Generalised stopping criterion for iterative decoders

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A generalised stopping criterion for iterative decoders is proposed. It is designed for detecting early convergence and non-convergence and it allows for a saving of 60 to 75% in computational complexity. The UMTS high speed evolution system (HSDPA) is chosen for simulations.

Introduction: The invention of turbo codes [1] is one of the most valuable contributions to the forward error correction community. However, the complexity of turbo decoders may be prohibitive for small devices such as mobile phones. A solution for reducing the decoding computational complexity, power consumption and for increasing battery life is to dynamically control the number of iterations. For doing so, many criteria have been proposed for detecting blocks correctly decoded before the maximum number of iterations allowed in receivers is reached. In [2] the criterion is based on the cross-entropy (CE) between the distributions of the estimates at the outputs of the decoders at each iteration and in [3] two simple criteria are proposed, the sing-change-ratio (SCR) and the hard-decision-aided (HDA). In [4] and [5] iterations are stopped when estimations of BER and FER, respectively, correspond to targeted performances while in [6] they are stopped when the variance of extrinsic values between consecutive iterations exceeds a predefined threshold. Estimates of signal-to-noise ratio (SNR) values and of the second moment of the conditional log likelihood ratios (LLR) of the emitted bits given the observation of received bits are used in [7] and [8], respectively.

While attention has been paid largely to the detection of early convergence of iterative decoders, stopping iterations upon detection of non-convergence of the iterative process has been rarely addressed. In [9] detections of early convergence and non-convergence, based on estimations of the mean absolute values and the mean number of sign changes of LLR values, are proposed.

A generalised criterion, derived from HDA [3], is proposed in this Letter. It allows for stopping iterations when blocks are detected as being either correctly decoded or undecodable before the maximum number of iterations allowed in receivers is reached.

Generalised criterion: The HDA criterion [3] is characterised by a very low computational complexity. It performs better than CE [2] and SCR [3] for small to medium SNRs and small to medium interleaver sizes, which makes HDA well suited for the high speed downlink packet access (HSDPA) system, since a BLER of 10% is targeted for initial transmissions (erroneous blocks being corrected by the physical layer hybrid ARQ functionality).

The proposed generalised criterion, derived from HDA, consists in calculating differences of tentative hard decisions of information bits obtained from LLR values at iterations i and $i-1$:

$$\Delta = \frac{1}{N}\sum_{k=0}^{N-1}[\hat{u}_k(L_2^i) - \hat{u}_k(L_2^{i-1})] \qquad (1)$$

In (1), $\hat{u}_k(L_2^i)$ and $\hat{u}_k(L_2^{i-1})$ are tentative hard decisions of information bits made from LLR values after the second decoder, L_2^i and L_2^{i-1}, with k the bit index in the decoded sequence and i and $i-1$ iteration indices. This metric is then normalised by the block length N. The decoder structure is shown in Fig. 1, with extrinsic values Le_1 and Le_2, LLR values L_1 and L_2 and y_s, y_{1p} and y_{2p}, respectively the received systematic, first parity check and second parity check bits streams.

After computing this metric, iterations are stopped in two cases, either when Δ is smaller than the convergence threshold η_{conv} (the block is considered as being correctly decoded) or greater than the non-convergence threshold $\eta_{non\text{-}conv}$ (the block is considered as being definitely lost). Otherwise, the iterative decoding process continues.

HSDPA system and simulation assumptions: HSDPA has been standardised within the Release 5 framework of 3GPP. It provides high data rates (up to about 12.8 Mbit/s) on the high speed downlink shared channel owing to adaptive modulation and coding (with QPSK and 16QAM modulation schemes), a possible massive allocation of spreading codes (up to 15 codes at spreading factor 16 for a single user), link adaptation with a quasi-infinite granularity (closed loop based on users' requests), fast hybrid ARQ (including HARQ of types 1, 2 and 3) and fast scheduling performed by the MAC-hs entity (deported in the base station).

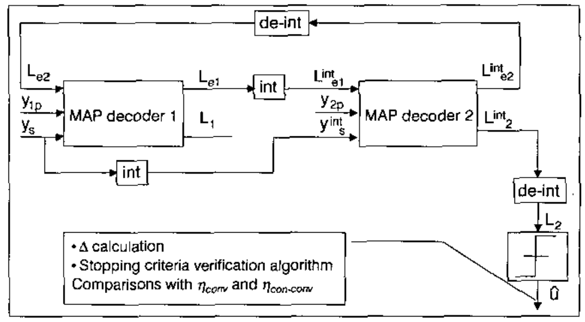

Fig. 1 *Turbo decoder structure*

In the simulations, the transmitter chain is composed of the Release 5 UMTS [10] rate 1/3 $(15, 13)_8$ PCCC turbo encoder, the double stage rate matching, and the parallel bit level channel interleaver. The receiver comprises inverse functions, and a bit level Max MAP turbo decoder performing a maximum of eight iterations. In simulations, the hybrid ARQ and link adaptation functionalities are disabled. The modulation scheme is 16QAM, using the default unrotated 16QAM constellation and conventional Gray mapping. Code blocks processed by the turbo encoder have a fixed size of 954 bits (938 information bits and 16 CRC bits). The overall coding and puncturing rate is 1/2, i.e. the rate matching punctures 1/2 of parity check bits (symetrically on both parity check bits streams generated by constituent encoders) in order to have 1920 bits sent through the air interface over a period of 2 ms.

Results: A single-path Rayleigh faded propagation channel at a vehicle speed of 50 km/h was used in simulations. The early convergence detector is either disabled (OFF) or the threshold η_{conv} is set to 2% and the early non-convergence detector is either disabled (OFF) or the threshold $\eta_{non\text{-}conv}$ is set to 20%. This gives four configurations {OFF, OFF}, {2%, OFF} corresponding to HDA [3], {OFF, 20%} and {2%, 20%}. In Fig. 2 block error rates (BLER) and bit error rates (BER) are plotted with dashed and plain curves, respectively, and averaged numbers of iterations are shown in Fig. 3.

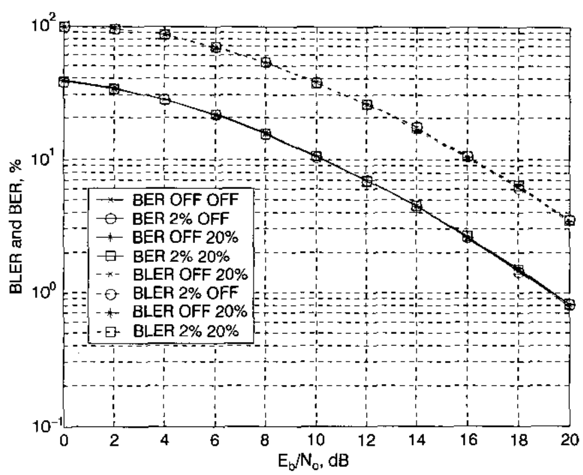

Fig. 2 *Performance on single-path Rayleigh channel*

When both detectors are enabled ($\eta_{conv}=2\%$ and $\eta_{\text{non-conv}}=20\%$), the averaged number of iterations is significantly reduced. It is grossly comprised between two and three (about 60 to 75% computational complexity reduction), without any performance losses, instead of eight iterations performed when both detectors are disabled. These averaged numbers of iterations can be slightly lowered and the trade-off is as usual the gain in computational complexity against the affordable performance degradation.

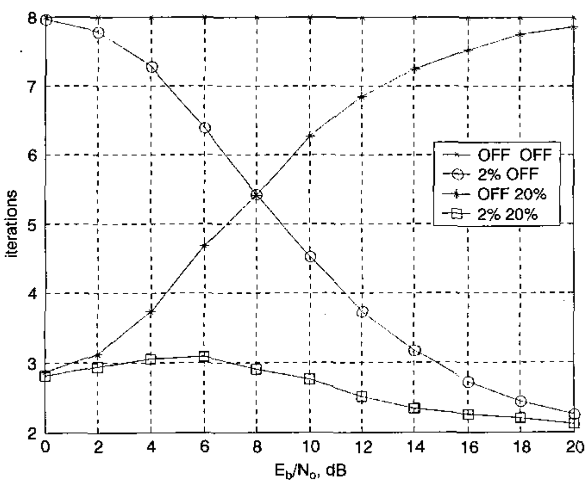

Fig. 3 *Average number of required iterations on single-path Rayleigh channel*

Conclusion: The proposed generalised stopping criterion allows for significant reduction in the number of iterations performed by turbo decoders. This corresponds to saving between 60 and 75% of the overall turbo decoding computational complexity, without any performance degradations.

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LDPC coded unitary space-time modulated OFDM system in broadband mobile channel

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A unitary space-time modulated orthogonal frequency division multiplexing (OFDM) system using low density parity check (LDPC) code for increasing the system performance is proposed. The proposed system can obtain space-time diversity using unitary space-time modulation without linear operation like space-time block code (STBC), and large coding gain using LDPC code.

Introduction: High data rate and high quality multimedia services are demanded in beyond third generation and a fourth generation (4G) mobile communication, since application services are increasing. To meet this demand, orthogonal frequency division multiplexing (OFDM) is attractive and has been widely studied in recent years. OFDM can achieve high frequency efficiency and a high data rate, since the signals are transmitted in parallel using many subcarriers that are mutually orthogonal. Moreover, since the frequency spacing of each subcarrier is minimum, OFDM can treat frequency-selective fading as a flat fading for each subcarrier. To achieve higher capacity and data rate for 4G, a space-time processing system has been proposed. Space-time coding (STC) has been studied as an effective transmit diversity technique to obtain high transmission performance. Among STC, thespace-time block code (STBC) becomes attractive because STBC has simplicity of implementation and decoding [1]. However, STBC requires the linear operation to separate and decode signals that are received simultaneously. Recently unitary space-time modulation (USTM) has been proposed to perform space-time diversity without linear operation. USTM constellations have been designed and shown to perform well for uncoded transmission. Ahn and Sasase [2] and Bahceci and Duman [3] proposed and investigated USTM using convolutional code and turbo code, respectively, for increasing the system performance. Low density parity check (LDPC) code has received much attention recently due to its simple decoding property [4]. LDPC code is a linear code, and is composed of huge size and a very sparse parity check matrix. LDPC code can achieve BER performance which is very close to the Shannon limit as well as turbo code when the LDPC code has sufficiently long code length. In this Letter, we propose an LDPC coded USTM/OFDM system. The proposed system can obtain space-time diversity using USTM without linear operation like STBC, and very large coding gain using LDPC code. By computer simulation, we show the BER performance of the proposed system can be improved compared with the conventional uncoded USTM/OFDM.

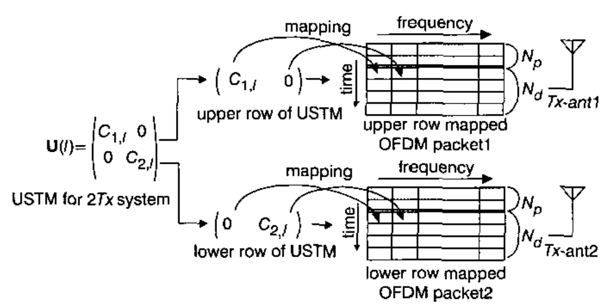

Fig. 1 *Concept of USTM/OFDM with two transmit antennas*

Proposed system: Fig. 1 shows the concept of the USTM/OFDM system with two transmit antennas. The USTM signal is a matrix, the rows of which are transmitted from two antenna elements and mutually orthogonal to each other. In the transmitter, the data stream is divided into bit sequences that consist of $R \cdot M$ bits, where R and M denote information bits per parallel symbol to be transmitted, and the number of transmit antennas, respectively. Each $R \cdot M$ bit sequence is mapped into the constellation $\mathbf{U}(l)$ $(0 \le l \le L-1)$ selected from $L = 2^{RM}$. The constellation of USTM can be written as

$$\mathbf{U}(l) = \begin{bmatrix} e^{j(2\pi l/L)} & \cdots & 0 \\ \vdots & \ddots & \vdots \\ 0 & \cdots & e^{j(2\pi l/L)} \end{bmatrix}, \quad (0 \le l \le L-1) \qquad (1)$$