Adaptive Interleave-Division Multiple Access – A Potential Air Interface for 4G Bearer Services and Wireless LANs

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***Abstract*— A novel adaptive modulation/channel coding scheme based on interleave-division multiple access (IDMA) in conjunction with reliability estimation is proposed. Unlike conventional adaptive modulation/channel coding schemes where the data rate is changed by switching between different modulation schemes and/or channel codes, in the proposed scheme the data rate is adapted by increasing or decreasing the number of signature sequences given a pre-defined target bit error rate. Possible applications include 4th generation wireless bearer services, wireless LANs, and ad-hoc networks.**

***Index Terms*— Wireless communications, multiple access technology, adaptive quality of service provisioning**

I. Introduction

Let us consider a heterogeneous network with N users as illustrated in Fig. 1. As part of this network, for the time being we consider only two users U_i and U_j, $1 \leq i, j \leq N$. We want to establish a full-duplex link between these two users, i.e., we assume that a feedback channel is available. The main goal is to maximize the data rate (up to a certain threshold) given a pre-defined target error rate (QoS), even if the channel capacity is unknown. Note that this goal is much different from most theoretical papers, where the aim is to minimize the bit or frame error rate. The assumption of an unknown channel capacity of the duplex link under investigation is particularly essential in realistic scenarios and disturbances. If a pre-defined maximum data rate is obtained (e.g., 1 Mbits/s) at the target error rate (e.g., 10^{-4}), our second goal is to reduce the transmit power.

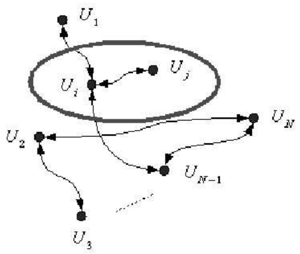

Fig. 1. Heterogeneous network with N users.

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In order to approach the main goal, we need a transmission scheme which is capable to operate close to the capacity limit. Conventionally, adaptation is done by switching between different data rates and/or channel codes. In the current paper, the data rate is adapted by switching between a different number of superimposed signature sequences, similar to the multicode transmission concept in UMTS [1]. For convenience, each signature sequence is referred to as a *layer* in the following. Due to layer-specific spreading, the corresponding receiver is able to recover the original data streams. A popular technique in this context is *code-division multiplexing* (CDM). In CDM, spreading is done layer specific. An alternative is *interleave-division multiplexing* (IDM), where interleaving is done layer specific, cf. Fig. 2. Due to chip-by-chip interleaving [2], [3], [4], IDM outperforms CDM as shown subsequently. Note from Fig. 2 that in (pure) IDM forward error correction and spreading are not user specific. Forward error correction and spreading can be done jointly using a low-rate code.

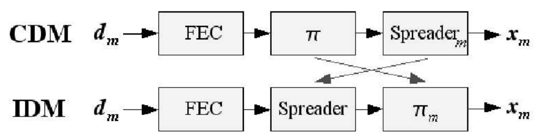

Fig. 2. Block diagram of CDM and IDM, respectively. The index m denotes the m-th layer.

The remainder of this paper is as follows: In Section II the system proposal under consideration is introduced. Numerical results featuring the basic functionality of the system proposal will be given in Section III. Finally, conclusions will be drawn in Section IV.

II. System Proposal

Throughout this paper, the complex baseband notation is used. Vectors are denoted in bold face.

Fig. 3 illustrates a block diagram of the IDM-based system proposal. First of all, the info data stream of user U_i is partitioned into layers of equal lengths. The data of the m-th layer is denoted as $\mathbf{d}_m$, $1 \leq m \leq M$, where M is the number of active layers. All layers are separately encoded

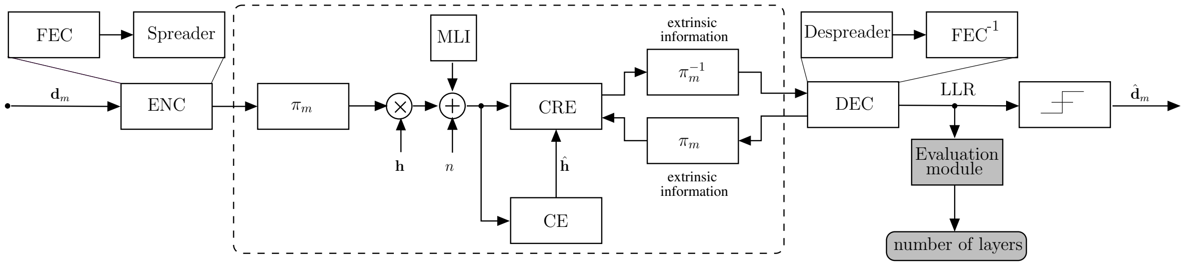

Fig. 3. IDM-based system proposal.

by the same forward error correction (FEC) encoder and spreader. Then, layer-specific interleaving is done by a chip-by-chip interleaver π_m. Assuming a linear channel, the impact of the transmission channel can completely be represented by a vector $\mathbf{h} = [h_0, \ldots, h_L]$, where h_l denotes the l-th (possibly time-varying) coefficient of the equivalent discrete-time intersymbol-interference (ISI) channel model, and L denotes the effective memory length. In Fig. 3, the contribution of the remaining layers are referred to as *multilayer interference* (MLI). Multilayer interference corresponds to *multiuser interference* (MAI) in case that multiple users are simultaneously active. Hence, the interference considered here consists of ISI and MLI. Additionally, at the receiver input additive white Gaussian noise is assumed.

An optimal receiver in the sense of a-posteriori probability detection is prohibitive, because the computational complexity exponentially increases with the number of layers. Therefore, a reduced-complexity receiver is considered here. According to the "turbo principle", the tasks of interference rejection and decoding are done separately, where extrinsic information is shuffled between these two devices. Specifically, (multilayer and intersymbol) interference rejection is done by a so-called *chip reliability estimator* (CRE). For example, the CRE may be based on the assumption that the interference is Gaussian distributed in order to reduce its complexity. Reliability information (usually called extrinsic information) is passed on to the decoder. The redundancy of the spreading and channel code is used in order to obtain an information gain. Estimates $\hat{\mathbf{h}}$ of the channel coefficients $\mathbf{h}$ are provided by a channel estimator, which will be explained later on. After a certain number of iterations, a final decision is done.

It is important to note that besides delivering hard decisions after the final iteration, the receiver under consideration internally computes reliability values. Based on the perception that the outputs of a (true) a-posteriori probability receiver can be used in order to obtain unbiased estimates of the bit error rate without having access to the transmitted data stream [5], in [6] it has been proposed to use a-posteriori probability values (or the corresponding log-likelihood ratio (LLR)) directly for the adaptation of adaptive transmission schemes. Given the log-likelihood values of the m-th layer, the actual error rate of this particular layer can easily be computed. Within an *evaluation module*, this actual error rate can be compared with the pre-defined target error rate, which serves as the desired quality of service. Note that with an increasing number of layers the MLI continuously raises, and hence the error rate degrades. The novel idea here is to increase the number of layers if the actual error rate is below the target error rate, and to decrease it otherwise. Hence, the evaluation module determines the number of layers which are transmitted in the next burst. All what is needed then is (i) a transmission scheme which operates sufficiently close to the channel capacity and (ii) a low-cost receiver which sufficiently approximates an a-posteriori probability receiver so that the reliability values are meaningful. We believe that both demands are sufficiently satisfied by the transmission scheme under investigation. Further note that a knowledge of the channel characteristic is not necessary. This is an essential advantage compared to conventional adaptive modulation/channel coding techniques, where the adaptation is based on the channel characteristics. Having tens or even hundreds of layers in mind, another advantage is the comparably small granularity of achievable data rates.

The basic functionality of the system proposal has been demonstrated by an experimental modem. Possible applications include 4th generation wireless bearer services, wireless LANs, ad-hoc networks, (underwater) sensor networks, among others.

Some details concerning the proposed transmitter and receiver are described in the following subsections.

A. OCDM/IDMA

Up to now, we took only one active transmitter U_i into account. All layers considered so far belong to the same user, i.e., only multiplexing (IDM) instead of multiple access (IDMA) has been studied. In case of multiplexing, it is not necessary to distinguish the data streams by assigning different chip-by-chip interleavers. The corresponding MLI can completely be avoided by choosing orthogonal code sequences (such as Walsh-Hadamard codes, for example). This motivates the idea of bunching multiple layers in a group. Within the group, orthogonal code sequences and the same interleaver shall be used. This technique is referred to as *orthogonal code-division multiplexing* (OCDM).

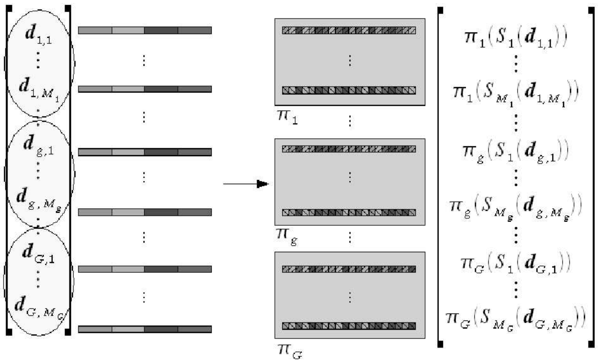

Fig. 4. Principle of OCDM/IDMA.

In case of multiple active users, a natural extension is to assign one ore more groups to each user, where different groups are separated by group-specific chip-by-chip interleavers. The corresponding hybrid scheme has been proposed in [7], [8]. It is referred to as *orthogonal code-division multiplexing/interleave division multiple access* (OCDM/IDMA). OCDM/IDMA is a generalization of IDMA [2], [3] as well as of m-OCDMA [9], [10]. The principle of OCDM/IDMA is illustrated in Fig. 4. In this figure, $\mathbf{d}_{g,m}$ corresponds to the m-th layer of the g-th group, where $1 \leq m \leq M_g$, $1 \leq g \leq G$, and M_g is the number of active layers of the g-th group. All layers are of equal length L, i.e., the chip duration is assumed to be the same. The groups are generally asynchronous, whereas the layers within groups of one user are always synchronous. The layers do not necessarily have to have the same transmit power. Layers which carry most significant data, i.e. typically the lower layers, may be assigned a higher power than those layers which carry least significant data, facilitating unequal error protection. The transmitted signal can be written as

$$\mathbf{x} = \sum_{g=1}^{G} \sum_{m=1}^{M_g} \pi_g(S_m(\mathbf{d}_{g,m})) \cdot e^{j\phi_g}, \tag{1}$$

where $S_m(\mathbf{d}_{g,m})$ denotes the chip sequence corresponding to $\mathbf{d}_{g,m}$, and ϕ_g a group-specific phase offset, which is defined in order to exhaust the available signal space. This phase offset shall be known at the receiver. In the following numerical results, we set $\phi_g = (g-1)\pi/G$.

In Fig. 5, Tanner graphs featuring CDM(A) without interleaving, CDM(A) with interleaving, IDM(A), and CDM/IDM(A) are plotted. These graphs illustrate that CDM/IDM(A) outperforms the other techniques.

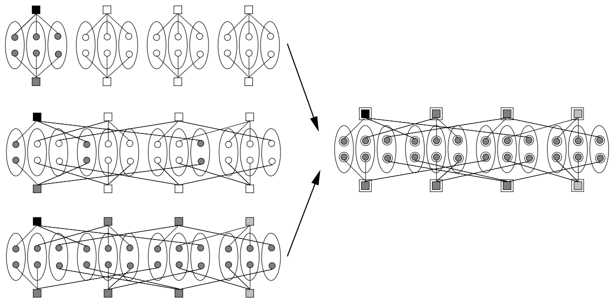

Fig. 5. Tanner graphs featuring CDM(A) w/o interleaving (top), CDM(A) with interleaving (middle), IDM(A) (bottom), and CDM/IDM(A) (right).

B. Pilot-Layer Aided Channel Estimation (PLACE)

Channel estimation is an important task for at least two reasons: Maximizing the data rate implies that we want to operate as close as possible to the channel capacity in all fading conditions. Secondly, reliable channel estimates $\hat{\mathbf{h}}$ are required in order to obtain meaningful reliability values at the output of the CRE even after many iterations. Channel estimation is particularly difficult for time-varying channel conditions.

In order to solve all these requirements, a novel channel estimation scheme called *pilot-layer aided channel estimation* (PLACE) is proposed. For each user the first layer carries training symbols, which are known at the receiver. Therefore, this layer is referred to as the *training layer*. (Some symbols of the training layer may carry info bits or signaling bits (such as the number of active layers). For reasons of simplicity, however, we assume a pure training layer in the following.) This training layer is proposed to be transmitted continuously. This is particularly helpful in the acquisition phase and for re-synchronization after a deep fade. The data layers are linearly superimposed, depending on the pre-defined QoS and the current channel condition.

PLACE is similar to the burst structure of the uplink dedicated physical channel in UMTS [1]. In the latter case, however, training data (and control information) are carried in the quadrature component, whereas the info bits are carried in the inphase component. PLACE is also similar to superimposed pilot-aided channel estimation [11]. It is anticipated that due to the chip-by-chip processing fast fading channels can be tracked.

C. Adaptation Strategy

A reasonable adaptation strategy is shown in Fig. 6, where M is the number of active layers, M_{max} is the maximum number of layers (according to the maximum pre-defined data rate), dM is an incremental number of layers, P is the actual transmit power, P_{max} is the maximum transmit power (for all superimposed layers), P_{min} is the minimum transmit power, dP is an incremental power difference, P_{est} is the error rate estimated by the evaluation module, P_t is the pre-defined target error rate, and ϵ is a threshold value. For example,

$$\begin{aligned} dM &= \lceil K_1 \cdot |\log(P_{est}) - \log(P_t)| \rceil \\ dP &= K_2 \cdot |\log(P_{est}) - \log(P_t)|, \end{aligned} \qquad (2)$$

where K_1 and K_2 are suitable positive constants.

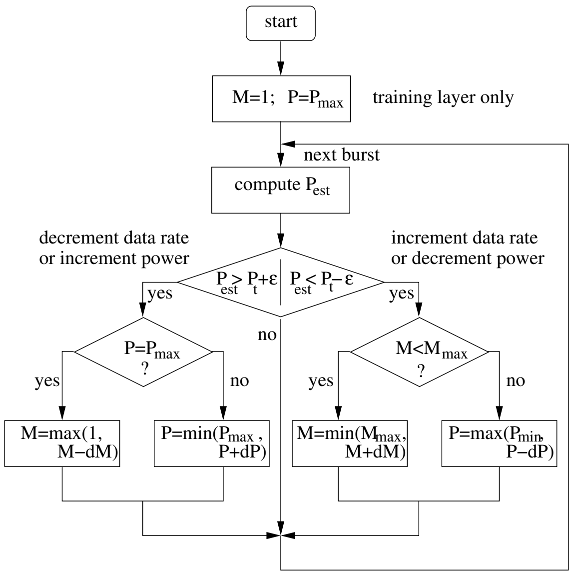

Fig. 6. Flow diagram of proposed adaptation strategy for a single user.

III. Numerical Results

In Fig. 7 simulation results for a single active user on the AWGN channel are illustrated. All curves are for the case without FEC encoding. Perfect channel knowledge in conjunction with the Gaussian CRE of [2], [3] is assumed. The bottom curve represents binary antipodal signaling; the corresponding bit load is $b = 1$ bit/symbol. This curve serves as a benchmark. The remaining three curves are for a bit load of $b = 3.75$ bits/symbol. The top curve shows the performance of m-OCDMA [9], [10]. The next curve shows the performance of pure IDMA [2], [3], [4]. The final curve shows the performance of OCDM/IDMA as proposed in [7], [8]. IDMA and OCDM/IDMA asymptotically approach the single-user bound, being much more bandwidth efficient, however. It is interesting to see that even on the AWGN channel OCDM/IDMA outperforms IDMA by about 2 dB. More importantly, however, is the fact that soft-decision Monte Carlo simulations [5] deliver essentially the same results as conventional hard-decision Monte Carlo simulations, see Fig. 7. Therefore, the soft outputs of the CRE are indeed suitable for system adaptation. Similar results have been obtained for IDMA with FEC encoding.

IV. Conclusions

The proposed system appears to be a suitable candidate for future wireless communication systems. Its strength is

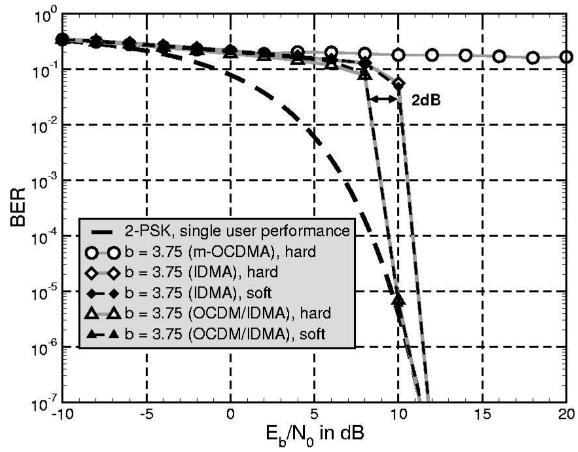

Fig. 7. Monte Carlo simulation results.

chip-by-chip processing in conjunction with adaptation on the physical layer.

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