ADAPTIVE ERROR CONTROL CODING FOR A MOBILE BROADBAND SYSTEM

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Abstract: The Mobile Broadband System (MBS) under research in Europoe aims at offering to mobile users an ATM-based radio access to the future integrated broadband communications network. It is well known that a high-quality transmission channel can be assumed within the fixed broadband network; however, this is far from being expected in the radio links of the future mobile system.

This paper is concerned with the design of an error control coding scheme for a first stage of MBS implementation. The proposed scheme is a very flexible, hybrid one, which is based on the concatenation of a convolutional inner code, and a Reed-Solomon outer code, and combines ARQ and FEC features in an adaptive manner, under an implicit evaluation of the channel state. It is powerful enough to cope with poor channel conditions and is compatible with both ATM characteristics and those of a wide range of services (including required BER below 10^{-6} with a delay of a few ms). A set of simulation results illustrates the advantages of the proposed error control coding scheme.

I. INTRODUCTION

The aim of MBS (Mobile Broadband System), currently under research in Europe, is to offer to mobile users an ATM-based radio access to the future integrated broadband communication network [1]. The design of an air interface for MBS represents a considerable challenge: the range of services will be very wide, with a variety of service requirements and characteristics, including user bit rates over 100 Mbit/s, much above those allowed by current mobile radio systems; it is necessary to resort to mm-wave frequencies for radio transmission, since the required high gross bit rates imply a very wide band for the system, which is not available below 30GHz. In spite of having to comply with propagation channels which are very hostile to high data rate transmission, the service quality targets can be quite ambitious (e.g. bit error rates below 10^{-6} with a delay of a few ms), not far from those intended within the fixed broadband network. It is well known that a high-quality transmission channel can be assumed in this network: for this reason, no error control is performed on the user information carried by each ATM cell (48 octets), only the header (5 octets) being designed to have some error control capabilities. However, further error control capabilities must be devised for mobile broadband communications, since the corresponding radio link quality can be rather poor.

This paper is concerned with the design of an error control coding scheme for a first stage of MBS implementation, which is compatible with the air interface characteristics and main current specifications, as defined by the European RACE project R2067-MBS; it expresses the views of the authors, not necessarily those of the project. After a brief characterization of the MBS air interface, given in sec. II, the chosen error control coding scheme is presented in sec. III. It is a very flexible, hybrid FEC/ARQ scheme, combining error correction and error detection capabilities in an adaptive manner. A set of simulation results on the achievable performance are reported and discussed in sec. IV, for a specific outdoor scenario [2].

II. AIR INTERFACE OF THE MOBILE BROADBAND SYSTEM

The main assumptions with regards to the air interface of MBS were established as follows [3]:

- A paired band (62-63 and 65-66GHz) is available for radio transmission at mm-wave frequencies.
- A hybrid TDMA/FDMA multiple access scheme is adopted, allowing a flexible use of the frames in accordance with service characteristics, namely the corresponding (fixed or variable) user bit rate.
- Each standard burst carries the user information inherent to one ATM cell (ATM-1 option) or two ATM cells (ATM-2 option).
- Two modulation schemes (4-OQAM and 16-OQAM: Offset Quadrature Amplitude Modulation) can be employed, using the same gross symbol rate in a given class of environments; the choice of the 4- or the 16-OQAM modulation schemes is related to options ATM-1 and ATM-2, respectively.
- First implementations, with somewhat restricted capabilities, are intended for the near future, and upgrading potentials according to technological progress were not neglected; for example, a linear power amplification is not required in the first stage of system implementation, since only

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a 4-OQAM scheme, with low envelope fluctuation, is intended for this stage.

For a first stage of MBS implementation (ATM-1 option only) and operation within most of the foreseen outdoor environments, the current set of air interface specifications [4] includes (see Fig. 1):

- A gross bit rate of 40 Mbit/s;
- 40 standard slots per frame, with a slot duration of 19.2μs and a frame duration of 768μs;
- Duration of the standard bursts equal to 17.8μs, and a guard period of 1.4μs;
- 712 bits per standard burst, with 40 bits for training and tails and 672 bits for coded information related to the contents of the ATM cell (384 "traffic" bits plus a header of 40 bits) and to some additional signalling data;
- Medium-size bursts with 328 bits (288 for coded information), designed to occupy medium-size slots with a duration of $\frac{19.2}{2}$=9.6μs, also for a guard period of 1.4μs.

These specifications (see also Fig. 1) imply an upper bound on the user bit rate per carrier: 384x40/768=20 Mbit/s. This means that a user bit rate of 17 Mbit/s (which is one of the design goals [1]) is expected to be achievable without occupying the whole frame.

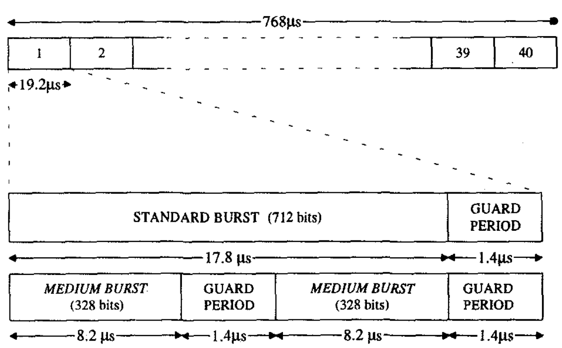

Fig. 1 - Several MBS specifications

The MBS/RACE project activities concerning to the characterization of the propagation channels have clearly shown that those channels strongly depend on cellular configurations and antenna issues; a certain directivity of BS and MS antennas, in both the horizontal and vertical planes, helps in avoiding too a high delay spread (for the MBS goals) in certain scenarios and can also allow reduced transmit power requirements and an increased frequency reuse. A very reliable ray-tracing tool, conveniently adapted to both indoor and outdoor environments, has been employed to obtain propagation data for performance evaluation of MBS radio transmission, by Monte-Carlo simulation, including the impact of antenna/cellular issues. Simulation results obtained so far show that the current specifications of the physical channels are reasonable from the radiotransmission point of view: in fact, they allow simple adaptive equalisation/diversity schemes [2] to perform quite well, certainly with the help of MS antennas (e.g. switchable-beam schemes) capable of exhibiting some directivity in both azimuth and elevation.

Any attempt for designing a suitable error control coding scheme for MBS should take into account the specific type of "time-varying channel" which is inherent to the radiotransmission conditions throughout the system. We must keep in mind the mandatory use of some kind of space diversity (at least implemented as a two-branch, RSSI-driven switched micro-diversity) [2]: space diversity techniques are very useful for reducing both multipath fading and shadowing effects, which can strongly affect the link quality (sometimes rather poor and lasting for several frames at unfavourable levels).

III. ADAPTIVE ERROR CONTROL SCHEME

A. Error Control Schemes for Time-varying Channels

It is well known that conventional ARQ (Automatic Repeat-request) schemes can provide a reliable error control, even with a low overhead of check digits, provided that the channel quality is high or moderate [5]; however, under poor channel conditions, the number of repetitions becomes excessive. Conventional FEC (Forward Error Correction) schemes can be very suitable for predictable, quasi-time-invariant channels, but provide unnecessary correction power for much of the time if the channel is time-varying, and, even with a heavy overhead of check digits, are not able to ensure a reliable transmission whenever the channel is in the "bad state"; with time-varying channels, a deep interleaving scheme is often employed so as to spread out the error bursts and get an approximately uniform error distribution at the decoder input, thus allowing satisfactory performances through the use of a FEC scheme (e.g. the well-known GSM case); a well-known, powerful solution for FEC applications is the concatenation of convolutional inner codes (with soft decoding) and Reed-Solomon outer codes. Both a high number of repetitions (ARQ) and a high interleaving depth (FEC) can imply an unacceptably long decoding delay.

Hybrid error control schemes, combining ARQ and FEC features, can offer good trade-offs between throughput, delay and transmission reliability. This is the case with the so-called "Type II hybrid system" [5], which uses a rate 1/2 invertible error-correcting code and requires transmission of check digits for error correction only when they are needed; "information" and "parity" sequences can be either detected to be "correct" or combined for FEC decoding. Other interesting possibility is to send successive check digits to build up a code which can be finally powerful enough for a successful decoding, never throwing away (differently from the "Type II" hybrid scheme) the already transmitted digits [6, 7]. Such "hybrid ARQ/FEC" schemes can be regarded as *adaptive* error control schemes, and are especially interesting for applications where the channel is time-varying; this is due to their ability to "match" the coding redundancy to the channel state.

B. Adaptive Error Control Scheme for MBS

The chosen error control scheme is based on the concatenation of a convolutional inner code with rate 3/4 and an outer code, constructed over the Galois field GF (2^6), which is a (60, 36) shortened Reed-Solomon (RS) code. It is assumed here that the information contents of a standard traffic burst consists of 54 octets, i.e., the user information within an ATM cell plus an "expanded header" of 6 octets, which includes some radio specific signalling. The standard burst exhibits the corresponding coded information (plus training and tails), obtained as follows: two (60, 36) RS codewords, sharing the information contents reported above, are punctured so as to achieve a pair of (42, 36) RS codewords with reduced redundancy; this pair is then submitted to the above-mentioned inner encoding, thus leading to the overall data contents of the standard burst (672 coded bits). The medium-size burst - *to be sent only when required* - contains the RS redundancy not transmitted in the standard burst; both blocks of firstly deleted coded symbols are submitted to the inner convolutional encoding so as to obtain the data contents of the medium-size burst (288 bits, as reported in sec. II). When building either the standard burst or the medium-size burst, 3 tail bits are added to the bits provided by the outer encoder before being submitted to the inner encoding, due to the "blockwise" inner decoding requirements. In this paper, we assume that the inner code is a 16-state convolutional code having free distance equal to 4, which results from a rate-1/2 puncturable convolutional code [8].

Other characteristics of the proposed error control scheme are as follows:

- It does not employ *interburst* interleaving (but *intraburst* interleaving is recommendable).
- A soft inner decoding is intended, with a modified Viterbi algorithm which is able to provide reliability information along with the decoded output; this information can be used to erase *unreliable* bits [9].
- Error-and-erasure correction by the outer decoder is not performed beyond the "packing radius" (i.e., *uncorrectable* RS blocks are identified).

A "threshold dependent" error control can be implemented with the proposed scheme. Appropriate thresholds can be established in a flexible way, taking into account service characteristics and requirements (and, possibly, channel state and traffic conditions). Two types of thresholds have to be selected:

- One "inner" threshold controls the production of erasures by the inner decoder, following ideas of [9].
- Two "outer" thresholds are concerned to requests for additional transmissions; either the transmission of the "complementary" medium-size burst or a retransmission of the standard burst can be requested (as well as a retransmission of the medium-size burst), in accordance with the selected "outer" thresholds.

It must be noted that additional transmissions are also requested when uncorrectable RS blocks are found, since a bounded-distance, hard decoding is employed for the outer code.

The inner decoding employs a modified version of the Viterbi algorithm, as proposed in [9]. It provides reliability information on the maximum-likelihood decoded sequence, by erasing bits which are "probably in error", and is very suitable for high-speed implementations, an important issue in MBS. During the decoding procedure, the "inner threshold" reported above (say, T) is just compared to the difference between the "best" and the "second best" path metrics, for each state; when that difference falls below T, all the corresponding bits which are different in the two paths are erased (*they can be regarded as unreliable bits*) and decoding continues. Therefore, the output of the inner decoder consists of both decoded bits and, as a side information, a set of erasures. 6-bit RS symbols containing any erased bits are regarded as RS erasures by the outer decoder. The number of erased RS symbols per (42, 36) block is compared to both an "upper" and a "lower" outer threshold (say, Y and X, respectively, with Y>X), appropriately selected. The number of RS erasures gives an important information about the "channel state"; by comparing that number with X and Y, we can implicitly classify the channel as "good", "mildly poor" or "very bad", with regards to the required quality of service. Certainly, a realistic, compromise choice has to be made for the inner threshold: if T is too high, a lot of correct bits will likely be replaced by erasures (i.e., a high "false-alarm" probability arises) which means a very pessimistic evaluation of the link quality; if T is too low, an optimistic evaluation of the link quality is unavoidable and too a high percentage of erroneous bits will not be replaced by erasures (i.e., the probability of undetected error events can become unacceptably high).

The proposed ARQ/FEC protocol performs the following steps with regards to the transmission of the information contents of a standard burst:

1. Transmit the standard burst (672 bits of coded data plus training and tails).
2. Use the modified Viterbi algorithm, with threshold T, to perform soft inner decoding and give reliability information to the outer decoder.
3. Let ρ_1 and ρ_2 denote the number of 6-bit erasures in the "first" and the "second" (42, 36) blocks at the input to the RS decoder:

    3.1. If $\rho_1 > Y$ *or* $\rho_2 > Y$, (i.e., the channel state is found to be "very bad") discard the received burst and let the transmitter execute step 1, etc.

    3.2. If $\rho_1 \leq X$ *and* $\rho_2 \leq X$, try to perform RS decoding (hard, bounded distance); if there is a successful decoding for both (42, 36) blocks, send the decoded bits to the data sink, as well as a *positive acknowledgement* (ACK) to the transmitter, and otherwise send a *negative acknowledgement* (NAK), so as to request the execution of step 4, etc.

3.3. If the pair (ρ_1, ρ_2) does not match conditions 3.1 or 3.2, send a NAK to the transmitter, so as to request the execution of step 4, etc.

4. Transmit the complementary medium-size burst, i.e. 288 bits of coded data, plus training and tails (Since the channel state has been found to be "mildly poor", the received burst can be useful, with some help from the complementary redundancy.).

5. Use the Viterbi algorithm, with the same threshold T, to perform soft inner decoding and give reliability information to the outer decoder.

6. Try to perform RS decoding; if there is a successful decoding for both (60, 36) blocks, send the decoded bits to the data sink, as well as an ACK to the transmitter; otherwise, send a NAK, so as to request the execution of step 4, etc., if it is the first decoding failure with the medium-size burst, or let the transmitter execute step 1, etc. if this is not the case.

The proposed error control scheme is an *adaptive* one, able to combine error detection and error correction features under an implicit evaluation of the channel state, either discarding or taking advantage of already received data which cannot ensure a reliable decoding. It assumes selective additional transmissions and clearly exhibits an ability to "match" the coding redundancy to both the channel state and the required service quality, through the appropriate selection of the inner threshold (T) and the pair of outer thresholds (X, Y); moreover, it should be noted that these thresholds can be either fixed or variable during transmission, according to channel/service requirements. It is expected that a good compatibility with other air interface issues (namely efficient medium access and power control schemes) can be achieved.

IV. PERFORMANCE EVALUATION BY SIMULATION

A. Simulation Assumptions

The simulation scenario adopted here for performance evaluation of the proposed error control scheme has already been considered [2] for performance evaluation of the equalization/diversity schemes reported in [10], which require a single decision-feedback equalizer (DFE). Under the simulation assumptions now adopted with that (outdoor) scenario, the time-varying nature of the transmission channel leads to a strongly variable number of errors in the DFE decisions: (see Fig. 2, where the corresponding values of average "channel BER" are, respectively, 10^{-2} (A), $2.5\text{x}10^{-3}$ (B) and $7\text{x}10^{-4}$ (C)): many bursts have just a few errors or no errors at all; only for a small percentage of the simulated bursts the number of erroneous decisions is found to be high. However, the error distributions are quite dependent on $\overline{E}_b / N_0$, and it is clear that, for low $\overline{E}_b / N_0$ values, the percentages of "mildly poor" and "very bad" bursts can become very significant.

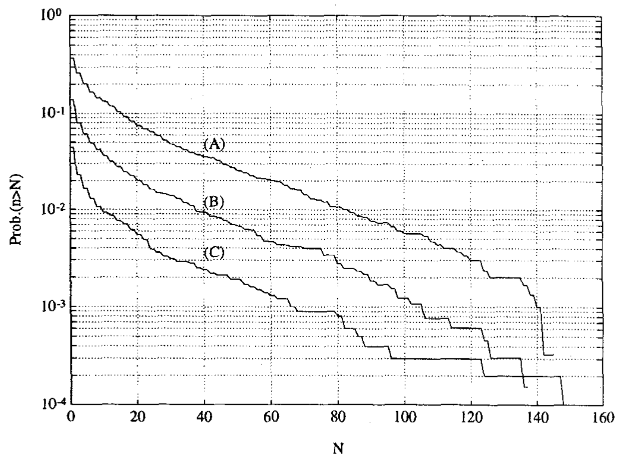

Fig. 2 - Example of error distributions (n:number of DFE errors per standard burst): $\overline{E}_b / N_0$=7 dB (A), $\overline{E}_b / N_0$=10 dB (B) and $\overline{E}_b / N_0$=13 dB (C)

In order to reduce the computation time, we employed a simulation approach similar to that presented in [11], very useful for obtaining accurate estimates of residual bit error rates below 10^{-6}. Unquantized samples at the decision device of the DFE (binary decisions: ±1) were employed for inner decoding, through the above-mentioned modified Viterbi algorithm; the decoding operations were only performed for the bursts with DFE errors (a perfect decoding, without output errors and erasures, was assumed for the "error-free" bursts). Output decoding was performed using the low-complexity decoding algorithm proposed in [12]. Intraburst interleaving was employed, at the bit level, so as to get a more uniform distribution of the DFE errors during each burst interval. We have always considered the same channel state for first transmissions of complementary medium-size bursts, so as to check out the ability of the proposed scheme for dealing with "long" periods (say, more than ten frames, i.e. 7.68ms) of "mildly poor" transmission conditions, *while keeping a low decoding delay*. The ACK and NAK short messages were supposed to be not affected by decoding errors (e.g. thanks to a very powerful, specific error protection).

A set of simulations has been carried out for the three values of $\overline{E}_b / N_0$ already considered (see Fig. 2). The simulated BER results are depicted in Fig. 3, for several combinations of thresholds (the average "channel BER" is also shown, for the sake of comparison). The selected thresholds were the following:

- T = 1.6 (dashed line) and 2.1 (solid line);
- (X, Y)=(4,10) (circle), (2,8) (triangle) and (0,6) (square).

In Fig. 4 we show the corresponding results which pertain to the "throughput efficiency", defined as the ratio between the 'Minimum number of standard slots' for transmitting a certain amount of traffic information (assuming no additional transmissions) and the 'Average number of effectively required standard slots' (due to the necessary additional transmissions).

Note that two medium-size slots are equivalent to one standard slot (see Fig. 1).

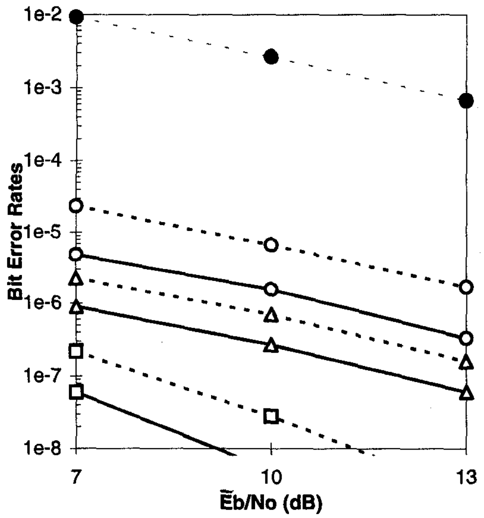

Fig. 3 - Residual BER for combinations of the chosen thresholds

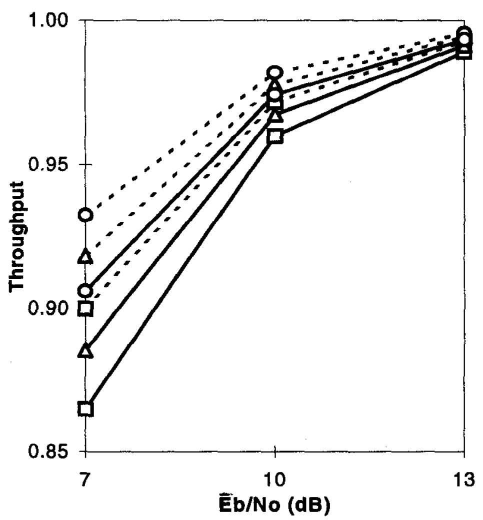

Fig. 4 - Throughput efficiencies for combinations of the chosen thresholds

By comparing Figs. 3 and 4, we can conclude that good trade offs exist between transmission reliability and throughput efficiency, even for the assumed "worst-case" conditions. Though the appropriate selection of both inner and outer thresholds, it is possible to achieve a low residual BER together with an excellent throughput efficiency (compatible with a very low decoding delay) or, if required, an extremely low residual BER together with an acceptable throughput efficiency. The proposed scheme is powerful enough to cope with poor channel conditions and is compatible with both ATM characteristics and those of a wide range of services (crucial issues in MBS).

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