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Exploiting variable-length padding bits for decoder performance improvement with its application to compressed video transmission[☆]

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ABSTRACT

In many practical communications systems, the number of bits transmitted by each frame is fixed, whereas the number of information bits to be transmitted can be variable. One example is a typical communication system that consists of a source encoder followed by a channel encoder, where the former generates a variable-length information sequence per source data and the latter converts a fixed-length information sequence into a fixed-length codeword. Therefore, dummy bits are often padded (or stuffed) to the information bits prior to channel encoding at the transmitter so as to fill the gap, and they are usually discarded after channel decoding at the receiver. This paper proposes a novel use of such padding bits (PBs) for improving the performance of channel codes. We first observe that by considering the PBs embedded in the information part of systematic block codes as known bits and expurgating them at the receiver, it is possible to enhance the performance of the channel decoders, provided that the PBs are judiciously rearranged in the information sequence and they are perfectly known at the receiver. Nevertheless, the length of the PBs is random by nature, and transmitting such side information per frame would reduce the throughput. Therefore, we also develop a novel detector that can estimate the PB length reliably without the help of explicit side information. Through computer simulations employing an actual turbo code, the effectiveness of the proposed receiver with the PB length detector is demonstrated in terms of frame error rate (FER) performance. Furthermore, as a practical example of the proposed technique, its application to typical video transmission systems is addressed, where the performance improvement in terms of peak signal-to-noise ratio (PSNR) is observed.

1. Introduction

The multimedia source encoders that perform video, image, and audio compression usually produce coded bits of variable length as a result of entropy coding. On the other hand, the data in the physical (PHY) layer is usually transmitted block-wise as a unit such as a *frame*, and the length of the frame is fixed *a priori*. Therefore, the length of the codeword after the channel coding should be subject to this frame length constraint. Since the rate of a channel code is usually determined *a priori*, the number of information bits to be encoded by this code is fixed as well. As a consequence, there should be a mismatch between the length of the source coded bits and that of the information bits to be encoded by a channel code. A practical and straightforward approach is to simply insert (or *stuff*) some dummy bits, which will be called *padding bits (PBs)* in what follows, such that the length of the source coded sequence matches that of the information bits processed by the fixed-rate channel encoder. This scenario can be applied not only to a simple source–channel coding system, but also to a general framework of computer networks. Classic Ethernet, for example, defines the minimum frame length in order to enable collision detection, and if the actual data frame is shorter than the minimum length required, dummy bits are appended to the data field, known as *byte stuffing* [1]. More recent wireless LAN standards such as IEEE 802.11n and 11ac introduce frame aggregation [2], where more than a single packet may be contained in one transmission frame. In such systems, each packet may have a few inherent null bytes to form a single transmission frame in the medium access control (MAC) layer. Furthermore, null data packets (NDPs) may be inserted when the number of packets generated by the upper layer is not sufficient to fill the entire frame. In this case, these NDPs can be also considered as a collection of PBs.

In the conventional system, since the PBs are simply inserted prior to the channel encoder at the transmitter and then automatically removed

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after the channel decoder at the receiver, it neither improves throughput nor enhances reliability, thus resulting in a reduction in terms of bandwidth efficiency or effective throughput. A straightforward approach that exploits these redundant PBs from the viewpoint of improving reliability is to replace the redundant null bits in the data frame with the parity bits of the systematic codes [3]. However, such an approach not only lacks flexibility in code design due to the fact that the number of null bits is inherently a random variable, but also makes the soft decision decoding challenging even if such information is available from the output of the symbol detector. In this work, we take an alternative approach inspired by *code doping* [4] for the iterative decoding of turbo code; Code doping is a technique to improve the performance of iterative decoder and accomplished by inserting the pilot bits to the original information bits or by replacing the output of the outer coded bits with the pilot bits, provided that their positions and values are known at the receiver side. In [5], code doping is employed for removing the error floor of bit-interleaved coded modulation system with iterative detection (BICM-ID). The *trellis pruning* is also an equivalent technique, and such an approach is recently studied in the framework of cooperative communications [6]. The study in [7] applies trellis pruning to a turbo code in order to simplify the decoding process and reduce the computational complexity. Trellis pruning is applicable not only to the block code but also to the convolutional code [8]. As related work, the unequal error protection (UEP) using *path-pruned* convolutional code is proposed in [9] and [10], where the latter study combines the use of rate-compatible punctured convolutional codes [11].

Even though PBs almost always exist in practical fixed-length systems, they are often unexploited or simply ignored. This paper focuses on such a practical scenario where there is a mismatch between the length of source coded bits and that of information bits to be processed by the subsequent channel encoder. Specifically, this paper proposes an approach that makes effective use of the redundant PBs for performance improvement of the subsequent channel decoder, assuming that the underlying error correcting codes have a systematic block coding structure such as turbo codes. In order to make the detection of PB event feasible at the receiver without specific side information, we assume that some *marker bits* are inserted at the boundary between the source coded bits and PBs. This approach itself shares some similarity with the problem of the *frame synchronization* in coded systems (e.g., [12–14]). In our work, however, the constraint on synchronization can be somewhat relaxed; we allow detection errors of PB length to the extent where the critical degradation at the receiver can be avoided. The proposed detector then first estimates the PB length blindly by locating the marker bits in the received signal. The information of the estimated PB length is utilized for trellis pruning of the turbo decoder, where the maximum *a posteriori* (MAP) decoding based on the BCJR algorithm is incorporated.

The *joint source–channel coding (JSCC)* is a well-known class of studies that utilize the redundant information to improve the quality of received signals. In [15], an approach is developed that exploits the redundancy caused by a source encoding process for channel coding so as to improve the quality of the received image. The study in [16] proposes an iterative decoding of variable-length codes adopted in MPEG-4 Visual where substantial error protection capability is demonstrated. The joint source–channel decoding for H.264/MPEG-4 advanced video coding (AVC) is also proposed in [17] and [18]. The application of UEP according to the priority of H.264/MPEG-4 AVC video layer is proposed in [19] and [20]. In [21], an adaptive optimization of rates for source and channel codes according to the channel state information (CSI) is proposed for motion JPEG 2000 video transmission. The studies in [22] and [23] consider fixed size packet transmission where the adaptation of information bit length is achieved by utilizing additional channel codes. Although these JSCC-based approaches are effective under a given bandwidth constraint, the applicability is limited to their specific source coding scenarios. Therefore, they may not be necessarily applicable to a general concatenated system of source and channel coding. Moreover, the side information on the rate of channel codes employed in each transmission must be shared by the transmitter and receiver in advance. To the contrary, the proposed technique is applicable to more general source–channel coding systems and the decoder is adjustable to the variety of redundant PB lengths without side information.

Our main contributions in this paper are summarized as follows:

1. Assuming that the PB length is a random variable and the marker bits are inserted at the end of the source coded bits by the transmitter, we develop its detection algorithm at the receiver side without explicit side information. This can be performed prior to turbo decoding by utilizing the systematic nature of turbo codes.
2. The performance of the proposed PB length detector is theoretically analyzed and based on this result, the length of the marker bits is optimized.
3. Through computer simulations, we demonstrate that the proposed system leads to an enhancement in terms of frame error rate (FER) performance.
4. By judiciously designing the frame structure according to the importance of source bits in a practical video transmission system, a quality improvement in terms of peak signal-to-noise ratio (PSNR) is demonstrated.

We note that the method developed in this work aims not at improving decoding performance in high SNR region (i.e., error floor region) of turbo codes but rather at enhancing the reliability in terms of FER performance in the waterfall region. Consequently, it may reduce the probability of retransmission and thus is expected to also improve the throughput if it is operated with automatic repeat request (ARQ).

This paper is organized as follows. In Section 2, we give a brief description of the transmitter and receiver models considered in this work. Section 3 describes the proposed padding approaches at the transmitter. The trellis pruning technique is also introduced in order to utilize the known positions of PBs at the decoder. Section 4 describes a method to detect the PB length in a received frame as well as the theoretical analysis of its detection error rate. The derived theoretical and simulation results are compared in Section 5, where the improvement of FER performance by the proposed PB-exploiting system over a conventional turbo coding system is also elucidated. In Section 6, as a practical example of the proposed technique, its application to compressed video packet transmission systems is addressed, which reveals a noticeable improvement in terms of video quality. Finally, Section 7 concludes this work.

Throughout this paper, we employ BPSK as our modulation for simplicity of analysis, but our concept is also applicable to more general systems operating with, e.g., high-order QAM, possibly with bit-interleaved coded modulation (BICM) [24].

2. System overview

In this section, we develop our encoder model that has the PB insertion event followed by the description of our proposed channel decoder.

2.1. PB-Aware transmitter model

Fig. 1(a) shows our PB-aware system model at the transmitter side. Suppose that the source encoder generates a binary data sequence of length K', denoted by $\mathbf{s} = (s_0, s_1, \ldots, s_{K'-1}) \in \{0,1\}^{K'}$. Let $\mathbf{u} = (u_0, u_1, \ldots, u_{K-1}) \in \{0,1\}^K$ denote a binary information sequence of constant length K, to be processed by the channel encoder of rate $R = K/N$ in the PHY layer, where N represents the frame length. As is often the case with practical systems, we assume that the length of the binary data sequence received from the upper layer varies randomly for each frame. Therefore, there is a gap between K' and K. Without loss of generality, we assume that $K' \leq K$. In order to meet the constraint on the frame length of N with a fixed code rate R, the PBs of length