4) \tag{12}$$

for a certain $\overline{\text{SNR}}$ value at the receiver. By solving (11) it is possible to find the attenuation factor values that allow to have the same performance as the QPSK modulation without channel attenuation (i.e., $\alpha = 1$).

The main aim of this technique is to achieve the best SER for a certain $\overline{\text{SNR}}$, that is the QPSK SER for such $\overline{\text{SNR}}$, but with the advantage of a higher throughput in the case the α factor allows to use a more efficient modulation order.

IV. Numerical results

In this Section the numerical results obtained by resorting to computer simulations, carried out by using the MUDiSP3 tool [10], will be described. In particular the attention will be focused on performance comparison between the proposed AM techniques. In deriving our simulations we have focused on a WiMAX environment with:

- radio frequency carrier $f_c = 3.5$ GHz;
- bandwidth of 10 MHz;
- maximum Doppler deviation $f_d \simeq 408$ Hz (mobility terminals up to 125 km/h);
- coherence time $T_c \simeq 1.03$ ms;
- maximum delay spread equal to 20 μs (worst case);
- ITU-R vehicular channel model A, with 6 paths [11].

Using an oversampling factor 8/7 we can structure an OFDMA symbol over a 11.429 MHz bandwidth. The useful symbol time is $T_b = 89.6$ μs and the guard time T_g is 1/8 of an OFDMA symbol duration, hence, a maximum of 11.2 μs delay spread can be tolerated [7].

In Fig. 4 a performance comparison between the theoretical SER for the three static modulation orders and the simulated SER for the three proposed techniques is shown. For what concerns the *Target SER* technique we have considered two different values to be respected by the adaptation algorithm, that are SER = $7 \cdot 10^{-2}$ and SER = $5 \cdot 10^{-3}$. In Fig. 5 the performance comparison in terms of throughput is shown. The throughput is expressed as the average useful bits per symbol for different $\overline{\text{SNR}}$ values at the transmitter side, as in (5). The best SER performance, quite similar to the QPSK, can be achieved by using either the MSER or the TSER (with Target SER = $5 \cdot 10^{-3}$). However the MSER technique permits to achieve a higher throughput as it is shown in the Fig. 5. It is evident that if the Target SER value is higher the SER performance worsens. The SER curves behavior is reversed in the Throughput curves. From the throughput point of view the best case is represented by the MT method that is always higher of the static modulation throughput.

By observing Figs. 4 and 5 it is evident that the choice of the best method depends on the specific system requirements. While the *Maximum Throughput* technique and the *Minimum SER* technique represent two methods well suited for application requesting a strict respect of, respectively, throughput or SER, the *Target SER* technique is well suited for generic application where a target SER can be adjusted following the requested QoS level. As a matter of fact, the *Taget SER* technique has been designed for respecting a generic target SER, with the possibility of switching that value whenever the QoS requirements of a specific application request it.

Moreover it is possible to see that by considering different target SER we can have different performance also compared to MSER technique; in case we select a low target SER we will

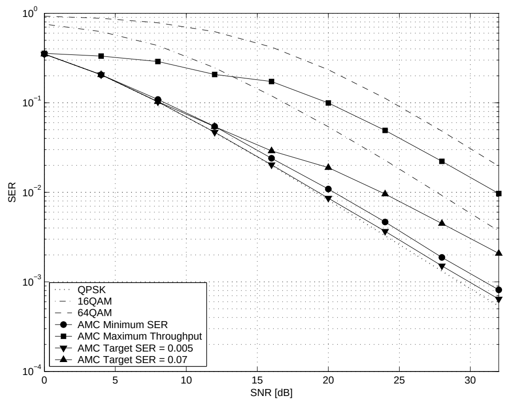

Fig. 4. Performance comparison in terms of SER.

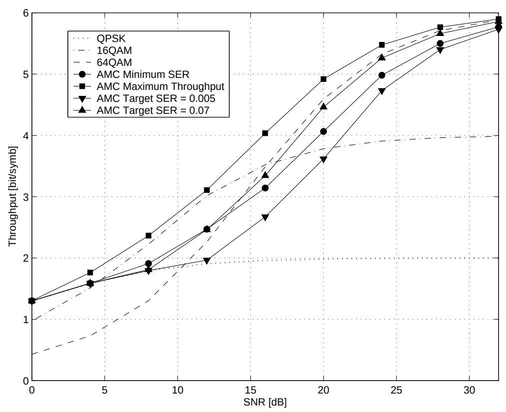

Fig. 5. Performance comparison in terms of throughput.

have better performance that MSER because the target is lower than QPSK performance with the results that we will exploit more infrequently the higher order modulation schemes.

In Fig. 6 it is possible to see the usage in percentage for each modulation scheme for the three considered techniques, by assuming, as an example three $\overline{\mathrm{SNR}}$ values, that are 8dB, 16dB and 24dB. These results confirm that the Maximum Throughput technique allows to use as much as possible higher order modulations. The TSER scheme presents a reduced use of high order modulations if the SER target is particularly stringent (TSER1 and TSER2 correspond, respectively, to the cases TSER=$5 \cdot 10^{-3}$ and TSER=$7 \cdot 10^{-2}$). Finally, the Minimum SER technique presents a good exploitation of high order modulations with results quite similar to the TSER with higher target SER value.

We can observe that the MSER represents a good trade-off between throughput and SER performance. The *Minimum SER* technique results in better performance in terms of SER (because it aims to respect the best modulation order in terms of SER), but with higher throughput values because the α factor fluctuations are exploited with the aim of obtaining a higher throughput.

In any case it is possible to see that the use of an adaptive link control algorithm based on modulation change allows

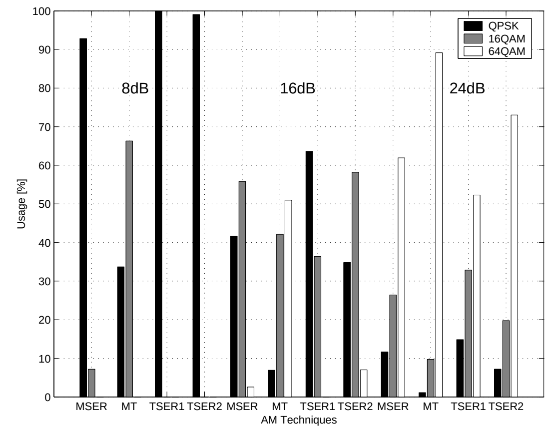

Fig. 6. Percentage of utilization for the used modulation orders.

the system to use the best modulation order in any physical environment by respecting the QoS constraint.

V. Conclusion

In this paper the adaptive modulation issues for a wireless OFDMA based system have been considered. These techniques are based on the physical channel estimation on the uplink, and select the best modulation and coding scheme by using a three state model. Three techniques have been introduced with the aim of minimize the SER, maximize the throughput or select the best modulation order for a certain SNR value. The proposed techniques allows to respect with a different flavor the QoS in terms of SER or throughput, even if all of them shows advantages respect to statical modulation schemes.

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