Adaptive Link Layer Strategies for Asymmetric High-Speed Wireless Communications

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***Abstract*—A forward error correction (FEC) strategy and a medium access control (MAC) protocol that are thoroughly tailored to complement and support a high-speed asymmetric physical layer design based on equalization and precoding is presented and fully discussed. Both proposals exhibit a high degree of adaptability and flexibility, which allows for increased data throughput while providing a wide range of quality-of-service requirements. Fast link layer adaptation is made possible through the joint design of link and physical layers. The adaptive FEC algorithm is based on the use of variable-rate trellis coded modulation with fast channel estimation, while the MAC protocol employs a centralized, dynamic slot allocation technique. The overall system design is shown to achieve high spectral efficiency, while minimizing energy consumption at the portable unit.**

***Index Terms*—Adaptive forward error correction, asymmetry, data link layer, medium access control protocol, precoder.**

I. Introduction

An ASYMMETRIC concept has been recently developed for the physical layer design of high-speed indoor wireless networks [1]. Asymmetric intersymbol interference (ISI) mitigation combines the use of a reverse-link decision feedback equalizer (DFE) with a forward-link Tomlinson–Harashima precoder (THP) in order to relieve the portable station (PS) from the energy demanding tasks related to channel equalization. This is achieved by exploiting the reciprocity of the wireless channel under a time division duplex (TDD) scheme whereby the slowly-varying channel parameters are estimated during reverse DFE-based transmission and then used by the THP for fixed preequalized forward transmission. As a result of this strategy, a remarkable reduction in the complexity of the PS is achieved. This paper shows that the asymmetric concept has interesting and positive implications on the design of some of the link layer elements of the system.

It has been widely acknowledged that link layer adaptability is a key element in designing efficient and reliable wireless data networks [2]–[4]. Adaptability is needed at the access control sublayer in order to accommodate the elastic demand of packet data and multimedia in a multiuser environment. Adaptive channel coding is also required to satisfy different data rates and quality-of-service (QoS) requirements while simultaneously compensating for time- and space-varying channel conditions. The quality of the wireless channel, expressed in terms of the signal-to-noise ratio (SNR), is a random variable showing wide variations throughout the cell. By adapting the data rate to follow the SNR variations, cellular spectral efficiency can be increased. However, fast and accurate channel estimation is required to make adaptive strategies feasible.

A recent trend in wireless systems has recognized that the design of adaptive link layer strategies should be closely linked to the physical layer design [3]. An example of this joint approach for system design are time division multiple access (TDMA) systems such as general packet radio service (GPRS), which use dynamic time slot aggregation, channel monitoring, and adaptive coding to optimize data throughput [2]. The centralized asymmetric system resembles TDMA systems in the sense that users are time-multiplexed and that variable data rates can be achieved through time slot aggregation. Furthermore, the joint physical/link layer design approach is remarkably appropriate in this case, given some specific properties of the physical layer that are exclusive of asymmetric systems.

1) As a consequence of the ISI mitigation strategy, reverse link training must immediately precede forward transmission. This condition imposes certain restrictions on the procedures for medium sharing that are reflected on the need for a specific medium access control (MAC) protocol.
2) Precoded transmission delivers an ISI-free signal at the PS receiver. This allows for the straightforward use of trellis-based techniques for channel coding on the forward link, that would otherwise be difficult to implement (i.e., convolutional coding combined with DFE, see [5]).
3) The asymmetric physical layer concept is restricted to slowly varying wireless environments, such as offices, homes, etc. This fact has interesting consequences on the channel coding scheme and its adaptation.
4) Per-frame adaptive channel estimation is available at the base station (BS) as a by product of the precoder adaptation and the exploitation of channel reciprocity. Information obtained during the reverse training stage can be used at the BS to adapt the coding scheme employed on the THP-based forward link. This eliminates the need for the transmission of side channel information.

This article presents a set of techniques for link layer operation of asymmetric systems. These include an adaptive forward error correction (FEC) algorithm for the forward (precoded) link and an asymmetric MAC protocol that has been carefully suited to support the broadband asymmetric architecture and the adaptive FEC scheme. Section II reviews the asymmetric physical layer concept and some of the most relevant performance

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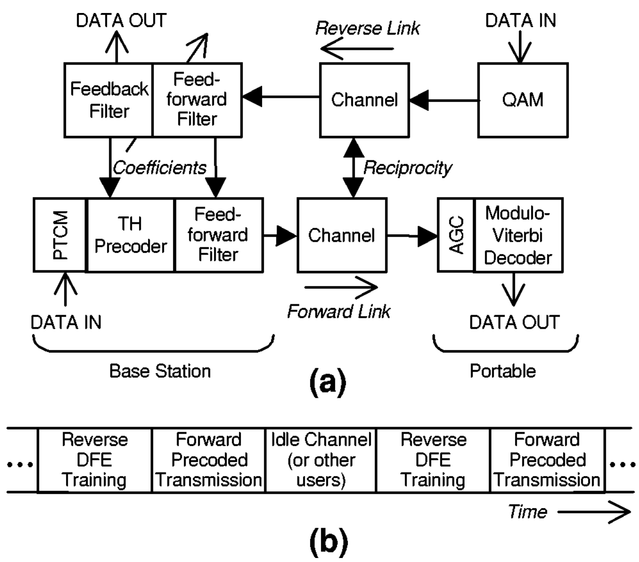

Fig. 1. Asymmetric wireless system architecture. (a) Block diagram. (b) Alternating TDD sequence of operation. Forward trellis coded modulation (TCM) coder is adaptive. Up-link channel coding is not shown (optional).

indicators. Section III presents and discusses the adaptive FEC scheme, including simulation results. Section IV is a description of the proposed asymmetric MAC protocol and some of its performance metrics. Section V summarizes the work.

II. Asymmetric Physical Layer Design

Fig. 1(a) presents a simplified block diagram of the centralized high-speed asymmetric wireless communication system that motivated the design of the adaptive link layer techniques presented in this paper [1]. The reverse link is DFE-based, while the forward link relies on the use of precoding. Both links operate on a packet basis, alternating the use of the same bandwidth allocation under a TDD scheme. The highly asymmetric configuration seeks to concentrate most of the tasks related to the operation of the link at the BS in order to minimize energy consumption at the PS. Transmission on the reverse link is initiated by sending a training sequence from the PS to the BS in order to "learn" the channel [Fig. 1(b)]. Adapting a DFE at the BS receiver does this. Once the coefficients have sufficiently converged, data can be sent by the PS, while the DFE at the BS continues to be adapted under a decision-directed mode. Following the completion of reverse link transmission, the most recent set of coefficients from the DFE is loaded into a fixed preequalizer consisting of a THP [6] and a linear feedforward filter (FF). Forward link transmission follows immediately. The use of the same carrier frequency for both links ensures that within a small time window $\tau (\tau \ll 1/f_{\mathrm{doppler}})$, the channel can be assumed identical in both directions (reciprocity). The signal received at the PS is virtually ISI-free for as long as the channel remains unchanged. Accurate automatic gain control (AGC) is required at the portable receiver in order to detect the ISI-free signal with minimum error probability. Such AGC requires a short training preamble (a few microseconds) [1].

The system considered in this study employs M-QAM constellations (with M ranging from 4 to 64 on the forward link, $M = 4$ on the reverse link), operating at 10 Mbd over a 2.4-GHz carrier. The half-symbol-spaced feedforward filter (16 taps) and the feedback section (8 taps) of the DFE are adapted using a variable step-size LMS algorithm. Square-root raised-cosine shaping filters with roll-off factor of 15% are placed at both ends of each link. Receivers also include a fifth-order front-end Butterworth filter placed before the A/Ds, which operate at twice the symbol rate. Moreover, ideal synchronization is assumed. The model employed for the wireless channel is time varying. For more details about this system (see [1]).

Since the fixed preequalizer is not designed to track channel response changes in a continuous manner, transmission in the forward direction degrades progressively and can only be maintained for a limited time, as dictated by the coherence time of the channel. Within each forward frame, the SNR at the decision point (slicer SNR) deteriorates progressively, as the channel response changes, until the link quality drops below the minimum required to maintain the link. At this point, reverse link transmission must be resumed to update the DFE and the THP-FF prequalizer. Forward ISI mitigation is, therefore, realized on a per-frame adaptive basis. Using a popular approximation for the coherence time of the channel [7], it can be shown that such Doppler spread corresponds to a coherence time of about 8 ms (at a 2.4-GHz carrier frequency). The maximum forward frame time is set to be 5% of the coherence time of the channel, that is, 0.4 ms (4 Ksymbol). Fig. 2(a) shows the degradation in slicer SNR for an ensemble of 500 trials corresponding to mobiles located 20 meters away from the BS, moving at 3 m/s ($f_{\mathrm{doppler}} = 24$ Hz). As seen, a 4-ksymbol frame time corresponds to an average loss of 5–8 dB in slicer SNR.

It can also be seen from Fig. 2(a) that the slicer SNR obtained after a 1024-symbols training sequence from scratch is different by less than 1 dB from that obtained after a 512-symbols "warm-started" retraining stage, if the coefficients obtained $1/20f_{\mathrm{doppler}}$ and $1/10f_{\mathrm{doppler}}$ seconds before the retraining stage are used as the initial DFE state. This fact has important consequences on the design of the link layer strategies, since it allows for the use of shorter training sequences when a recent coefficient set is available at the BS for the corresponding MS.

Fig. 2(b) also shows that the mean slicer SNR on the fixed precoded link degrades in a way that can be roughly approximated as linear, with a slope that is proportional to the slicer SNR value reached at the end of the training stage. This key observation constitutes the basis of the channel estimation method employed by the adaptive coding scheme: slicer SNR during forward transmission can be estimated from the target value reached by slicer SNR during the reverse training stage.

Inter-cell interference is also a parameter to consider in the design of the proposed system. Worst case simulations not included in this study show that uncorrelated cochannel interference must be at least 20 dB below the desired signal in order to keep degradation of the down-link performance under 3 dB at the end of a 4000-symbol frame. A simple C/I calculation assuming cells with uniform size and uncoordinated TDD operation, coupled with a propagation exponent of four, shows that a frequency reuse factor of seven ensures that cochannel interference will provide the desired cochannel attenuation of 20 dB or

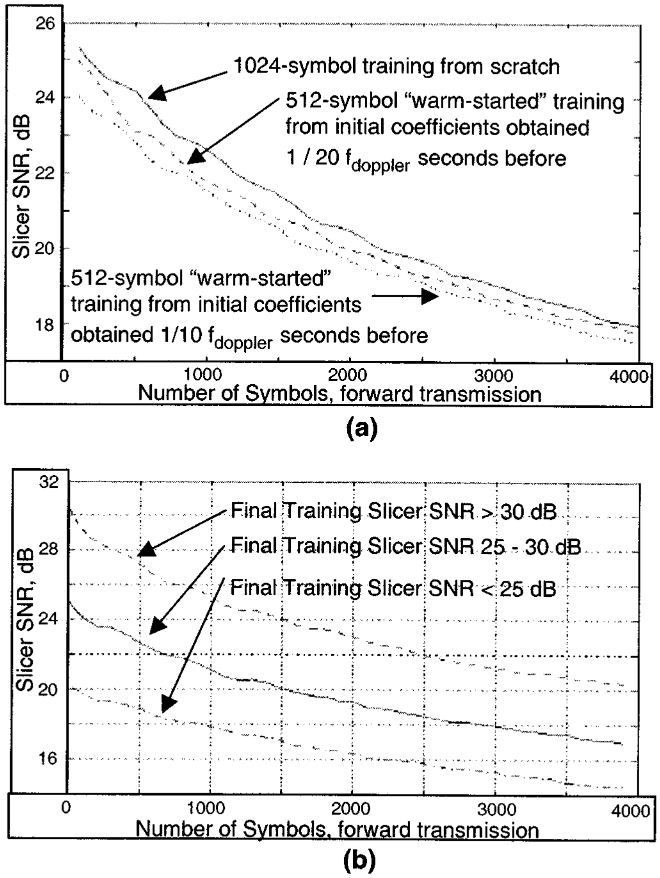

Fig. 2. Performance of the asymmetric wireless system. (a) Average slicer SNR as a function of time and training mode (from scratch versus "warm-start" retraining). (b) Average slicer SNR as a function of time and final slicer SNR value during training stage. 500-channels ensemble, QPSK. Mobile speed is 3 m/s.

more. The propagation exponent of four accounts for the physical obstructions encountered by other cell interference in subdivided office environments.

III. Adaptive FEC

A. General

The error control techniques presented in this section are designed to operate on the forward link of the system, but they are based on the channel estimate obtained from the reverse link training stage and exploit the fact that channel information is available at the BS. It must be remarked that in most applications (Internet, file retrieval, etc.), it is the forward link which must support the majority of the data traffic. Channel coding can also be used on the reverse link in combination with the DFE: however, most proposed schemes are not directly compatible with the use of precoding as postulated in this work [5]. A simple solution for the reverse link that is compatible with the DFE/precoder combined operation and is MS power amplifier-friendly, assumes fixed rate-3/4 TCM using a constant-amplitude 4-QAM constellation (1.5 information bits per modulation symbol). This code is obtained by puncturing out the fourth and sixth bits of every sequence of six consecutive coded bits out of the rate-1/2 code presented in the next section. At the receiver, tentative decisions are employed by the feedback filter to eliminate ISI, while soft decisions are fed to a Viterbi decoder to estimate the data. This scheme provides bit-error rates (BERs) below 10–6 as long as the slicer SNR is better than 7 dB, although it deteriorates sharply below this point since tentative decisions become unreliable as DFE error propagation rate soars.

Error protection on the forward link is based on the use of pragmatic trellis coded modulation (PTCM). The code rate is made adaptive on a per-frame basis and within each frame by changing the size of the transmitted constellation. Target BER and channel estimation are used as inputs to the code rate determination algorithm. Channel estimation is performed by combining information from the physical layer and decisions from an error detection mechanism. Performance is quantified through outage probability and the average data rate. Outage proability is estimated by the fraction of randomly generated channels for which the actual BER is below that predicted by the channel estimation algorithm. The next subsections explain the rationale for the choice of the channel coding scheme, the description of the adaptive rate algorithms and the performance evaluation.

B. Pragmatic TCM

As opposed to DFE receivers, the absence of error propagation in THP systems makes them naturally suited for use in combination with TCM. In fact, precoded TCM has been identified as an attractive option when channel coding is required in equalized systems [8]. In this case, the Viterbi decoder must be slightly modified in order to handle the modulo operation involved in a THP receiver. This is a consequence of the fact that the received constellation in a THP-based system is comprised of several replicas of the original constellation, thus, requiring a special procedure for calculating the Euclidean distances involved in a soft Viterbi decoding algorithm [9].

The motivation behind pragmatic TCM (PTCM) is to achieve variable coding rates using a single conventional Viterbi decoder [10]. This feature makes this family of codes highly attractive for wireless low-power applications, although its use is limited to slowly changing channels, as will be explained later. Fig. 3(a) shows the block diagram of the particular set of PTCM codes selected for this application. One single rate-1/2 convolutional code is used for all constellation sizes. Increasing the number of uncoded bits results in larger constellations and more bits transmitted per symbol. The decoding capabilities that must be sustained by the portable unit are much simpler than with most families of rate compatible TCM schemes, for instance [11] and [12]. This feature is quite important in the design of energy-efficient portable terminals.

Fig. 3(b) shows the 64-state, rate-1/2 convolutional encoder selected and Fig. 4 shows the performance of the corresponding set of PTCM codes for $m = 1$ through 5 bits/symbol. It must be noted that for $m > 1$, the performance of the code is independent of the number of states. The main objection posed to the use of PTCM is the introduction of uncoded bits in the process of signal mapping, which induces parallel branches in the Viterbi decoder [11]. In fast fading situations, the decoder may encounter difficulties in discriminating among parallel branches, sharply degrading the performance of the code. However, indoor

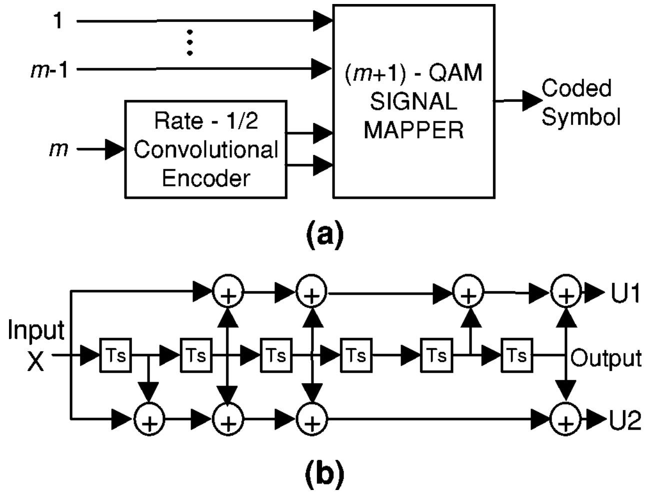

Fig. 3. The set of PTCM codes selected. (a) Block diagram. Rate is m bits per symbol. (b) 64-state, rate 1/2 convolutional encoder employed. Ts is a one-symbol time delay.

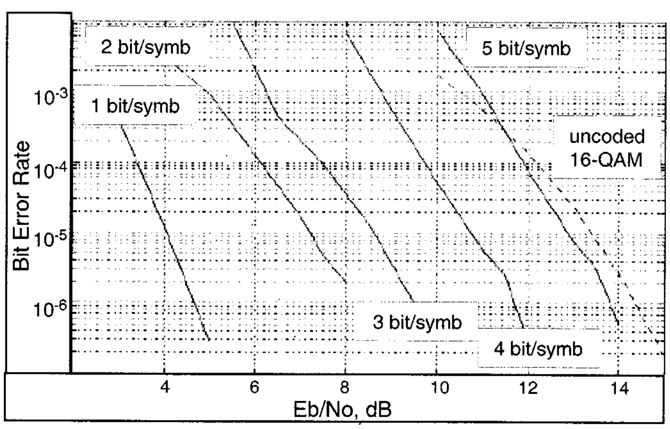

Fig. 4. Performance of the PTCM codes selected.

channels are characterized by slow variations. With Doppler shifts on the order of a few tens of Hertz, they can be modeled in the short term as purely Gaussian channels with slowly changing noise variances. Simulations show that the performance of convolutional codes over a slowly changing AWGN channel can be closely predicted by the static Gaussian case. Fig. 5 plots BER as a function of time for the PTCM code presented above (64-QAM, 5 bits per symbol) operating over a channel with linearly degrading SNR. The recorded BER shows a good match with the BER predicted. (The prediction is obtained by simply mapping of E_b/N_o values into BER using the corresponding curve from Fig. 4.) It can be concluded that although PTCM codes are optimized for AWGN channels, their performance over slowly varying Gaussian channels is approximately the same.

C. Adaptive FEC Algorithms

The selection of the PTCM code rate on the forward link of the system considered is based on the prediction of instantaneous slicer SNR. Such estimation is performed by the BS transmitter on a per-frame basis, as shown in Fig. 6. The estimation is based on the slicer SNR value reached at the end of the DFE reverse link training stage, as well as information from previous

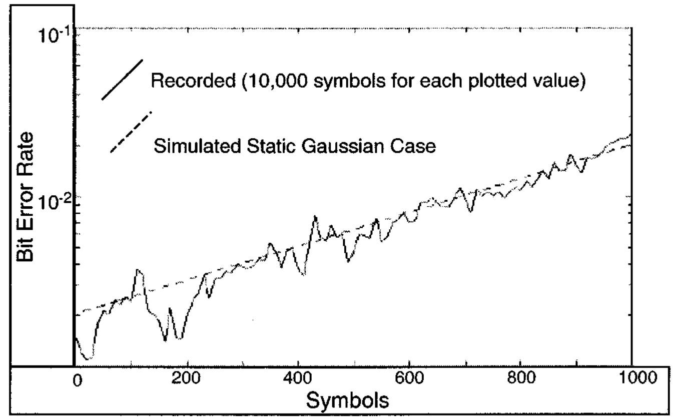

Fig. 5. Perfomance of the PTCM code (5 bits/symbol) on an AWGN channel with linearly degrading Eb/No (1 dB drop over 1000 symbols). Decoding depth is 70 symbols.

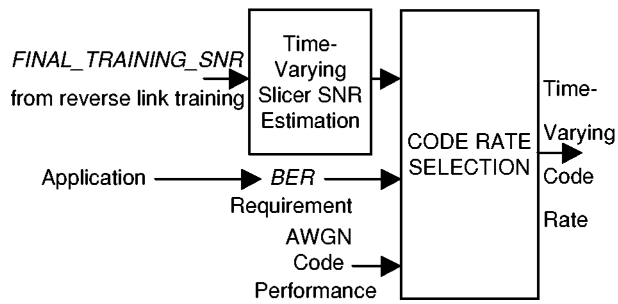

Fig. 6. Information flow for the rate selection algorithm.

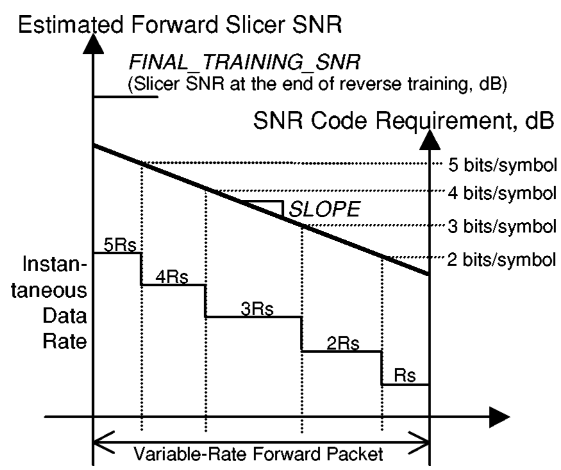

Fig. 7. Model for forward link slicer SNR prediction and subsequent code rate determination.

frames. Simulation results in Fig. 2(b) show that the mean slicer SNR on the fixed precoded link decreases in a way that can be roughly approximated as linear, with a slope that is proportional to the slicer SNR value reached at the end of the training stage immediately preceding. While the linear approximation is not claimed to be the optimal model for the slicer SNR, it will be seen shortly that it allows for an increase in the effective channel throughput.

The model proposed for estimating the instantaneous slicer SNR value at the BS is presented in Fig. 7. Down-link slicer SNR is assumed to be linearly decreasing at a rate give by *SLOPE*, dropping from an initial estimation given by *FINAL_TRAINING_SNR—MARGIN*. As seen, two parameters are required: *SLOPE* and *MARGIN*, while the value of *FINAL_TRAINING_SNR* is read from the training DFE stage in decibel. Two algorithms for channel estimation are proposed.*Algorithm A* uses current frame information only, obtained from the training stage. *MARGIN* is set to a positive constant value and *SLOPE* is calculated in a manner that is directly proportional to *FINAL_TRAINING_SNR*, according to

$$SLOPE = A^{*}(FINAL\_TRAINING\_SNR\text{-}MARGIN) + B$$

where A, B are constants that depend on the rate of change of the channel and can be determined from linear curve fitting of the observed slope values. *Algorithm B* uses current frame information combined with slicer SNR information from the previous frame. Immediately after sending the training sequence, the portable unit reports to the BS the initial and final values of slicer SNR from the previous forward frame. The BS then uses this information to estimate the parameter *SLOPE*. The parameter *MARGIN* is set to a positive constant value. This is the most accurate method of the two, since the BS has fresh knowledge of the rate of change of the channel, which is closely related to the mobile speed.

D. Performance

The algorithms for code rate determination were tested via statistical simulations using an ensemble consisting of 1000 random wireless channels. The channel responses correspond to a dynamic, multipath model based on field measurements [1]. The mobile velocity in each channel is random and uniformly distributed between 0 and 3 m/s and the transmission distance is 20 m. The average RMS delay spread of the ensemble is 75 ns. Received signal power for each channel is lognormally distributed with a standard deviation of 8 dB. *Algorithm A* was tested over single frames, but *Algorithm B*, which makes use of inter-frame information, was tested over the second frame of a sequence of two (inter-frame time uniformly distributed between 50 and 150 μs, that is, 5000–15 000 symbols). For the aforementioned conditions, a linear regression fit for *SLOPE* produced the values of the model parameters $A = 0.95 \times 10^{-3}$ and $B = -9.29 \times 10^{-3}$.

Fig. 8 illustrates the performance in terms of outage probability (BER $= 10^{-4}$) and mean bit rate. Also shown in Fig. 8 is the outage performance corresponding to fixed rate PTCM and uncoded QAM. To better interpret this plot consider the following. If a fixed uncoded binary phase-shift keying (BPSK) constellation (1 information bit per symbol) was used, the outage probability of this channel ensemble would be a little more than 4%. A similar outage (6%) would be achieved using PTCM with fixed rate of 2 information bits per symbol. If *Algorithm A* was used with a convenient value for the parameter *MARGIN*, 6% outage would be achieved at an average rate of about 2.4 information bits per symbol. If *Algorithm B* were used instead, then the average bit rate would reach 3.2 information bits per symbol, representing a throughput gain of 60% over the fixed code rate case. These results corresponds to

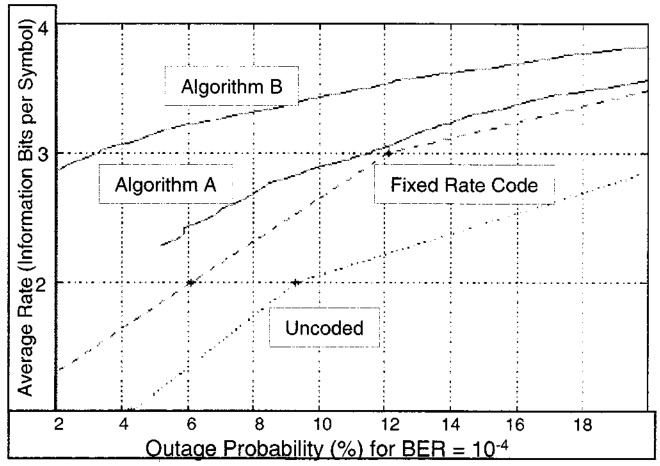

Fig. 8. Performance of the proposed methods for adaptive coding rate. The curves corresponding to Algorithms A and B were generated by varying the parameter MARGIN. Forward length frame is 4000 symbols.

MS located at a fixed distance from the cell: it is expected that, when additional SNR variability due to fluctuating propagation distance is included, the adaptation algorithm will provide even greater throughput gains. In a suggested application scenario, *Algorithm B* can be used whenever there is information available from recent frames, leaving *Algorithm A* as the default choice for the remaining cases, such as after long idle periods.

Fig. 9(a) shows the throughput of each algorithm as a function of the frame length. It can be seen that longer frames reduce the throughput of both algorithms, since in these cases the channel prediction into the future becomes less accurate. The plot also displays the maximum achievable throughput rate, that is, the average rate that could be sent if the transmitter knew *a priori* the instantaneous value of the slicer SNR and the rate of the PTCM code was adjusted accordingly. As shown, there is a significant margin for improvement in code rate determination at the precoded transmitter. However, the proposed methods tend to close the gap when shorter frames are used. It can be concluded that the effectiveness of the channel predictor is also limited by the channel variability, which increases with forward transmission time and the channel Doppler spread. Fig. 9(b) displays the average rate achieved by each of the proposed algorithms for different BER requirements. Overall, these results indicate that PTCM codes are a viable choice for the precoded forward link of an asymmetric system and that the proposed concept for adaptive FEC have a significant impact on the data throughput of the wireless system.

IV. Asymmetric MAC Protocol

A. General

The objective of the MAC protocol is to allow for an efficient use of the radio channel in a manner that is compatible with the centralized architecture and the asymmetric ISI mitigation and adaptive coding strategies, allowing at the same time an energy-efficient design of the portable units. In order to facilitate coordination in channel access and high spectral efficiency, the proposed protocol allows transmission using one of two possible symbol rates.

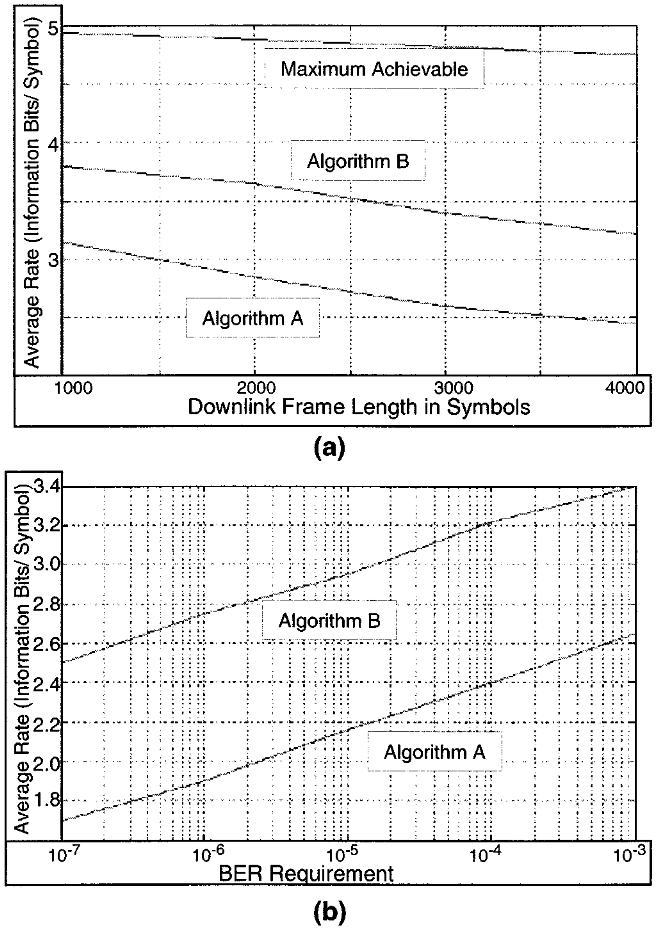

Fig. 9. Performance of the proposed methods for adaptive coding rate. (a) As a function of forward link frame length at an outage probability of 6% ($BER = 10^{-4}$). (b) As a function of the BER requirement at an outage probability of 6%.

1) In the *low rate* transmission mode, the system operates at 1 Mbd with either 1 or 2 bits per symbol (BPSK or QPSK). A 10-chip BS-dependent code is used for spreading each symbol, in a similar way as in the ETSI HIPERLAN standard [18]. The transmitted signal exhibits, therefore, a total bandwidth of 10 MHz. Low rate bursts are used to convey MAC signaling and short data segments on both links, while allowing messages to be detected without the need for equalization.
2) In the *high rate* transmission mode, the system operates at 10 Mbd with a variable number of bits per symbol on the forward link (depending on the rate of the channel code, which is adaptive, as explained in Section III) or 1.5 bits/symbol on the reverse link (as explained in Section II-A). The high rate link is highly asymmetric, based on the use of equalizers on the reverse link and precoders on the forward link, as described in Section II. Precoded forward transmission can only be scheduled immediately after a reverse-link training stage (from scratch, 1024 symbols) or retraining stage ("warm start", 512 symbols).

Medium access is controlled following a multiservice dynamic reservation (MDR) approach that loosely resembles the one proposed in [14]. Channel time is structured as a sequence of frames of equal duration. Each frame is initiated by a low-rate BS broadcast called Beacon. A MAC frame is, therefore, referred to as a beacon repetition period (BRP). The protocol relies on the use of the MAC signaling broadcasted in the beacon to clearly demarcate the slot boundaries for transmission during the remainder of the BRP.

Energy conservation at the PSs is possible by virtue of a number of MAC mechanisms. First, the low bit-rate feature allows for the transmission of short forward and reverse link MAC messages without requiring training sequences. Secondly, by broadcasting the transmission schedule, a mechanism is provided by which MSs can spend most of the time in sleep/doze mode, wake up only to decode the beacon and then to transmit or receive data on the assigned slots. Also, the use of a contention-based random access procedure, which leads to collisions, is limited to signaling messages and occasional short data packets: the bulk of the data transfer is done under contention-free rules. The use of allocation algorithms that provide contiguous slots and the practice of buffering data in order to make the bandwidth requests as large as possible, also contribute to energy savings. Finally, uncoded forward low-speed data transmission is employed to convey information at low power cost for the MS by eliminating the need for channel decoders.

B. Channel Frame Structure

In addition to the beacon, three periods are defined within the BRP: the access period (AP), the static allocation period (SAP) and the dynamic allocation period (DAP). The AP, which is contention-based (slotted ALOHA [15]), is employed by MSs to attempt access to the wireless network. The bulk of data transmission is carried out during the SAP and the DAP. Both are contention-free, high-rate transmission periods used for reverse and forward transmission. SAP slots are defined once and remain valid until a new beacon message removes them. On the contrary, every DAP slot must be announced within the beacon associated with the given frame; they are assigned on a demand basis. DAP is placed at the end of the BRP (Fig. 10), in order to provide the maximum time for the MSs to process the beacon messages concerning DAP slot allocations and act upon them.

The beacon message contains a fixed preamble for sync acquisition, frame delimiters, codewords identifying BS and message length, MAC signaling, and short data segments. The MAC messages, used for slot assignment, are called slot assignment messages (SAMs). Each SAM includes a message ID header (8 bits), slot start time index (12), slot length (10), mobile ID (8) and mode (3-bit word which indicates one of the following: reverse link training, reverse link data, forward link data, AP demarcation, SAP slot addition and deletion) for a total of 41 bits (plus a seven-bit CRC appendix). Slot start time and length are measured in units of 50 μS. The first SAM is used to announce the AP. Then SAP slots are added or removed and finally DAP slots are indicated. SAM's are always implicitly acknowledged when MSs respond on the slots assigned for reverse link training. If the MS does not use the assigned slot, the BS may decide to resend the corresponding SAM for a number of times before it declares the MS out of coverage. BSs can also send null SAMs, which are MAC layer signaling messages not intended

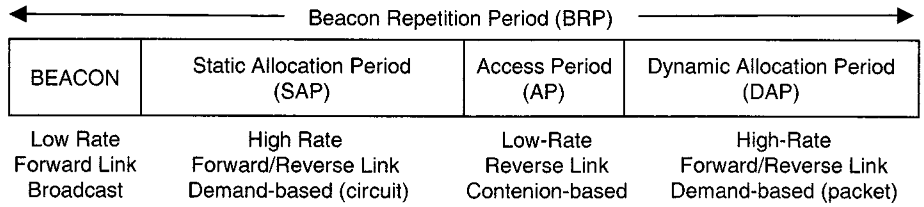

Fig. 10. Time structure of the proposed MAC protocol.

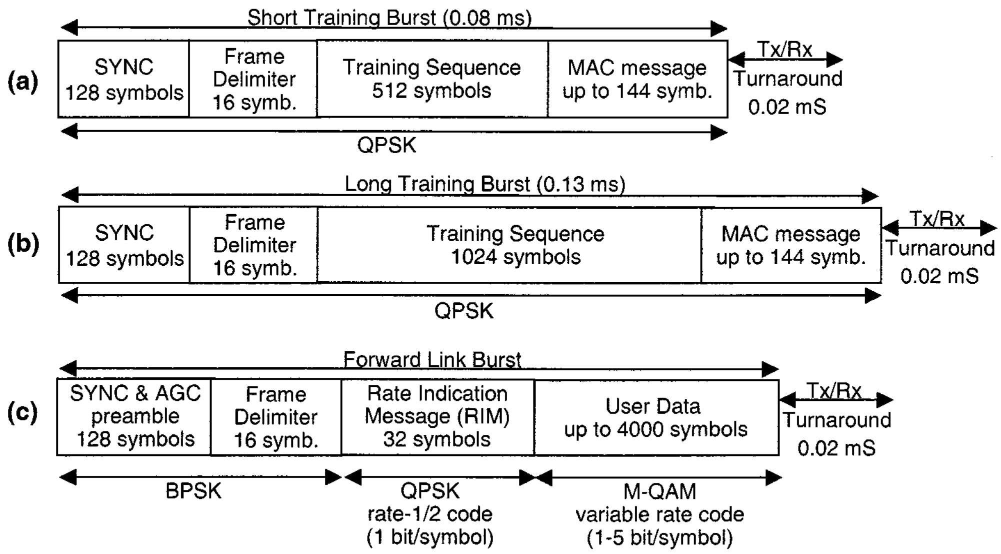

Fig. 11. Some of the burst structures of the proposed MAC protocol. (a) Short training burst (reverse). (b) Long training burst (reverse). (c) Forward transmission burst.

for slot assignment, but for carrying MAC information instead, such as registration acknowledgment. After all SAMs for the current BRP have been transmitted, there exists an option for sending short user data segments, which is employed when data buffers contain less than 40 bytes or when the network traffic is light. Each SAM and short message is protected by a CRC codeword, but error correcting codes are not used to allow for fast decoding of signaling messages and timely response. Since both rates are transmitted at the same energy level, bit energy in the low rate mode is ten times higher than in the high rate mode and, thus, low BER performance is possible without any error correction coding.

The access messages are transmitted during the AP. Access slots are defined to be 50 μs long and the time limits of the AP are defined in the beacon by the BS. A MS wanting to access the system randomly selects a slot index according to the maximum number of slots defined in the beacon message. Two options are considered: access message (MAC layer signaling, such as registration and bandwidth requests) and short user data segments. This latter option is used to transmit data when the MS buffer contains small amounts of data (up to 56 bytes) and a high-speed connection is not justified. Access messages requiring acknowledgment can be explicitly or implicitly answered by the BS during the upcoming beacon messages. For instance, a registration message is explicitly acknowledged, while a SAP or DAP bandwidth request is implicitly acknowledged when the BS grants the required bandwidth. If MS does not receive an acknowledgment to an access message after a number of beacon messages have been received, it may decide to resend the message using a new randomly obtained AP slot index.

Slots are allocated to MSs in the SAP and DAP for one of three different tasks: reverse link coefficient training, reverse link data transmission, and forward data transmission. Slot timing is defined in units of 50 μs. SAP and DAP slot assignments consist of a reverse-link training stage followed by reverse-link data transmission, forward-data transmission or both, always running at the high symbol rate (10 Mbd). The message structure is such that a minimum of 20 μs is always provided to allow for transmit/receive turnaround, needed for TDD operation. The reverse link training stage can be short or long [see Fig. 11(a)]. The BS decides whether the stored coefficients are fresh enough to allow for a "warm start" with a short 512-symbol training sequence, or a 1024-symbol long training sequence is required. As shown in Section II, short training is possible if the most recent set of coefficients is not older than $1/10 f_{\text{doppler}}$, that is, a few miliseconds in low mobility environments. If there is no data scheduled for transmission at the end of the training stage, the MS may send MAC layer messages, such as information regarding slicer SNR from the previous frame, which is required for the operation of *Algorithm B* from Section IV. The signaling messages are encoded using fixed rate-1/2 TCM. If reverse data transmission is requested, the BS schedules additional data slots after the training slots. During reverse data transmission, the physical layer is continuously adapted, in order to follow changes in the channel response, in a decision-directed mode. Thus, reverse link data frames are not bounded and the BS can allocate as many consecutive slots to a single mobile as desired (and available). Reverse link data transmission is always performed using QPSK and a rate 3/4 convolutional code, that is, at 15 Mb/s. The rate-3/4 code is a punctured version of the rate-1/2 code used for the signaling part.

As a consequence of asymmetry, forward data transmission differs from reverse data transmission in several ways. First, forward data frame duration is limited to 4000 symbols, as seen in Section II. Accordingly, reverse training frames should be alternated with forward 4000-symbol data frames. Another remarkable feature of forward transmission is the use of variable rate coding, as discussed in Section III. In particular, it is envisioned that rate determination be accomplished using *Algorithm B* from Section III if the previous forward frame was transmitted less than $1/10 f_{\text{doppler}}$ seconds before and *Algorithm A*, otherwise. Fig. 11(b) illustrates a maximum-length forward data transmission burst. The sync preamble has special characteristics, since it serves also as an AGC training preamble needed for precoded transmission. The sequence is constructed according to the method proposed in [1]. A rate indication message, which is protected using the rate 1/2 PTCM code, is sent to inform the MS of the different rates to be used in the frame. The data block follows, up to 4000 symbols long. The actual number of bits to be accommodated in the forward frame depends on the code rate selection (1000–5000 bits per forward block), which in turn is a function of the channel quality.

In order to maximize the efficiency in the use of the channel, the MAC protocol must favor the use of short retraining sequences and the *Algorithm B* for code determination over the use of long training sequences and the *Algorithm A*. For this reason and to the extent of the possible, consecutive slots within a certain DAP must be allocated to the same MS. This ensures that "fresh" channel estimates are available most of the time at the BS.

C. Performance

The first application scenario considered for the evaluation of the MAC protocol is web browsing (packet data transfer on the DAP). In this situation, N users in the same cell download web pages at an average rate of one page every ten seconds, with interarrival times that are exponentially distributed. Page size S (in bytes) is lognormally distributed: the mean of $\log_2(S)$ is set to 11.5 and the standard deviation of $\log_2(S)$ is 2.3 [16], [17]. The BRP is set to 40 ms and the DAP is set to 10 ms. BER requirement is 10^{-6} at 6% outage, which gives average bit-rate values of 1.9 and 2.75 b/s using *Algorithms A* and *B*, respectively [see Fig. 11(c)]. Assuming that the beacon and the AP combined span 5 ms, this leaves 25 ms for constant bit-rate traffic on the SAP. Fig. 12(a) shows the average delay as a function of N. It can be seen that the delay suffered by the Internet packets is small even for a large number of users. The bottom curve shows the average delay when the DAP and the offered load are doubled (effective load remains the same). The smaller delays observed in this case confirm that larger DAPs can handle traffic better, which is a consequence of the slot allocation discipline and the training and code adaptation policies. It must

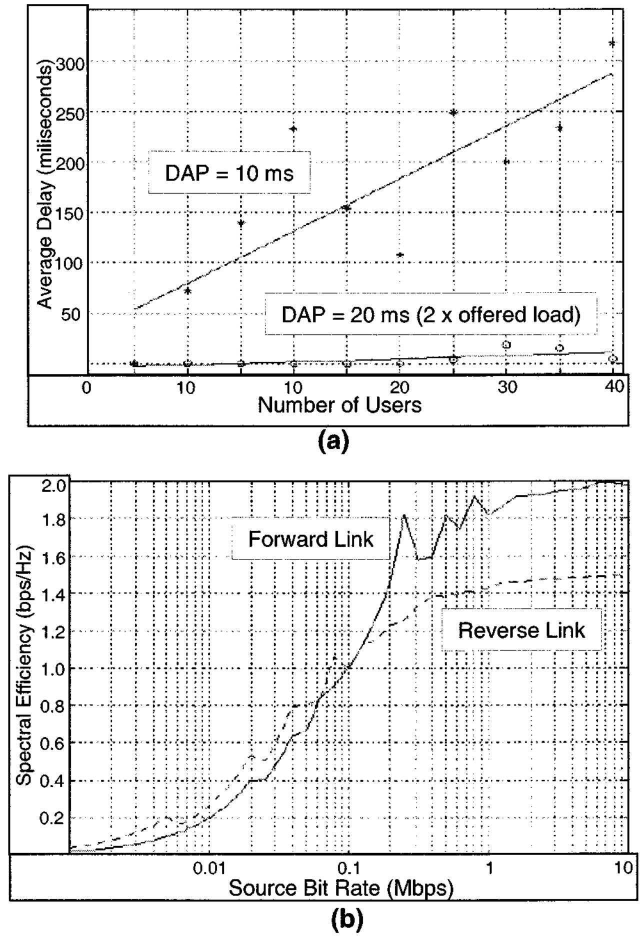

Fig. 12. Performance of the proposed MAC protocol. (a) Average delay for web browsing application scenario. (b) Spectral efficiency for 6% outage at $BER = 10^{-6}$.

be remarked that the upper limit for the duration of the DAP is constrained by the duration of the BRP and the time allocated to the other periods. The duration of the BRP cannot be made too large if the protocol is supposed to handle delay constrained, real-time traffic.

Next, let us consider the overall spectral efficiency of the system, expressed in b/s/Hz. In this case, constant bit-rate transmission on the SAP is considered, with source rates varying between 1 kb/s and 10 Mb/s. Again, only users located at a constant 20-meter radius from the BS are considered. The spectral efficiency is measured by normalizing with respect to the time actually employed for data transfer (overhead periods such as the beacon and the AP will be considered in a later calculation). The plot in Fig. 12(b) shows that the use of the channel with low-rate sources is poor (less than 0.2 b/s/Hz for sources up to 10 kHz). This is due to the fact that relatively long training sequences must be sent every BRP, both for reverse and forward transmission.

The forward link is slightly worse due to turn-around delays and extra training sequences. This situation can be mitigated by not assigning SAP slots to the same low-rate source on every BRP, but using instead a periodic assignment pattern; this technique requires data buffering and, consequently, it introduces some extra latency.

As the data rate increases, the reverse link benefits from relatively training-free transmission (long blocks require one single training sequence) and the forward link benefits from short retraining and better adaptive coding (*Algorithm B* instead of *A*). For high-rate sources, the forward links achieves almost 2 b/s/Hz. Admittedly, when a frequency reuse factor of seven is brought into the picture, the spectral efficiency is reduced by the same factor, giving about 0.28 b/s/Hz. Furthermore, some of the channel time must be dedicated to overhead beacon and AP signaling, resulting in a bandwidth efficiency of about 0.23 b/s/Hz. This is still higher or comparable to most rate-adaptive cellular systems for packet transmission, such as GPRS (0.08–0.2 b/s/Hz) and IS-95-B ($\sim$ 0.1 b/s/Hz in omni cells, $\sim$ 0.3 b/s/Hz in 3-sector cells) [2]. The fact that the spectral efficiency of the proposed system is comparable to these adaptive, optimized cellular data systems proves its potential and the effectiveness of the strategies presented.

The scenario described above for web browsing leaves about 25 ms for constant rate applications to be accommodated in the SAP. Fig. 12(b) can be used to calculate the average time required by constant bit applications on the SAP. For instance, one bidirectional video-conference channel (about 120 kb/s on each link) can be sent with an efficiency of about 1.1 b/s/Hz, that is, each channel would consume about 0.87 ms of every 40-ms BRP. Therefore, about 28 bidirectional video-conferencing channels can be accommodated simultaneously with 40–50 web users. Down-link full-motion MPEG video (about 4 Mb/s) can be sent at approximately 1.9 b/s/Hz, meaning that each video channel would require some 8.4 ms of the SAP, that is, the system has the capacity of supporting three full-motion video channels in addition to 40–50 web users in the same cell. A similar calculation gives a total of 62 simultaneous bidirectional 10-kb/s speech connections.

This figure will certainly be much higher if a BER of 10^{-4} is considered instead of the 10^{-6}, in accordance to the requirements of most speech compression algorithms.

V. Concluding Remarks

The work presented in this paper is a clear example of the benefits of a close interaction between the design of the physical and data link layers of a wireless network. Specifically, it has been shown that the error control mechanisms can be adapted using physical layer information for channel estimation, with a direct impact on data throughput. Furthermore, a MAC design protocol has been presented that is carefully tailored to accommodate the required physical layer functions, as well as the error control mechanisms.

The work confirms the positive impact of fast channel estimation and code rate adaptation in improving cellular efficiency. It also highlights the need to look at the physical layer when targeting fast channel estimation. In equalized/precoded systems, the case for the use of slicer SNR as channel quality indicator has been made, since this parameter best reflects not only the quality of the channel, but also the effect of the equalizer and the residual ISI. When alternating the use of DFE and THP in a TDD basis, the value of slicer SNR from the training stage can be used for channel estimation without the need for the transmission of side information. Finally, the possibility of incorporating information from the past (previous frame) into the channel estimation process has also been proven useful, especially when dealing with slowly varying channels.

Systems that require training such as this asymmetric system based on adaptive equalization, poses both advantages and disadvantages in the context of packetized traffic. On the plus side, the transmission of training sequences can be restricted to precede data bursts and suppressed when there is no traffic, as the proposed MAC protocol postulates; low-rate access provides for fast channel set-up, thus, eliminating the need for "link maintenance" during idle periods. On the negative side, small or infrequent data packets reduce the efficiency of the use of the medium, since training sequences can be long compared with data packets. A mechanism for mitigating this problem that uses low-rate transmission for small packets has been proposed. For the case of constant bit-rate traffic, the use of fixed slot allocations (SAP) also reduce the impact of overhead messages by making unnecessary the exchange of signaling messages once traffic has been established.

A downside of the proposed MAC protocol is the difficulty of establishing up/down-link inter-cell coordination for interference control. This problem can be mitigated through the use of a relatively high frequency reuse factor as suggested, possibly coupled with a cell planing strategy that makes use of physical divisions for cell energy containment. Additionally, inter-cell coordination consisting of synchronization of the beacon and SAP periods, plus fixed allocation of forward and reverse data slots on the SAP and the DAP, would help mitigate the interference problem at the cost of forfeiting a some of the flexibility of the MAC protocol.

The overall system design, including the asymmetric ISI mitigation technique, the adaptive FEC algorithms and the MAC protocol, represent a viable solution that achieves its main goal of reducing the energy consumption at the portable unit in a simple, cost-effective and bandwidth-efficient way, while reaching an spectral efficiency that is comparable to that of some existing wireless data systems.

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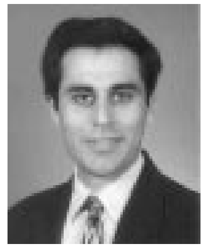

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