Robust image transmission over packet loss channels

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2006

Li, X. (2006). Robust image transmission over packet loss channels. Master's thesis, Nanyang Technological University, Singapore.

<https://hdl.handle.net/10356/47484>

<https://doi.org/10.32657/10356/47484>

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Robust Image Transmission Over Packet Loss Channels

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A thesis submitted to the Nanyang Technological University
in fulfilment of the requirement for the degree of
Master of Engineering

2006

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Abbreviation

ACK	Acknowledgement
ARQ	Automatic Repeat reQuest
AUCP	Adaptive Unequal Channel Protection
CRC	Cyclic Redundancy Checking
DCT	Discrete Cosine Transform
DWT	Discrete Wavelet Transform
EEP/ELP	Equal Error/Loss Protection
FEC	Forward Error Correction
GEC	Gilbert-Elliott Channel
HLP	Hybrid Loss Protection
IL-ULP	Improved Layered Unequal Loss Protection
JPEG	Joint Photographic Experts Group
JSCC	Joint Source-Channel Coding
L-ULP	Layered Unequal Loss Protection
MPEG	Moving Picture Experts Group
PMF	Probability Mass Function
PSNR	Peak Signal-to-Noise Ratio
QoS	Quality of Service
R-D	Rate-Distortion
ROI	Region of Interest
RS	Reed-Solomon
SPIHT	Set Partitioning in Hierarchical Trees
TCP	Transmission Control Protocol
UEP/ULP	Unequal Error/Loss Protection

Abstract

Wavelet-based image codecs, such as SPIHT and JPEG2000, have shown to be superior to DCT-based image codecs in not only the coding efficiency but also the functionalities such as the progressive property, which allows the bit stream to be truncated in any position and thus provides rate scalability. Although the progressive image coding can provide excellent compression and scalability performance, it makes image bit streams very sensitive to channel noise. Therefore, how to robustly transmit progressive image over lossy channels becomes an important issue and has received much attention in the past a few years.

In this research, our objective is to study, design, and implement efficient FEC-based joint source-channel coding schemes for image transmission over packet loss channels. In particular, by noticing that most existing unequal loss protection (ULP) schemes do not consider the minimum quality requirement and usually have high computation complexity, we propose a layered ULP (L-ULP) scheme to solve the problems. By smartly choosing the number of layers, the proposed L-ULP is able to achieve the low complexity and minimum quality requirements with slight performance degradation.

We then extend the L-ULP scheme to the cases when the bit rate is high or the number of packets is large, i.e., cases with long decoding delay. In order to deal with this delay problem, more than one packet group are adopted in the proposed scheme. The FEC encoding and decoding processes are then applied to each individual packet group instead of all the packets.

By integrating L-ULP with the pre-interleaving, we also propose an improved L-ULP (IL-ULP) scheme to enhance the performance of the L-ULP. Pre-interleaving is able to delay the occurrence of the first unrecoverable loss in the progressive bit stream. Experimental results show that the proposed IL-ULP is able to achieve better distortion

performance than that of the hybrid loss protection (HLP) scheme while the complexity is as low as that of L-ULP.

Moreover, we apply IL-ULP to the transmission of JPEG2000 encoded images. We evaluate the contributions of lost channel packets to the overall end-to-end distortion, which is used to calculate the optimal FEC allocation. We then compare the performances between L-ULP and IL-ULP, and the results demonstrate that, for JPEG2000 codestreams, IL-ULP can still achieve superior performance to that of L-ULP.

Acknowledgments

I would like to express my sincere gratitude to my supervisor, Dr. Cai Jianfei, for his intellectual guidance, constant encouragement and persistent patience throughout my Master study. Without his suggestions and support, this thesis cannot reach its current stage.

I am also grateful to Mr. Ni Zefeng and Wu Jianhua, for their stimulating discussions and valuable assistance. Thanks also go to other staffs and graduate students in Center for Multimedia and Network (CeMNet), particularly Dr. Gao Deyun, Yang Wenxian, and Zhan Wenning. I benefit so much from their friendly suggestion, discussion and cooperation.

Special thanks are for my wife, my parents, and my siblings. Thank all of them for their priceless love.

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Chapter 1

Introduction

1.1 Research Background

Visual communications are becoming increasingly important in our daily life [1]. During the past decade, we have witnessed significant progress in transmitting visual information, in the form of digital images and videos, over packet-switched networks such as Internet. However, existing packet-switched networks usually have no quality of service (QoS) guarantee and typically provide the best effort services. When the number of packets sent exceeds the channel capacity, packets will be discarded randomly, causing loss of data and even decoding failure if the lost data are not retransmitted. Therefore, to investigate robust image transmission over packet-switched networks is an urgent and challenging task.

1.1.1 Characteristics of Packet Visual Communications

There are two ways to send data through a network: packet switching and circuit switching. In circuit switching, a fixed connection is established on demand for a call exclusively and all messages follow this route until it is released. However, in packet-switched networks, a message is broken into packets, each of which is sent to the destination separately through maybe different routes. Thus, if one path is busy or fails, other paths can be launched without interrupting the whole transmission. A packet-switched channel can

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multiplex different signals, such as video, images and voice, together and thus the channel capacity is utilized more effectively than that of the channel based on circuit switching. In addition, by using dynamic bandwidth allocation, packet-switched channels can conduct variable bit rate (VBR) transmission directly, which matches the traffic pattern of visual information well. Therefore, packet switching is the dominant approach for multimedia delivery [2]. All the new wireless services, such as general packet radio service (GPRS), bluetooth, and 3G, use packet switching.

However, a fundamental and inevitable problem of packet-switched networks is that it is hard to guarantee the timely transfer of a packet [2]. In general, the delay of a packet comes from four sources. The first one is the nodal processing delay, which checks bit errors and determines the output link. The second one is the packet queuing delay. This delay depends on the congestion degree of the routers and can be pretty trivial in a high speed network. The third one is the transmission delay, which is the time to send bits through the communication channel. It is computed by L/R , where L is the packet length and R is the channel bandwidth. The last one is the propagation delay, which is calculated as D/s with D the end-to-end distance of the physical link and s the propagation speed in the medium. The overall packet delay can be decreased by appropriately selecting the packet size and packet switching strategies.

Another major problem of packet-switched networks is the packet loss due to either traffic congestion or physical layer channel impairment such as wireless channel fading. In fact, the packet delay problem is eventually a problem of packet loss for delay constraint multimedia communications since a packet arriving at the destination beyond the tolerable delay is useless. The packet loss problem may result in severe damage to visual information. This is because digital visual information, which is of huge data volume, is typically compressed by highly efficient compression schemes. The compressed bit stream is very sensitive to the channel impairment. A single error in a bit stream may cause all

the following bits become useless. In video coding, the effects of channel impairment may extend both spatially and temporally due to the use of predictive coding. Therefore, the packet loss remains as one of the most challenging problems in the applications of visual communications.

1.1.2 Error Control Techniques for Packet Visual Communications

In order to combat the packet loss, many error control schemes [3, 4, 5, 6, 7] have been proposed. In general, they can be divided into three categories: network error control techniques, error resilient techniques, and joint source-channel coding (JSCC). Network error control techniques refer to those approaches applied on the network side to provide the reliable communications while they do not take the characteristics of source coding into consideration. Error resilient techniques refer to those error-combating techniques performing at source coding. JSCC is the one that jointly considers source coding efficiency and transportation reliability. For each group, there are quite a number of sophisticated methods, and thus we will concentrate on some well-known approaches that have been broadly applied.

1.1.2.1 Network Error Control Techniques

Forward error correction (FEC) [8] and automatic repeat request (ARQ) [5] are two main network error control techniques.

FEC can achieve good performance in both error detection and error correction. However, FEC results in increasing transmission overhead, or reducing the available bandwidth for the payload data even when the channel is in good conditions. Reed-Solomon (RS) code [9], rate-compatible punctured convolutional (RCPC) code [10] and turbo product code [11, 12] are the popular error correcting codes.

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Figure 1.1: Structure of a RS codeword.

Figure 1.1 shows the structure of a RS codeword $C(N, k)$, where N is the block length and k is the number of source symbols. For a systematic code, the first k encoded symbols are source data and the remaining $N - k$ symbols are redundancy, which are simply appended to the source symbols. The error correction capability of a $C(N, k)$ code is mainly determined by the minimum Hamming distance d_{min} . The Hamming distance between two code words is defined as the number of symbols that disagree. In the case of erasure channels such as packet loss channels, because the locations of the erased symbols are known, an (N, k) RS code can recover any pattern of symbol erasures as long as the number of erased symbols is not larger than $N - k$. It is obvious that the longer the block length, the better the performance of a RS code. Increasing the block length, however, will introduce additional delay and increase the implementation complexity. In practice, a symbol of RS codes usually contains 8 bits or less, thus the maximum block length N_{max} does not exceed 255 bytes [13].

ARQ is another classic error control technique. ARQ needs a feedback channel to send retransmission requests while FEC does not have such a requirement. There are three types of basic ARQ schemes: Stop-and-Wait ARQ, Go-Back-N ARQ and Selective-Repeat ARQ [14]. In Stop-and-Wait ARQ, the sender transmits a packet and then waits for an acknowledgement (ACK). Once the ACK has been received, the next packet is transmitted. Stop-and-Wait ARQ is the simplest reliability technique and is adequate only for a very simple communications protocol. In GO-Back-N ARQ, the transmitter sends packets continuously without waiting for ACKs. If it receives a negative acknowl-

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edgement (NAK), the transmitter will retransmit the indicated erroneous packet and all following packets. In Selective-Repeat ARQ, the sender transmits packets continuously and the receiver sends ACKs back for each successfully received packet. A packet is retransmitted if its ACK is not received before the expiration of the timeout. Selective repeat provides the highest throughput and is employed by the well-known transmission control protocol (TCP).

Of these two error control techniques, ARQ has been shown to be more effective but with longer delay due to retransmission. Thus, ARQ may not be suitable for delay-constrained multimedia applications. As a consequence, hybrid ARQ is proposed which combines ARQ with FEC codes [6, 15, 16] to make a good compromise.

In practical packet-switched networks such as Internet, the error control techniques are often applied in different network layers. In particular, FEC and ARQ are typically used in the link layer to combat the physical channel impairment while in the transportation layer ARQ is often used to handle the packet loss due to network congestion. The TCP transport protocol is such an example, which uses ARQ with cyclic redundancy checking (CRC) codes to ensure all the packets are received correctly at the destination.

The main problem of these network error control techniques is that they do not consider the characteristics of source coding. They simply treat visual information as data and try to provide error-free transmission for each part of the bit stream. In fact, visual communications may not need error-free transmission since certain types of degradation may be imperceptible to the human visual system (HVS). Thus, direct application of traditional network error control techniques results in the waste of network resource. Moreover, the error-free transmission is usually obtained at the cost of network bandwidth or delay which will also greatly influence the performance of visual communications.

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1.1.2.2 Error Resilient Techniques

Existing error resilient techniques can be divided into two categories: error resilient encoding techniques and error resilient decoding techniques. The error resilient encoding techniques are those techniques performing on the encoder end while the error resilient decoding techniques are those performing on the decoder end.

A class of the error resilient encoding techniques is to limit error propagation. Error propagation in an image is usually caused by variable length coding, i.e., a small portion of errors may propagate to a large range of the bit stream. A common approach is to use resynchronization codewords. A resynchronization codeword can be added after a certain number of image blocks or after a certain number of bits, and there is no relationship among the data starting with different resynchronization codewords. In this way, the errors will be localized and prevented from impacting the entire bit stream. Due to its simplicity, such an approach has been widely adopted in image and video coding standards such as JPEG2000 [17] and H.263 [18].

Another class is to code visual information into multiple layers, termed as layered coding [7]. In layered coding, visual information is first classified into layers according to their importance, and then different layers are compressed by different coding methods. When network congestion occurs, mainly the enhancement layers with lower priority are dropped. In this way, the layered coding technique prevents the important information from dropping.

Error concealment is one of the most important error resilient decoding techniques. The error concealment techniques are usually associated with error detection techniques, such as CRC, data quantity check and syntax check. This is because error concealment techniques cannot be applied until errors are detected. Most error concealment techniques take advantage of two facts. One is that images of natural scenes have predominantly low-frequency components, i.e., there exists high redundancy in spatial and temporal

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domain. The other fact is the human eyes can tolerate more distortion to high-frequency components than to low-frequency components [3]. There are two types of error concealment for image coding: spatial concealment and frequency concealment. In spatial concealment, the missing pixel value can be recovered by interpolating the neighbor pixel values. In frequency concealment, given that we use discrete cosine transform (DCT), the low-frequency DCT coefficients in a block can be estimated by corresponding DCT coefficients or DC coefficients in neighbor blocks [19]. Although error resilient techniques work fine under low packet loss cases, it is not suitable for medium to heavy packet loss cases where the error control techniques such as FEC and ARQ should be applied.

1.1.2.3 Joint Source and Channel Coding

Traditionally, source and channel coding have been considered independently. The reason behind this is Shannon's principle of separability. According to Shannon's theory, an optimal communication system can be constructed by considering the source and channel coder designs separately. However, such a design may lead to intolerable long delay and extremely high complexity. In practical situations, it is of benefit to employ joint source-channel coding to achieve an optimum performance while maintaining moderate delay and complexity.

Any scheme jointly considering source coding efficiency and transportation reliability can be considered as a JSCC scheme. A class of the JSCC schemes is to jointly select source coding and transportation parameters such as in [20, 21, 22, 23] while still keeping the independence between source coding and transportation modules. Another class is to establish more connections between source coding and transportation, which is termed as priority-based transmission, i.e., providing different priorities for information with different importance. However, how to define the importance of different information and how to give them different priorities are very challenging tasks. There are many

ways to provide transportation priorities. For example, the transportation priorities are provided by different FEC codes in [24, 25, 26, 27, 28], different ARQ schemes in [29], different modulations in [30], different network QoS classes in [31], and different network routing paths in [32].

1.2 Problem Statement and Research Objective

Joint source-channel coding has been an active area of research. The techniques reported in the literature more or less focus on how to minimize a certain cost function (distortion, power consumption, delay, etc.) under given constraints. However, since joint source-channel coding involves many facets of communications, information theory, and signal processing, it is not easy to design an optimal JSCC image scheme with low complexity, and satisfying real-time requirement. Furthermore, with the emergence of JPEG2000 [33, 34], the latest image compression standard, many new challenges have been brought out to the issue of robust image transmission. For instance, compared with SPIHT [35] and JPEG [36], JPEG2000 incorporates many new concepts such as code-block, precinct, and quality layer. Although they can improve the compression performance and enhance error resilience of JPEG2000, they make it much more difficult to analyze the effect of lost channel packets on the distortion of the reconstructed image.

In this research, our objective is to study, design, and implement efficient and reliable joint source-channel coding for image transmission over packet loss channels. In other words, we will focus on joint source and channel bandwidth allocation to achieve the best image quality at the receiver end under the constraints of bandwidth, complexity and so on. In order to realize this goal, we need to investigate the existing joint source-channel coding schemes, and based on the investigation, we will propose our schemes to improve the overall performance.

1.3 Summary of Contributions

The major contributions of this thesis are listed as follows:

- Propose a layered unequal loss protection (L-ULP) scheme

Most existing unequal loss protection schemes do not consider the minimum quality requirement and usually have high computation complexity. In this research, we propose a L-ULP scheme to solve these problems by smartly choosing the number of layers. Numerical results show that the proposed L-ULP scheme can satisfy the minimum quality and low complexity requirements with slight performance degradation.

- Extend L-ULP to image transmission with delay constraints

In the case of high bit rate image transmission or having lots of packets, the FEC encoding and decoding processes in the ULP based schemes should be applied to each individual packet group instead of all the packets in order to avoid long processing delay. In this research, we extend the L-ULP to image transmission with delay constraints, where fast and efficient FEC allocations are performed among different packet groups and also within each packet group.

- Combine L-ULP with pre-interleaving

The performance of our proposed L-ULP is not as good as that of the hybrid loss protection (HLP) scheme [37]. In this research, we aim at improving the L-ULP performance by integrating it with pre-interleaving. This new scheme is named improved L-ULP (IL-ULP). The basic idea is to delay the occurrence of the first unrecoverable loss in the source data bit stream since the performance of a progressive image transmission is determined by the location of the first unrecoverable loss instead of the number of losses.

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- Apply IL-ULP to JPEG2000 image transmission

In this research, we apply IL-ULP to JPEG2000 coded images. JPEG2000 is quite different from the commonly used progressive image codec such as SPIHT since upon detecting an error JPEG2000 does not have to stop decoding due to its intrinsic error resilience properties. In particular, code-blocks in JPEG2000 are independently coded. Hence, bit errors in the bit stream of one code-block will not affect other code-blocks.

1.4 Outline

The remainder of this thesis is organized as follows. Chapter 2 gives a comprehensive literature review on recent techniques on FEC-based joint source-channel coding. Chapter 3 introduces the proposed L-ULP scheme and its extensions for progressive image transmission over packet loss channels. Finally, conclusion remarks are given in Chapter 4.

Chapter 2

A Review of Joint Source-Channel Coding

In this chapter, we first give a brief introduction on the techniques of joint source-channel coding. In Section 2.2, we describe the well-known set partitioning in hierarchical trees (SPIHT) image coding algorithm [35]. Then, we give a detailed review of the three most representative JSCC schemes: the unequal loss protection framework [24], the hybrid loss protection [38] and the fast local search method [39] in Section 2.3, 2.4 and 2.5, respectively. Finally, summary is given in Section 2.6.

2.1 Overview

Recently, we have seen extensive studies in FEC-based joint source-channel coding for progressive image and video transmission over noisy channels [22, 27, 28, 40] and over packet loss channels [24, 25, 26, 38, 41, 42, 43]. The common idea is to combine progressive source bit streams with unequal error/loss protection (UEP/ULP), i.e., the more important information is given more protection. Progressive image bit streams could be a good fit for UEP/ULP schemes because the earlier portions of a progressive image bit stream are always more important than the later ones. Compared with equal error/loss protection (EEP/ELP), considerable performance gain could be achieved with UEP/ULP

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with the added advantage of graceful performance degradation in case of channel mismatch. The complexity of UEP/ULP, however, is much higher than that of EEP/ELP since it is not trivial to find the optimal UEP/ULP solution.

Many schemes [22, 24, 27, 44, 45, 46, 47] have been proposed to find the optimal UEP/ULP solutions. In [27], dynamic programming was employed using fixed-length source data blocks. In [22], the authors also considered fixed-length source data blocks and developed an empirical model for optimal source channel rate allocation. In [24], Mohr et al. developed a ULP framework with fixed-length channel coding blocks, and used a greedy and iterative search algorithm to find the optimal channel coding rates which is computationally expensive. Stockhammer [44] gave an $O(N^2L^2)$ algorithm, where N is the number of packets and L is the packet length, to find the optimal ULP solutions under the conditions that the operational rate-distortion function is convex and the packet loss probability is monotonically decreasing with respect to the number of lost packets. Dumitrescu [45] presented an algorithm which under the same conditions as [44] reduces the complexity to $O(NL^2)$. Stankovic [46, 47] further reduced the complexity to $O(NL)$ by making use of the rate optimal solution with a local search for finding the distortion optimal solution.

Besides faster algorithms, near-optimal ULP solutions and simplified ULP architectures are also of interest [24, 25, 48]. In [24, 48], the gradient search method and the Lagrangian optimization approach are used to approximately achieve optimal FEC assignment. Kim et al. [25] reduced the complexity of the ULP by employing dynamic programming to find the optimal channel coding rates for each bit-plane instead of for each channel coding block. Although the experimental results in [25] show that the scheme can be executed much faster than the ULP, its improvement much depends on the number of bit-planes.

In addition to high complexity, most existing ULP schemes do not consider the minimum image quality requirement. This gives rise to applying extra ULP process to the

early portions of a bit stream, resulting in low quality reconstructed images. By observing this problem, a hybrid ULP and ELP (HLP) scheme is proposed in [37]. The basic idea of the HLP is to constrain the early portions of a progressive bit stream with ELP whose corresponding PSNRs are less than a threshold while ULP is applied to the rest of the bit stream.

2.2 Set Partitioning in Hierarchical Trees

The SPIHT algorithm [35] is motivated by, and has several features in common with, the embedded zerotree wavelet (EZW) method [49]. These two algorithms are a significant breakthrough in lossy image compression in that they provide substantially better compression performances than prior techniques such as JPEG [36] and vector quantization [50].

The SPIHT algorithm includes three basic steps [51]: 1) searching for sets in spatial-orientation trees in a wavelet transform; 2) grouping the wavelet transform coefficients in these trees into sets defined by the highest bit-plane of the magnitude; and 3) coding and transmitting bits associated with the highest remaining bit-planes first.

Spatial orientation trees are used to take advantage of the self-similarity among wavelet coefficients in different scales. The organization of wavelet coefficients into a tree is based on relating each coefficient at a given sub-band (parent) to a set of four coefficients along the same spatial orientation in the next higher frequency sub-bands (children). Figure 2.1 depicts the key for parent-child relationship of coefficients for a 2-D wavelet transform with two levels of decomposition. The coefficient marked with a star has no children.

The SPIHT algorithm transmits the wavelet coefficients in bit-plane order with most significant bit-plane first. For each bit-plane there are two passes, sorting pass and refinement pass. In the sorting pass, SPIHT sets a magnitude threshold 2^n . For a given

Figure 2.1: Examples of parent-child relationships in SPIHT.

threshold, the algorithm searches each tree, and the significant coefficients are found out and coded using the set partitioning method. In the refinement pass, the precision of all previously significant coefficients is enhanced by outputting the n -th most significant bit of the binary representation of their values. After the refinement pass, n decreases by 1 and another sorting pass begins.

SPIHT produces a fully embedded bit stream, in a manner that the quality of the decoded image is the best for the number of bits received up to that moment. Additionally, the transmission of the bit stream can stop at any point and a lower quality image can still be decompressed and reconstructed.

2.3 Unequal Loss Protection

Although the progressive image coding of SPIHT can provide excellent compression and scalability performance, it makes bit streams vulnerable to channel noise such as packet loss in the Internet. An error in a bit stream may cause all following bits become useless because, in a progressive bit stream, the later bits refine earlier bits, and the earlier

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Figure 2.2: An illustration of the ULP structure proposed by A. E. Mohr et al. Each row is a stream and each column is a channel packet. The numbers 1~32 are source data, and the symbol F is FEC.

bits are essential for the later bits to be useful. That is, the various parts of such a bit stream are not equally important. The earlier parts should have higher priorities since they are more important to the overall quality of the reconstructed image. A natural way to protect such a bit stream is unequal loss protection. ULP is the common idea of most joint source-channel coding schemes [13]. In this section, we will review the ULP framework proposed in [24].

Figure 2.2 depicts the structure of the ULP framework. In this framework, FEC is added to each message fragment to provide protection against packet loss such that the fragment and FEC together form a stream. The message is separated into L streams such that each stream contains one byte of each of N channel packets. In Figure 2.2, there are totally $L = 7$ streams and $N = 6$ packets. For a given stream i , $i = 1, 2, \dots, L$, the only condition under which the entire stream can be decoded is that the number of lost packets is not larger than the number of FEC bytes.

Let m_i be the number of source data bytes contained in stream i and let $f_i = N - m_i$

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be the number of FEC bytes allocated to the same stream. Then we can define $\vec{f}$ as

$$\vec{f} = (f_1, f_2, \dots, f_L), \quad (2.1)$$

whose elements are the length of FEC allocated to each stream. Furthermore, since the bit stream is progressive, it is natural to require that $f_i \geq f_{i+1}$, for $i = 1, 2, \dots, L - 1$.

For a specified $\vec{f}$, $M_i(\vec{f})$ is defined to be the sequence of source data bytes in stream i . Then the data bytes contained in the first j fragments can be denoted as

$$M(j, \vec{f}) = M_1(\vec{f})M_2(\vec{f}) \dots M_j(\vec{f}). \quad (2.2)$$

Let $PSNR[M(i, \vec{f})]$ be the peak signal-to-noise ratio (PSNR) value of the reconstructed image when the first i streams are successfully decoded, the incremental PSNR of stream i can be defined as

$$g_i(\vec{f}) = PSNR[M(i, \vec{f})] - PSNR[M(i-1, \vec{f})].$$

The quantity $g_i(\vec{f})$ is actually the amount by which the PSNR of the reconstructed image increases when stream i is decoded, given that all preceding $i - 1$ streams have already been decoded.

Finally we define $c(k) = \sum_{n=0}^k p_n$, for $k = 0, 1, \dots, N$, where p_n is the probability that n packets are lost. It is easy to know $c(f_i)$ is the probability that stream i can be successfully decoded. We can now compute the expected PSNR of the reconstructed image by summing over all the L streams

$$G(\vec{f}) = \sum_{i=1}^L c(f_i)g_i(\vec{f}). \quad (2.3)$$

Note that within the proposed ULP framework, finding the optimal FEC length of each stream is of high complexity. To reduce the complexity, the authors developed a local search hill-climbing algorithm [24]. The algorithm adopts a parameter Q to achieve the tradeoff between the complexity and the performance. The larger the Q is, the more likely the algorithm finds the global optimum FEC lengths but with longer execution time.

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Figure 2.3: Block diagram of the (a) ULP and (b) HLP scheme. Each row is a stream and each column is a channel packet. In (b), the shadow area in the source data part is for the minimum quality requirement.

2.4 Hybrid Loss Protection

Besides its high complexity issue, another shortcoming of the ULP framework is that it merely try to maximize the expected PSNR of the decoded image at the receiver end. Maximizing the expected PSNR is not necessarily able to guarantee a given quality of service in multimedia applications. Instead, it is shown that maximizing the expected PSNR may yield a considerable probability of obtaining failure transmissions (that is, decoded images are with PSNR values lower than a certain threshold) and, even worse, it does not provide any control on the probability that this event occurs.

Figure 2.3(b) illustrates the block diagram of the HLP scheme proposed in [38]. As a comparison, the traditional ULP framework is also illustrated in Figure 2.3(a). Let $PSNR_{min}$ denote the minimum quality threshold. As mentioned in Section 2.1, the basic idea of HLP is to assign the same amount of protection to the earliest part of the bit stream up to a quality equal to $PSNR_{min}$. A ULP allocation can then be utilized only to the remainder of the bit stream. That is why the method is name hybrid loss protection.

Define the failure probability P_f as the probability that the quality of the decoded image is less than $PSNR_{min}$. Let P_{f0} be the desired value of this failure probability.

Then the optimal FEC assignment $\vec{f}$ is the solution to the following problem

$$\begin{cases} \vec{f} = \arg \max_f G(\vec{f}) \\ \text{subject to } P_f < P_{f0} \end{cases} \quad (2.4)$$

$\vec{f}$ and $G(\vec{f})$ have been defined in Section 2.3. For the sake of completeness, we repeat their definitions here

$$\vec{f} = (f_1, f_2, \dots, f_L),$$

and

$$G(\vec{f}) = \sum_{i=1}^L c(f_i) g_i(\vec{f}).$$

Suppose the ELP part contains L_1 streams, then we have $f_1 = f_2 = \dots = f_{L_1}$. P_f can then be calculated by

$$P_f = \sum_{n=f_1+1}^N p_n, \quad (2.5)$$

where p_n is the probability that n packets are lost.

A three-step FEC allocation algorithm is also proposed for the HLP scheme [38]. 1) estimation of the channel statistics; 2) optimization of the ELP part, which calculates the amount of source data that meets the $PSNR_{min}$ requirement; and 3) search of the optimal ULP allocation for the remaining streams from $L_1 + 1$ to L .

The complexity of HLP is lower than that of ULP. However, since HLP still needs to find out the optimal ULP solution for the streams from $L_1 + 1$ to L , its complexity remains high, especially when L is large.

2.5 Fast Local Search Algorithm

For real-time applications such as live video streaming, due to the time-varying nature of the source and channel statistics, the FEC allocation solution must be dynamically obtained. In addition, memory limitations also require low-complexity algorithm design. However, it is clear that both the ULP and the HLP do not meet this requirement.

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Therefore, fast algorithms for the computation of optimal or sub-optimal FEC allocation solutions are necessary. In this section, we will review the fast local search algorithm proposed in [39].

We first define some symbols in order to facilitate the description of the local search algorithm. Let $\hat{F}$ be the set of L -tuple $\vec{f}$ such that $f_i \in \{0, 1, \dots, N-1\}$ for $i = 1, 2, \dots, L$. $\vec{f}$ is defined in Eqn. (2.1). f_i , L , and N also own the same definitions as those defined in Section 2.3. Similarly, let p_n denote the probability that n packets are lost, and ϕ denote the rate-distortion (R-D) function of the source codec. Then the task of the local search algorithm is to find a protection scheme $\vec{f}$ that minimizes the expected distortion

$$E[d](\vec{f}) = \sum_{i=0}^L P_i(\vec{f}) \cdot \phi(M(i, \vec{f})), \quad (2.6)$$

where $M(i, \vec{f})$ is the source data contained in the first i fragments and is defined in Eqn. (2.2).

$P_i(\vec{f})$ is defined as follows

$$P_i(\vec{f}) = \begin{cases} \Pr(X > f_1) & \text{for } i = 0 \\ \Pr(f_{i+1} < X \leq f_i) & \text{for } i = 1, 2, \dots, L-1 \\ \Pr(X \leq f_L) & \text{for } i = L \end{cases}$$

where X is a random variable whose value is the number of lost packets.

The local search algorithm is motivated by the fact that, if the operational R-D function of the source codec is non-increasing and convex, the distortion-optimal solution tends to provide stronger protection than (at least equal to) that of a rate-optimal solution [46]. The term rate-optimal means a solution that maximizes the expected number of correctly received source symbols

$$E[r](\vec{f}) = \sum_{i=0}^L P_i(\vec{f}) \cdot M(i, \vec{f}).$$

In fact, a rate-optimal solution is the equal loss protection $\vec{f} = (f_r, f_r, \dots, f_r)$, where

$$f_r = \arg \max_{i=0, \dots, N-1} (N-i) \sum_{n=0}^i p_n.$$

The local search algorithm starts from the rate-optimal solution $\vec{f}$ and searches for the best candidate in its neighborhood. Let $\vec{f} = (f_1, f_2, \dots, f_L) \in \hat{F}$. The neighborhood of $\vec{f}$ consists of the solutions of the form $(f_1 + 1, f_2, \dots, f_L)$, $(f_1 + 1, f_2 + 1, \dots, f_L)$, $\dots$, $(f_1 + 1, f_2 + 1, \dots, f_L + 1)$ that are included in $\hat{F}$. If the best candidate can offer better performance than the current $\vec{f}$, it is adopted as the new $\vec{f}$ and the search repeats. Otherwise, the search process stops.

The complexity of the fast local search algorithm is lower than all the previous algorithms such as [52]. However, it can only provide sub-optimal solutions although the solutions are claimed to be close to the optimal one.

2.6 Summary

In this chapter, we have studied the recent techniques on joint source-channel coding. In particular, we review several algorithms, which are related to our research work, including SPIHT, the ULP framework, the hybrid loss protection, and the fast local search algorithm. The literature review builds the foundation for our further research efforts.

Chapter 3

Layered Unequal Loss Protection for Progressive Image Transmission Over Packet Loss Channels

In this chapter, we propose a layered ULP scheme for fast progressive image transmission over packet loss channels. Our proposed L-ULP is able to tackle both the minimum quality requirement and the high computation complexity issue. We are able to achieve a complexity of $O(nN^2)$ for L-ULP, where n is the number of layers and N is the number of packets. The basic operation is the calculation of one R-D slope and the corresponding sorting. By choosing different number of layers, the complexity of L-ULP is adjustable. In particular, we use the rate-based optimal solution with a local search to find the average FEC allocation and use the gradient search to find the FEC solution for each layer. Experimental results show that the executing time of L-ULP is much faster than the traditional ULP scheme but the average distortion performance is not satisfying. Therefore, we further propose to combine the L-ULP with the pre-interleaving to have an improved L-ULP system. By using the pre-interleaving, we are able to delay the occurrence of the first unrecoverable loss in the source bit stream and thus improve the performance of L-ULP. Experimental results show that IL-ULP even outperforms HLP while the complexity of IL-ULP is almost the same as L-ULP.

This chapter is organized as follows. Section 3.1 describes the proposed layered ULP scheme. Section 3.2 applies the L-ULP to the cases with delay constraints. In Section 3.3, we present the improved L-ULP scheme. In Section 3.4, IL-ULP is utilized to transmit JPEG2000 encoded images. Concluding remarks are given in Section 3.5.

3.1 Proposed L-ULP Scheme

3.1.1 Problem Statement

As stated in Chapter 2, in the benchmark ULP scheme [24], a major problem is that it merely aims at maximizing the expected PSNR and does not have any control to prevent the occurrence of low PSNR events. It is well-known that a received image with the PSNR value below a certain threshold is useless for any practical application. As an example, Figure 3.1 shows the reconstructed standard 512×512 gray scale Lena image with the PSNR value of 22.17 dB. It is almost unreadable. Note that such a PSNR threshold can be determined according to the human visual system. In this research, for simplicity, we choose 25 dB as the lowest acceptable image quality. The ULP scheme does not consider such a minimum PSNR requirement and still allocates different amounts of FEC to the early portions of a bit stream whose corresponding PSNRs are less than the threshold. This results in unnecessary complexity increase and also increases the probability of unsuccessful transmission if we consider any image transmission with a PSNR less than the threshold as a failure transmission. Figure 3.2 shows a typical PSNR degradation performance of using ULP to transmit the Lena image coded at 0.2 bpp with 47 bytes per packet. It can be seen that a noticeable portion of the ULP results which is claimed superior to ELP is actually below 25 dB and thus meaningless. Such an observation has been pointed out in [38].

Another major problem of the ULP is its high complexity which has been mentioned in Chapter 2. As shown in Figure 3.3(a), the ULP allocates different amounts f_i of

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Figure 3.1: The reconstructed 512×512 Lena image with PSNR 22.17 dB.

Figure 3.2: The PSNR performance of using ULP to transmit the Lena image coded at 0.2 bpp with 47 bytes per packet under different numbers of lost packets. The channel loss model is an exponential PMF (probability mass function) with a mean loss rate of 0.2.

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Figure 3.3: (a): The ULP structure. (b): A general layered ULP structure.

FEC codes to different streams. Each stream is basically a channel coding block. A greedy and iterative search algorithm is used in [24] to find the optimal ULP solutions f_i , $i = 1, 2, \dots, L$. Although such a ULP scheme works fine for small packet size L such as the size of a ATM cell, it is not suitable for large values of L such as Internet packet sizes since, with the increase of L , the complexity and the execution time of ULP become intolerable.

3.1.2 L-ULP System Description and Analysis

A straightforward method is to simplify the ULP scheme by adopting layers. In layered ULP, a layer is defined as a group of consecutive rows with the same loss protection choices independent of other rows. The architecture of a general L-ULP scheme is shown in Figure 3.3(b). In a $L \times N$ rectangle where L is the packet length and N is the number of packets, each row is a channel coding block and each column is a packet. Let L_i and f_i denote the number of rows and the allocated FEC length for the i -th layer. Reed-Solomon codes with 8 bits/symbol are used as channel codes. An $(N, N - f_i)$ RS code encodes each segment of $N - f_i$ source symbols into a channel block of N symbols, and it can correct up to f_i symbol loss. The expected distortion in mean square error (MSE)

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at the receiver end can be formulated as

$$D = d_0 - \sum_{i=1}^n \Delta d_i \cdot C_i, \quad (3.1)$$

where d_0 is the distortion of not using any packet for reconstruction, n is the number of layers, C_i is the probability of correctly decoding the i -th layer, and Δd_i is the corresponding distortion gain. We can see that such a layered ULP is even more complicate than the conventional ULP [24] because we need to find not only the optimal FEC allocation f_i but also the optimal layer division L_i (Assuming n is given.).

In [25], Kim et al. simplified this problem by mapping each bit-plane into one layer. In this work, we divide layers according to source coding R-D curves. The basic idea is to let each layer have equal distortion gain while the first layer must satisfy the minimum quality requirement. The reason we divide layers with equal distortion gain is just for simplicity, and it also makes sense to proactical applications from the QoS point of view. Certainly, we can divide layers in other ways. The source R-D curve $D_s(R_s)$ of a progressive image coding scheme such as SPIHT can be obtained by extracting some R-D points such as the end points of each bit-plane during the encoding process followed by linear interpolation between neighbor R-D points [53]. Therefore, given the accumulated source data amount up to the i -th layer b_i , the corresponding distortion d_i is given by

$$d_i = D_s(b_i). \quad (3.2)$$

The layer division can be described as

$$b_i = \begin{cases} D_s^{-1}(d_s^h) & \text{if } i = 1 \\ L(N - \bar{f}) & \text{if } i = n \\ D_s^{-1}(d_s^h - (i - 1)\Delta d_s) & \text{if } i = 2, \dots, n - 1 \end{cases} \quad (3.3)$$

where d_s^h is the highest distortion determined by the minimum quality requirement, $\bar{f}$ is the average FEC protection, $\Delta d_s = \frac{d_s^h - d_s^l}{n-1}$ and $d_s^l = D_s(b_n)$. In particular, the amount of the first layer source data b_1 is obtained according to the minimum quality requirement

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d_s^h . Then, we compute the largest amount of source data b_n according to the average FEC protection $\bar{f}$. Based on b_n , we can obtain the lowest distortion bound d_s^l . After this, the $n - 1$ layers are divided based on the idea of having equal distortion gain Δd_s in the range of $[d_s^l, d_s^h]$ for each layer. Note that the actual value of b_i may need to change a little during the implementation due to byte alignment and channel block alignment.

In this research, we assume the alignment issues can be neglected and we re-formulate the expected distortion after the layer division as

$$D(\vec{f}) = d_0 - (d_0 - d_s^h) \cdot C_1(f_1) - \Delta d_s \sum_{i=2}^n C_i(f_i), \quad (3.4)$$

where $\vec{f} = \{f_1, f_2, \dots, f_n\}$, $f_1 \geq f_2 \geq \dots \geq f_n$. $C_i(f_i)$ can be derived as

$$C_i(f_i) = \sum_{k=0}^{f_i} P(N, k), \quad (3.5)$$

where $P(N, k)$ is the probability of having k lost packets out of the total N packets. $P(N, k)$ can be calculated according to the given packet loss model. In this research, we use the packet loss model with the exponential probability mass function (PMF), the same as that in [24]. Hence, $P(N, k)$ can be computed as

$$P(N, k) = \frac{\frac{1}{p_l} e^{-\frac{1}{p_l} \frac{k}{N}}}{\sum_{i=0}^{i=N} \frac{1}{p_l} e^{-\frac{1}{p_l} \frac{i}{N}}}, \quad (3.6)$$

where p_l is the mean packet loss rate. Other packet loss models such as the Gilbert-Elliott Channel (GEC) models [54] can also be adopted in our proposed system.

From Eqn. (3.4) we can see that d_0 , d_s^h and Δd_s are either constant or fixed after the layer division, and the expected distortion can be separated into the n independent units. A unit i is associated with a variable $C_i(f_i)$, which only depends on f_i and is independent of the FEC allocation for other layers. Thus, the problem of finding the optimal FEC allocation is the same as optimal bit allocation among n independent units, which can

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be solved by processing the falling convex hull of the R-D slopes, i.e., the gradient search method. Details can be found in [55, 56].

Note that we have not discussed how to obtain the average FEC protection $\bar{f}$ in Eqn. (3.3). One way is to simply try all the possible values over the entire range. Fortunately, the work in [47] points out that the total number of protection in the distortion-based optimal solution should not be less than that in the rate-based optimal solution. Therefore, we can limit the search range for $\bar{f}$ to be $[f^*, N - 1]$, where f^* is the rate-based optimal solution obtained by

$$f^* = \arg \max_f (N - f) \cdot C(f). \quad (3.7)$$

Based on the analysis above, the proposed L-ULP algorithm, $L - ULP(N, L, n)$, is summarized as follows:

- Step 1: Find the optimal ELP solution f^* by using common search methods such as exhaustive search or bisection search methods. Initialize the search range for $\bar{f}$ to be $[f^*, N - 1]$. Initialize $D^* = d_0$ and $\vec{f}^* = 0$.
- Step 2: Orderly pick a value for $\bar{f}$ from the search range according to the rule of exhaustive search or bisection search.
- Step 3: For the selected $\bar{f}$, compute b_i according to Eqn. (3.3).
- Step 4: Use the gradient search method [55, 56] to find the optimal ULP solution $\vec{f}$ which minimizes the distortion shown in Eqn. (3.4) under the total bandwidth constraint. If $D(\vec{f}) < D^*$, set $D^* = D(\vec{f})$ and $\vec{f}^* = \vec{f}$. Update the search range for $\bar{f}$ accordingly.
- Step 5: If we still have non-empty search range for $\bar{f}$, go back to Step 2. Otherwise, output $\vec{f}^*$.

3.1.3 Complexity Analysis of L-ULP

In L-ULP, the complexity of finding the optimal ELP solution is $O(N)$ because there are N possible values for f^* . For each value of $\bar{f}$, we need to compute at most nN R-D points and $n(N - 1)$ R-D slopes, and sort those R-D slopes to find the equal slope points. Hence, the complexity of the gradient search is $O(nN)$ for one $\bar{f}$ value. Since there are $N - f^*$ possible values for $\bar{f}$, the overall complexity for L-ULP is approximately $O(nN^2)$, which is much lower than the $O(N^2L^2)$ algorithms in [45] and comparable to the $O(NL)$ algorithm in [47]. Note that the basic operation for the algorithms in [45, 47] is the computation of the cost function, which involves L multiplications and L additions, while the basic operation in our proposed L-ULP is much simpler, i.e., calculating one R-D slope and the corresponding sorting. In addition, the complexity of our proposed L-ULP scheme is scalable through setting different numbers of layers n .

3.1.4 PSNR Criteria Vs Distortion Criteria

PSNR is typically employed as the image quality measurement in the literature such as in [24, 25]. However, in this research, we use the average distortion for the quality measurement. This is because the maximal average PSNR does not correspond to the minimal average distortion due to the logarithm function. It is well-known that the logarithm function magnifies small values but shrinks large values, which makes the average PSNR not appropriate to represent the expected quality of the received image. For example, if an image transmission system has 50% chance of obtaining the reconstructed image with PSNR 20 dB (Distortion: 25.5^2) and another 50% chance of obtaining the reconstructed image with PSNR 40 dB (Distortion: 2.55^2), the average PSNR will be $(20 + 40)/2 = 30$ dB while the PSNR corresponding to the average distortion is $10 \log_{10}(255^2 \times 2 / (25.5^2 + 2.55^2)) = 22.97$ dB which is more reasonable in terms of mean square error. Figure 3.4 shows the allocated FEC fractions of using the proposed

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Figure 3.4: The allocated FEC fractions of using the 3L-ULP to transmit the Lena image over the packet-loss channel model with the exponential PMF and $p_l = 0.2$. The total bandwidth is 0.25 bpp. (a): Using the PSNR criteria. (b): Using the distortion criteria.

L-ULP scheme with three layers (3L-ULP) to transmit the Lena image over the packet-loss channel model under different criteria. It can be observed that the system under the distortion criteria spends more bits on protecting the first layer than that under PSNR criteria in order to reduce the probability of obtaining large distortion.

3.1.5 Numerical Results

The six 512×512 images with 8 bits per pixel, “Lena”, “Barbara”, “Tank”, “Boat”, “Goldhill”, “Peppers”, are used as the test images, as shown in Figure 3.5. We choose the packet size of 100 bytes and the number of packets is 82, corresponding to a total bandwidth of 0.25 bpp. We use the exponential PMF packet loss model. SPIHT is adopted as the codec for source coding and RS codes with 8 bits/symbol are used for channel coding. The number of layers is fixed to be 3. We choose 25 dB as a PSNR threshold for the minimum quality requirement. Any image transmission with a PSNR value less than the threshold is deemed as a failure.

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Figure 3.5: The six 512×512 test images. From left to right, top to bottom: “Lena”, “Barbara”, “Tank”, “Boat”, “Goldhill”, “Peppers”.

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We compare our proposed L-ULP scheme with the HLP [38]. We have known that the HLP provides the foremost several rows with equal loss protection in order to satisfy the minimum quality requirement. It then considers each row of the rest as a layer and uses iterative search to find the optimal FEC allocation f_i which minimizes the overall distortion based on Eqn. (3.1). Therefore, the performance of the HLP should be the upper bound for our proposed L-ULP extension scheme. We try to compare their performance and complexity.

The performance parameters include “ P_f ” and “Average Distortion”. “ P_f ” denotes the probability of failure image transmission. “Average Distortion” is the distortion value without counting the contributions from failure image transmissions. The complexity of these two approaches is measured in terms of the number of loops executed in the corresponding optimal FEC allocation algorithms since the optimal FEC allocation is the most time-consuming portion in all the ULP schemes. For the HLP, the number of loops is considered as the number of times of calculating the distortion in Eqn. (3.1). While for the L-ULP extension scheme, the number of loops is considered as the number of times of calculating R-D slopes.

Table 3.1 and 3.2 show the performance comparison of transmitting test images under different packet loss rates. As shown in Table 3.1 and 3.2, the L-ULP scheme can always obtain comparable P_f values with those of the HLP. Then we can conclude that the proposed L-ULP can also guarantee the minimum quality requirement. In addition, the complexity of the L-ULP scheme is much lower than that of the HLP while its distortion performance is slightly inferior, up to 4.65 in distortion increase. Note that in the case of transmitting Barbara, the numerical results look exceptional, i.e., both schemes achieve the same distortion performance when the packet loss rates are 0.1, 0.15, and 0.2. Actually, under such simulation conditions, their performance is the same with that of ELP. This is because the Barbara image is quite difficult to be compressed and

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Table 3.1: The performance comparison of transmitting test images over the packet loss channel model with the exponential PMF and under different values of packet loss rate. The total bandwidth is 0.25 bpp.

Packet Loss Rate	Image	Scheme	$P_f(\%)$	Average Distortion	Number of Loops
0.05	Lena	L-ULP	0.04	35.69	5627
		HLP	0.03	33.82	326899
	Barbara	L-ULP	0.47	168.64	687
		HLP	0.47	167.42	22175
	Tank	L-ULP	0.11	89.05	3360
		HLP	0.08	87.92	215211
	Boat	L-ULP	0.18	90.16	2032
		HLP	0.14	87.99	143440
	Goldhill	L-ULP	0.04	71.70	4277
		HLP	0.04	69.95	240625
	Peppers	L-ULP	0.04	41.35	4335
		HLP	0.03	39.34	340932
0.1	Lena	L-ULP	0.17	54.20	8259
		HLP	0.17	51.03	678506
	Barbara	L-ULP	1.24	239.60	875
		HLP	1.24	239.60	19321
	Tank	L-ULP	0.59	103.29	6141
		HLP	0.59	100.48	364562
	Boat	L-ULP	0.52	125.25	3893
		HLP	0.52	122.08	298074
	Goldhill	L-ULP	0.22	93.28	7817
		HLP	0.22	90.18	603187
	Peppers	L-ULP	0.17	64.76	7263
		HLP	0.17	61.72	685788

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Table 3.2: The performance comparison of transmitting test images over the packet loss channel model with the exponential PMF and under different values of packet loss rate. The total bandwidth is 0.25 bpp.

Packet Loss Rate	Image	Scheme	$P_f(\%)$	Average Distortion	Number of Loops
0.15	Lena	L-ULP	0.43	74.81	4693
		HLP	0.48	71.69	1424802
	Barbara	L-ULP	5.25	351.22	875
		HLP	5.25	351.22	21325
	Tank	L-ULP	1.23	116.83	6694
		HLP	1.34	113.32	602125
	Boat	L-ULP	1.12	162.36	3781
		HLP	1.12	159.53	444257
	Goldhill	L-ULP	0.53	117.27	5269
		HLP	0.59	114.09	1145323
	Peppers	L-ULP	0.48	92.02	3774
		HLP	0.48	89.17	1154985
0.2	Lena	L-ULP	1.16	98.70	4711
		HLP	1.16	94.07	1399210
	Barbara	L-ULP	10.57	499.38	875
		HLP	10.57	499.38	22763
	Tank	L-ULP	1.96	128.79	4733
		HLP	2.12	125.17	993234
	Boat	L-ULP	2.48	200.27	3802
		HLP	2.48	197.46	481662
	Goldhill	L-ULP	1.16	142.63	4796
		HLP	1.16	137.98	1392785
	Peppers	L-ULP	1.27	129.13	4171
		HLP	1.27	124.61	1056064

it needs a lot of data to reach the minimum quality requirement, PSNR 25 dB. Then, at a total bandwidth of 0.25 bpp, there is no much space left for unequal protection. Numerical results in Section 3.3.3 will illustrate this point further.

3.2 Layered Unequal Loss Protection with Delay Constraint

3.2.1 Problem Formulation

So far, the delay problem in the ULP based image transmission has not been addressed much. Most of the ULP schemes consider all the packets of one image entirely for the FEC encoding and decoding processing. This works fine for low bit rate image transmission since the number of packets for one image is quite limited. However, in the case of high bit rate image transmission or having lots of packets, the conventional schemes will cause long delay or slow response to users' requirements since the FEC decoder has to wait until the arrival or the timeout of the last transmitted packet. In this section, we extend our previous L-ULP work in Section 3.1 to the case of image transmission with the delay constraint. In particular, in order to satisfy the delay requirement, we divide all the packets into several groups and FEC is applied to each packet group. We formulate the L-ULP framework under such a delay-constrained scenario and propose a two-step approach to find the optimal FEC allocation.

In the case with the delay constraint, we consider there are multiple independent rectangles. Each rectangle consists of a group of packets as shown in the right of Figure 3.3. The size of a rectangle is limited by the delay constraint. The expected distortion in this case can be written as

$$D = d_0 - \sum_{i=1}^m \sum_{j=1}^{n_i} \Delta d_i^j \cdot C_i^j, \quad (3.8)$$

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where m is the number of rectangles, n_i is the number of layers in the i -th rectangle, C_i^j is the probability of correctly decoding the j -th layer in the i -th rectangle, and Δd_i^j is the corresponding distortion gain. Suppose n_i is pre-chosen according to users' requirements or network configuration, which is typically a small value in order to keep low complexity. The L-ULP problem under the delay constraint can be summarized as: given all the values of n_i , how to find the optimal f_i^j and L_i^j so that the expected distortion shown in Eqn. (3.8) can be minimized, where f_i^j and L_i^j denote the allocated FEC length and the number of rows for the j -th layer in the i -th rectangle, respectively.

3.2.2 Proposed L-ULP for Delay-Constrained Transmission

Regarding to this problem of applying L-ULP under the delay constraint, we propose a two-step approach to perform the FEC allocation and the layer division. In particular, in the first step we consider the FEC allocation among rectangles. In other words, we only consider one layer in one rectangle at the first step. After we obtain the average FEC amounts for each rectangle, we can apply our L-ULP scheme in Section 3.1 directly to obtain the FEC allocation and the layer division in each rectangle. The basic idea here is to reduce the dependency of FEC allocation among different rectangles so that L-ULP can be processed in a fast manner. We believe such an approach can achieve sub-optimal performance which is close to that of global optimization using iterative search. This is because many studies show that under channel matching conditions the minimum distortion achieved by the ULP is only slightly better than that of the optimal ELP. Our numerical results in the next section also provide very supportive evidence for the effectiveness of our proposed scheme. In the following, we describe the proposed two-step approach in detail.

- Step 1: We consider only one layer in one rectangle and use the iterative search method to find the optimal FEC allocation $\vec{F}$ among the m rectangles which min-

imizes

$$D(\vec{F}) = d_0 - \sum_{i=1}^m \Delta d_i(\vec{F}) \cdot C_i(\vec{F}), \quad (3.9)$$

where $\vec{F} = \{\bar{f}_1, \bar{f}_2, \dots, \bar{f}_m\}$, $\bar{f}_1 \geq \bar{f}_2 \geq \dots \geq \bar{f}_m$, and $C_i = \prod_{j=1}^i [\sum_{k=0}^{\bar{f}_j} P(N_j, k)]$, where $P(N_j, k)$ is the probability of having k lost packets out of the total N_j packets.

Note that although we use iterative search at this step, the complexity is very low since m is usually a small number.

- Step 2: Based on the obtained $\vec{F}$, compute b_i according to Eqn. (3.3). Then use the gradient search method [55, 56] to find the optimal ULP solution for each rectangle independently.

3.2.3 Numerical Results of L-ULP with Delay Constraints

In this research, we choose a simple layer configuration (5,5) for our proposed L-ULP extension scheme, i.e., five layers in each rectangle. The width of each rectangle is limited to 80 packets and there are totally 160 packets. We choose the packet size of 100 bytes. This corresponds to a total bandwidth about 0.488 bpp. We use all other five test images excluding Barbara from Section 3.1. Other experimental settings, for example the channel model and the PSNR threshold and so on, are the same as those in Section 3.1.

We compare our proposed L-ULP extension scheme with the HLP [38]. The purpose of the HLP is to find the optimal FEC allocation f_i^j , which minimizes the overall distortion based on Eqn. (3.8). Figure 3.6 and 3.7 show the performance comparison of transmitting the test images under different packet loss rates. The performance is measured in terms of P_f and average effective PSNR. It is shown that our proposed L-ULP extension scheme achieves smaller probability of failure image transmission than the HLP for most cases, up to 0.7% reduction, while the PSNR degradation is not significant, around 0.2 dB to 0.8 dB. When the packet loss rate is 0.2, however, it seems L-ULP produces larger

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Figure 3.6: The performance comparison between HLP and L-ULP for Lena, Goldhill, and Tank, respectively. Figures in the left column are the probability of failure transmission under different packet loss rates. Figures in the right column are the average effective PSNR under different packet loss rates.

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Figure 3.7: The performance comparison between HLP and L-ULP for Peppers and Boat, respectively. Figures in the left column are the probability of failure transmission under different packet loss rates. Figures in the right column are the average effective PSNR under different packet loss rates.

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Table 3.3: The complexity comparison.

Packet Loss Rate	Image	Number of Loops	
		L-ULP	HLP
0.05	Lena	510	181459
	Tank	427	259781
	Boat	444	269105
	Goldhill	598	328820
	Peppers	627	406623
0.1	Lena	830	234963
	Tank	628	587333
	Boat	489	403224
	Goldhill	648	951920
	Peppers	615	873774
0.15	Lena	911	429428
	Tank	886	1261725
	Boat	684	887828
	Goldhill	853	1910653
	Peppers	821	1670046
0.2	Lena	956	793860
	Tank	964	1507635
	Boat	550	686333
	Goldhill	926	1609660
	Peppers	1075	2111596

P_f value than that of HLP for the Boat test image, as shown in Figure 3.7. This is because the higher the packet loss rate, the larger the optimal EEP FEC length tends to be. In addition, the Boat image is somewhat difficult to compress. As a result, there are not enough source bytes in the first rectangle to meet the 25 dB minimum quality requirement for such a case. Thus, the FEC length has to be reduced to increase the number of source bytes, which is followed by larger P_f values.

The complexity of these two approaches is measured in terms of the number of loops executed in the corresponding optimal FEC allocation algorithms, the same as in Section 3.1. Similarly, for the HLP, the number of loops is considered as the number of times of calculating the distortion in Eqn. (3.8). While for the L-ULP extension scheme,

the number of loops is considered as the number of times of calculating R-D slopes. The complexity comparison is shown in Table 3.3. It can be seen that the complexity of the proposed L-ULP extension, in the order of 10^2 , is much lower than that of the HLP which is in the order of up to 10^6 .

3.3 Improved Layered Unequal Loss Protection

As shown in Section 3.1 and 3.2, although our proposed L-ULP is able to achieve low-complexity and meet the minimum quality requirement, its average distortion performance is not as good as that of the HLP. In this section, we propose an improved L-ULP (IL-ULP) scheme to further enhance the performance of the L-ULP.

3.3.1 System Description

An interesting channel decoding scheme is reported in [57] which does not aim at achieving low error rate, but rather makes the first error as far from the beginning as possible under the actual channel condition. Such a channel decoding scheme can be well combined with progressive image coding since the performance of a progressive image transmission is very much determined by the location of the first unrecoverable error instead of the amount of errors. This important property has not been received enough attention in the area of joint source-channel coding for progressive data transmission.

Motivated by the idea presented in [57], in this work we propose an improved L-ULP (IL-ULP) scheme, which is a combination of pre-interleaving with our previous proposed L-ULP, for progressive image transmission over packet loss channels. By using the pre-interleaving, we are able to delay the occurrence of the first unrecoverable loss in the source bit stream and thus improve the performance of L-ULP. This is the major innovation of our proposed IL-ULP scheme. Figure 3.8 gives an example of the L-ULP with three layers to illustrate the basic idea of this innovation. Figure 3.8(a) is the original

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Figure 3.8: An example of the L-ULP with three layers. (a): the original L-ULP. (b): with the pre-interleaving.

L-ULP, where the source bit stream is placed in the rectangle from left to right and from top to bottom. Figure 3.8(b) is the L-ULP with pre-interleaving, where in the rectangle the source bit stream is still placed from top to bottom but in each layer it is placed in the vertical direction instead of the horizontal direction. Since such an interleaving is performed before channel encoding, we name it pre-interleaving. The advantage of applying the pre-interleaving can be explained by the following example. Suppose there exist unrecoverable packet losses and the first unrecoverable loss occurs in the j -th packet in the third layer. The positions of the lost packets can be obtained through checking the sequence number field in the header of each packet. For this case, the original L-ULP can use maximal $[b_1 + b_2 + (j - 1)]$ symbols for source decoding while the IL-ULP scheme can take maximal $[b_1 + b_2 + (j - 1) \cdot L_3]$ for source decoding, which will result in a better performance. The performance gain will be even more significant in the cases of small values of n and large values of L such as Internet packet sizes.

Note that the pre-interleaving is not suitable for the traditional ULP scheme [24].

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This is because, in the traditional ULP, different rows in the rectangle have different priorities and in order to provide higher priorities for more important information the source bit stream has to be placed from left to right and from top to bottom. We would also like to point out the pre-interleaving is not necessary to be applied to the first layer in the L-ULP, which is designed for the minimum quality requirement, since we consider any source decoding with data amount less than b_1 as a transmission failure. In addition, the pre-interleaving will only increase the processing complexity and will not increase any transmission delay.

3.3.2 Optimal FEC Allocation

The expected distortion for the proposed L-ULP scheme with the pre-interleaving can be written as

$$D(\vec{f}) = d_0 - \sum_{i=1}^n \sum_{j=1}^{N-f_i} \Delta d_i^j \cdot C_i^j, \quad (3.10)$$

where $\vec{f} = \{f_1, f_2, \dots, f_n\}$, $f_1 \geq f_2 \geq \dots \geq f_n$, n is the number of layers, C_i^j is the probability of correctly decoding the j -th packet in the i -th layer and Δd_i^j is the corresponding distortion gain. Given f_i , Δd_i^j can be obtained according to source coding R-D curves $D_s(R_s)$. C_i^j can be calculated according to the given packet loss model. Compared with the cost function in Eqn. (3.1), the cost function in Eqn. (3.10) is too complicate, which requires $n(N - f_i)$ multiplication and addition. This contradicts our initial objective, i.e., to propose a low complexity L-ULP scheme for fast image transmission over packet loss channels. Therefore, in this paper, we make a few assumption to simplify the cost function.

In particular, we assume that, for the packet loss model with the exponential PMF, lost packets are consecutive in an N -packet transmission and it can start at any position with equal probability. This assumption is reasonable since in practice packet loss caused by network congestion or channel deterioration tends to occur in bursts. Second,

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we assume that the R-D relationship within one layer is linear. Based on these two assumptions, we derive the cost function for IL-ULP as

$$D(\vec{f}) = d_0 - \sum_{i=1}^n \Delta d_i \cdot \left[\sum_{k=0}^{f_i} P(N, k) + \frac{1}{N - f_i} \sum_{k=f_i+1}^N P(N, K) \cdot \frac{N - k}{2} \right], \quad (3.11)$$

where $\Delta d_i \cdot \sum_{k=0}^{f_i} P(N, k)$ is the expected distortion gain due to correctly decoding the i -th layer and $\frac{\Delta d_i}{N - f_i} \sum_{k=f_i+1}^N P(N, k) \cdot \frac{N - k}{2}$ is the expected distortion gain due to pre-interleaving in the case of failing to recover lost packets. Since $[\sum_{k=0}^{f_i} P(N, k) + \frac{1}{N - f_i} \sum_{k=f_i+1}^N P(N, k) \cdot \frac{N - k}{2}]$ can be calculated off-line, the complexity of Eqn. (3.11) is the same as that of Eqn. (3.1).

Similarly, considering the effect of pre-interleaving, the rate-based optimal solution f^* should be

$$f^* = \arg \max_f (N - f) \sum_{k=0}^f P(N, k) + \sum_{k=f+1}^N \frac{N - k}{2} \cdot P(N, k). \quad (3.12)$$

Then, we directly apply the algorithm $L - ULP(N, L, n)$ described in Section 3.1 for IL-ULP. The only difference is that for IL-ULP we use Eqn. (3.11) and Eqn. (3.12) instead of Eqn. (3.4) and Eqn. (3.7).

3.3.3 Numerical Results

The six 512×512 images with 8 bits per pixel, “Lena”, “Barbara”, “Tank”, “Boat”, “Goldhill”, “Peppers”, are used as the test images. We choose the packet size of 100 bytes and use the exponential PMF packet loss model. All programs were run on a PC with a Windows 2000 operating system, 2.8GHz Intel Pentium 4 CPU and 512M RAM. SPIHT is adopted as the codec for source coding and RS codes with 8 bits/symbol are used for channel coding. We choose 25 dB as the PSNR threshold for the minimum quality requirement. Any image transmission with a PSNR value less than the threshold is deemed as a failure.

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Table 3.4 and 3.5 show the performance comparison of different FEC protection schemes for transmitting the six test images over the packet-loss channel with a total bandwidth of 0.25 bpp and 0.5 bpp, respectively. The performance parameters include “Average Distortion”, “Time”, “ P_f ”, and “ b_n ”. “Average Distortion” is the expected distortion under the selected FEC solution. “Time” is the executing time to find the optimal FEC solutions. “ P_f ” denotes the probability of failure image transmission. “ b_n ” is the total amount of bytes allocated for source coding under the selected FEC solution.

We mainly compare the performance of four schemes: the optimal ELP scheme (ELP), the HLP scheme [38], our proposed L-ULP scheme and our proposed IL-ULP scheme. There are three layers in both L-ULP and IL-ULP. Comparing L-ULP with ELP, we can expect that the optimal ELP scheme will have worse distortion performance but with lower complexity. On the other hand, from the numerical results demonstrated in Section 3.1, we know that the HLP have better distortion performance but higher complexity than the L-ULP. Therefore, ELP and HLP should be lower and upper bounds in terms of distortion and complexity for our proposed L-ULP scheme.

As shown in Table 3.4 and 3.5, the distortion performance of L-ULP is always not worse than that of ELP, up to 37 in distortion reduction. Although the distortion performance of L-ULP is always not better than that of HLP, the difference is small, at most 7 in distortion. On the other hand, the executing time of L-ULP, in the order of $10^{-1} \sim 10^0$, is much faster than that of HLP, in the order of $10^2 \sim 10^3$. Moreover, it is surprising that with similar executing time to L-ULP, IL-ULP achieves better distortion performance than HLP at most of the cases, up to 3 in distortion reduction. The reason for this is that pre-interleaving improves the loss resilience in the source bit streams. IL-ULP tends to allocate more bits to the source coding since it has better loss resilience performance which allows to have weaker FEC protection. We previously pointed out in Section 3.1 that, when transmitting Barbara at 0.25 bpp, ELP may achieve the same

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Table 3.4: The performance comparison of different FEC protection schemes for transmitting the six test images over the packet loss channel model with the exponential PMF. The total bandwidth is 0.25 bpp and $p_l = 0.2$.

Image	Scheme	Average Distortion	Time(s)	$P_f(\%)$	b_n (bytes)
Lena	ELP	128.96	-	1.96	2300
	HLP	94.07	763.76	1.16	3277
	L-ULP	98.70	0.11	1.16	3274
	IL-ULP	92.80	0.12	1.16	3624
	L-ULP+	93.12	0.11	1.16	3274
	HLP+LS	95.15	0.17	1.05	3071
Barbara	ELP	499.38	-	10.57	4700
	HLP	499.38	1.55	10.57	4700
	L-ULP	499.38	0.17	10.57	4700
	IL-ULP	499.38	0.17	10.57	4700
	L-ULP+	499.38	0.17	10.57	4700
	HLP+LS	499.38	0.016	10.57	4700
Tank	ELP	137.18	-	4.13	3300
	HLP	125.17	458.69	2.12	4023
	L-ULP	128.79	0.1	1.96	3920
	IL-ULP	123.27	0.16	2.67	4630
	L-ULP+	124.43	0.1	1.96	3920
	HLP+LS	125.25	0.15	1.96	3959
Boat	ELP	218.92	-	2.67	2700
	HLP	197.46	126.49	2.48	3277
	L-ULP	200.27	0.08	2.48	3278
	IL-ULP	196.61	0.05	2.48	3401
	L-ULP+	196.98	0.08	2.48	3278
	HLP+LS	197.64	0.11	2.29	3182
Goldhill	ELP	173.20	-	1.96	2300
	HLP	137.98	590.3	1.16	3276
	L-ULP	142.63	0.1	1.16	3244
	IL-ULP	137.21	0.09	1.16	3304
	L-ULP+	137.60	0.1	1.16	3244
	HLP+LS	139.85	0.17	1.16	3105
Peppers	ELP	166.44	-	1.96	2300
	HLP	124.61	457.01	1.27	2949
	L-ULP	129.13	0.09	1.27	2957
	IL-ULP	122.99	0.08	1.27	3074
	L-ULP+	123.69	0.09	1.27	2957
	HLP+LS	125.39	0.16	1.27	2922

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Table 3.5: The performance comparison of different FEC protection schemes for transmitting the six test images over the packet loss channel model with the exponential PMF. The total bandwidth is 0.5 bpp and $p_l = 0.2$.

Image	Scheme	Average Distortion	Time(s)	$P_f(\%)$	b_n (bytes)
Lena	ELP	87.00	-	1.14	3300
	HLP	52.69	5904.1	0.67	6554
	L-ULP	57.56	2.05	0.67	6562
	IL-ULP	51.02	1.90	0.59	6485
	L-ULP+	51.33	2.05	0.67	6562
	HLP+LS	54.33	0.63	0.75	6473
Barbara	ELP	259.86	-	2.97	5600
	HLP	236.52	203.6	2.97	6553
	L-ULP	240.69	0.45	2.97	6552
	IL-ULP	237.43	0.27	3.08	6668
	L-ULP+	236.53	0.45	2.97	6552
	HLP+LS	239.01	0.3	3.20	6643
Tank	ELP	107.44	-	4.12	6500
	HLP	90.10	2467.8	1.50	8415
	L-ULP	95.09	1.68	1.50	8405
	IL-ULP	87.07	2.47	1.94	9828
	L-ULP+	89.06	1.68	1.50	8405
	HLP+LS	90.54	0.47	1.57	8278
Boat	ELP	159.99	-	2.18	4800
	HLP	117.88	2491.2	1.37	6554
	L-ULP	122.35	1.17	1.37	6562
	IL-ULP	116.65	0.82	1.44	7110
	L-ULP+	117.50	1.17	1.37	6562
	HLP+LS	117.90	0.48	1.37	6598
Goldhill	ELP	130.13	-	1.14	3300
	HLP	90.14	4908.7	0.71	6554
	L-ULP	94.45	2.08	0.71	6563
	IL-ULP	88.38	1.83	0.71	6976
	L-ULP+	89.10	2.08	0.71	6563
	HLP+LS	90.46	0.58	0.71	6523
Peppers	ELP	106.50	-	1.14	3300
	HLP	68.65	4826.9	0.75	6553
	L-ULP	75.52	1.89	0.75	6551
	IL-ULP	66.92	1.89	0.75	6551
	L-ULP+	68.46	1.89	0.75	6551
	HLP+LS	72.65	0.53	0.75	5898

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performance with HLP and L-ULP. Table 3.4 demonstrates this again. From this table, we can find clearly that actually all the four schemes achieve the same distortion performance.

Figures 3.9 and 3.10 show the distortion comparison between ELP, L-ULP, HLP and IL-ULP under different packet loss rates. It can be observed that the three unequal protection schemes, L-ULP, HLP and IL-ULP always outperform ELP, and the gain becomes more and more larger with the increase of packet loss rate. The performance of L-ULP is slightly worse than that of HLP, and the distortion difference is less than 10. The performance of IL-ULP is the best, even outperforming HLP.

We have also listed the performance of the “L-ULP+” scheme and the “HLP+LS” scheme in Table 3.4 and 3.5. The “L-ULP+” scheme is almost the same as IL-ULP except that in the “L-ULP+” we do not consider the pre-interleaving effect in the FEC allocation. In other words, the FEC allocation for “L-ULP+” is the same as that for L-ULP. The “HLP+LS” scheme is the scheme of applying Stankovic’s local search algorithm (“HLP+LS”) [39]. It can be seen that the distortion performance of the “L-ULP+” scheme is slightly worse than that of IL-ULP and always better than that of HLP at most of the cases, which further demonstrates the effectiveness of the pre-interleaving. We can also see that the executing time of our proposed IL-ULP scheme is slightly shorter than that of the “HLP+LS” scheme at 0.25 bpp but slightly longer at 0.5 bpp. Overall speaking, the complexity of our proposed IL-ULP scheme is comparable to that of the “HLP+LS” scheme.

Furthermore, in order to understand how robust the algorithms can be, we have shown in Figure 3.11 the performance comparison between HLP and ILULP when the channel condition is not as expected. We choose Lena as the test image and 0.05 as the anticipated packet loss rate. As we can see from Figure 3.11, ILULP can always obtain better distortion performance than HLP, especially when the packet loss rate is larger than 0.1. That means ILULP is more robust against different channel loss conditions.

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Figure 3.9: The distortion performance comparison between IL-ULP, HLP, L-ULP, and ELP at 0.5 bpp. Figures in the order of from top to bottom are the results for Lena, Barbara, and Tank, respectively.

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Figure 3.10: The distortion performance comparison between IL-ULP, HLP, L-ULP, and ELP at 0.5 bpp. Figures in the order of from top to bottom are the results for Boat, Goldhill, and Peppers, respectively.

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Figure 3.11: The distortion performance comparison between IL-ULP and HLP when the channel condition is not as anticipated. Expected packet loss rate is 0.05.

By observing the “ b_n ” results in Table 3.4 and 3.5, an interesting finding is that the total number of source data in any distortion optimal ULP solution is not less than that in the distortion optimal ELP solution. Combining this with the finding in [47], we can further conclude that the average FEC protection in the distortion optimal ULP solution should be bounded by $[f_r^*, f_d^*]$, where f_r^* and f_d^* are the rate optimal ELP solution and the distortion optimal ELP solution, respectively. This conclusion can be applied to speed up the search for $\bar{f}$ in Eqn. (3.3). However, theoretical work is needed to prove this finding.

3.4 Improved Layered Unequal Loss Protection for JPEG2000

3.4.1 Introduction

JPEG2000, which is fundamentally based on the discrete wavelet transform (DWT) and embedded block coding with optimal truncation (EBCOT) [58], is the latest image compression standard. It offers a host of features beyond the capabilities of conventional

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Figure 3.12: JPEG2000 fundamental building blocks diagram.

JPEG, including superior low bit rate coding performance, region of interest (ROI) coding, good error resilience, and more [59]. Figure 3.12 shows the fundamental building blocks of a typical JPEG2000 encoder [60]. In pre-processing, the input image is first partitioned into tiles, followed by level offset and component transformation. After the pre-processing, DWT is applied to generate spatial frequency sub-bands. Each sub-band is further divided into code-blocks. Then, all wavelet coefficients are subjected to uniform scalar quantization with dead-zone. The resulting quantization indices in each sub-band are entropy coded independently on each code-block. This coding is performed by context-dependent, binary, and arithmetic coding of bit-planes, from the most significant bit-plane to the least significant bit-plane. The compressed code-block codestream in each resolution are parted into precincts. A precinct is actually a collection of three spatially consistent rectangles (one from each sub-band at a given resolution level). For each precinct, the compressed data are organized into *JPEG2000 packets*. The JPEG2000 packet is the basic component of a JPEG2000 codestream. Note that JPEG2000 packet is different from network packets.

From above brief introduction, we can see that JPEG2000 is much more different from former image compression schemes such as JPEG and SPIHT. Therefore, how to efficiently transmit JPEG2000 codestreams is a challenging research topic.

ULP has been applied to JPEG2000 to combat channel noise including Internet packet loss and bit errors in wireless links [61, 62, 63]. In [61, 62], the authors demonstrated that earlier quality layers should be assigned more protection, while they did not clarify how to optimize the FEC selection. In [63], V. Sanchez et al. proposed a method named

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adaptive unequal channel protection (AUCP) for optimal FEC selection. However, the AUCP scheme has two shortcomings. First, it simply neglects the contribution of code-blocks that are not included in a JPEG2000 packet for the first time, and no FEC will be provided for network packets that do not contain any new code-block data. This results in noticeable inaccuracy in the computation of the channel impairment effect. Second, in AUCP, the distortion analysis is performed at the JPEG2000 packet level instead of the desired network packet level.

3.4.2 End-to-end Distortion Analysis

Although IL-ULP works well on SPIHT bit streams, it may not be suitable for JPEG2000 in that, upon detecting an error, JPEG2000 does not have to stop decoding due to its intrinsic error resilience properties. For example, since in JPEG2000 each code-block in a sub-band is independently coded, bit errors in the codestream of any one code-block will not affect other code-blocks. JPEG2000 also provides modes variations such as *SEGMARK* in order to enhance error resilience [17]. When *SEGMARK* is used, the special four-symbol code, “1010,” is inserted at the end of each cleanup pass. A single error in any of the preceding coding passes is likely to corrupt at least one of the four symbols. Upon detecting the corruption, an error resilient decoder should attempt to discard those coding passes which it suspects may contain errors.

Therefore, in this research, we study the performance of applying the IL-ULP scheme to JPEG2000 encoded images. In particular, we take the characteristics of JPEG2000 into consideration and develop a model for end-to-end distortion analysis. Unlike the AUCP scheme, our end-to-end distortion analysis is performed in the network packet level and we consider the contributions of all the code-blocks.

Assuming all the N packets share the same packet loss probability, we can approxi-

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Figure 3.13: The IL-ULP scheme structure for JPEG2000. The j -th packet of the first layer is shown as an example of $p_{i,j}$.

mate the expected distortion at the receiver end as

$$D = D_0 + \sum_{i=1}^n \sum_{k=f_i+1}^{f_{i-1}} (P(N, k) \frac{k}{N} \Delta D_i) \quad (3.13)$$

where D_0 is the minimum distortion, n is the number of layers, $P(N, k)$ is the probability of losing k packets out of the total N packets, ΔD_i is the distortion loss for i -th layer, and $\frac{k}{N} \Delta D_i$ means the average proportional loss due to losing k packets in the i -th layer. ΔD_i is further derived as

$$\Delta D_i = \sum_{j=1}^{N-f_i} \Delta D_{i,j} \quad (3.14)$$

where $\Delta D_{i,j}$ is the distortion loss contributed by the part of data $p_{i,j}$, which locates in the i -th layer and belongs to the j -th network packet p_j , as shown in Figure 3.13.

If any part of the main header of the JPEG2000 codestream is corrupted, the decoding will fail. Thus, when any $p_{i,j}$ in the i -th layer contains information from the main header, the corresponding distortion loss should take the maximum value, i.e.

$$\Delta D_i = \sum_{s=1}^S \sum_{b=1}^{B_s} \sum_{cp=1}^{CP_b} G_{cp} \quad (3.15)$$

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where S is the total number of sub-bands, B_s is the total number of code-blocks in the s -th sub-band, CP_b is the total number of the included coding passes for the code-block b , and G_{cp} is the distortion loss corresponding to coding pass cp . For all other cases, $\Delta D_{i,j}$ depends on the number of new code-blocks included in $p_{i,j}$, i.e.

$$\Delta D_{i,j} = \sum_{b=1}^{B_{i,j}} \sum_{cp=CP_{b,i,j}}^{CP_b} G_{cp} \cdot \psi(i, j, b) \quad (3.16)$$

where $B_{i,j}$ and $CP_{b,i,j}$ are the total number of code-blocks and the first coding pass of code-block b in $p_{i,j}$, respectively, and

$$\psi(i, j, b) = \begin{cases} 1 & \text{if the code-block } b \text{ is first-time included} \\ & \text{in the } i\text{-th layer} \\ 0 & \text{otherwise} \end{cases}$$

Note that Eqn. 3.16 is the key difference between our proposed end-to-end distortion model and the model in the AUCP algorithm. Specifically, the AUCP only considers the distortion contribution from the brand new code-blocks in a JPEG2000 packet, while the definition of new code-blocks in our model is with respect to a particular layer. For example, we consider the distortion contribution from the b -th code-block in $p_{i,j}$ as long as it is first-time included in the i -th layer, no matter whether it has been included in the previous layers or not. This is reasonable because ΔD_i is defined for the case that the first uncorrected loss occurs in the i -th layer. Thus, even if a code-block has been included in the previous layers, all the corresponding pre-included coding passes should be correct with respect to ΔD_i , and we only need to consider the distortion caused by losing the first coding pass of the code-block included in the i -th layer. Through the above discussion, we can see that our end-to-end distortion model is more accurate than that in the AUCP algorithm.

Both G_{cp} and the related coding pass length, L_{cp} , can be obtained readily from JPEG2000 implementation software such as Kakadu [64]. All other parameters including D_0 , CP_b , $B_{i,j}$, and $CP_{b,i,j}$ can also be computed during the encoding process. Due to

the complexity of the JPEG2000 and the proposed end-to-end distortion analysis, a fast error control algorithm is desired. Thus, in this paper, we directly match the JPEG2000 quality layers into the IL-ULP layers, and we adopt the fast local search algorithm in [39] to find the optimal FEC allocation.

3.4.3 Experimental Results

In this section, we compare the performance between the L-ULP scheme and the IL-ULP scheme to see whether the IL-ULP can achieve better performance or not for JPEG2000 codestreams. The five gray scale 512×512 images, “Lena”, “Tank”, “Peppers”, “Gold-hill”, “Boat”, are used as test images. We choose the packet size of 100 bytes. We adopt the simplified Gilbert-Elliott channel (GEC) [54], a two-state Markov model, as the packet loss model. The GEC model has two states: Good state and Bad State. In Good state there is no packet loss while in Bad state packets are always lost. The testing average packet loss rate (p_l) is from 5% to 20% and the average burst length is fixed to 5 packets. RS codes with 8 bits/symbol are used for channel coding. We select 25 dB as a PSNR threshold for the minimum quality requirement. We fix the total bandwidth to 0.25 bpp, which corresponds to a total number of packets of 82. We fix the number of JPEG2000 quality layers, which is the same as the IL-ULP layers, as 5. All experimental results are obtained over 1000 independent simulations.

Figure 3.14 and 3.15 show the distortion performance of transmitting different images under different packet loss rates. It can be seen that IL-ULP always outperforms the L-ULP. This is not surprising. By carefully checking the FEC allocation, we found the main reason lies in the main header protection for a JPEG2000 codestream. For the L-ULP scheme, the main header typically spans over the entire $N - f_1$ packets in the first layer, as shown in Figure 3.16(a). Thus, any packet loss in the first layer will cause the decoding failure. On the other hand, for the IL-ULP case, which is shown in Figure 3.16(b), only

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Figure 3.14: The performance comparison between the IL-ULP and L-ULP for transmitting Lena, Tank, and Peppers under different packet loss rates.

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Figure 3.15: The performance comparison between the IL-ULP and L-ULP for transmitting Goldhill and Boat under different packet loss rates.

Figure 3.16: Arrangement of the main header in the first layer of the (a) L-ULP scheme and (b) IL-ULP scheme.

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the earliest several packets in the first layer contain the main header information. As a result, the L-ULP scheme tends to allocate more protection to the first quality layer than the IL-ULP does. When their first quality layers have the same source bit rates, the L-ULP scheme will leave smaller space for the remaining several layers than the IL-ULP. Therefore, compared with the IL-ULP, the L-ULP is allocated fewer source bytes.

3.5 Summary

In this chapter, we have presented the L-ULP scheme which considers the minimum quality requirement and solves the high complexity problem of ULP. The L-ULP can be executed in a much faster way than the HLP while the distortion performance of L-ULP is worse than that of HLP. By integrating L-ULP with the pre-interleaving, we have further proposed the IL-ULP scheme, which achieves better distortion performance than HLP while the complexity is as low as that of L-ULP. We have also applied IL-ULP to transmission of JPEG2000 codestreams, and experimental results show that the IL-ULP scheme performs very well. Therefore, the proposed IL-ULP scheme is very suitable for practical fast image transmission over packet loss networks.

Chapter 4

Conclusion and Future Work

4.1 Conclusion

This thesis has presented my two-year study on FEC-based joint source-channel coding for progressive image transmission over packet loss channels. In particular, we have studied the characteristics of packet visual communications, and have reviewed existing error control techniques for combating the packet loss problem, including network error control techniques, error resilient techniques and joint source-channel coding. Our objective is to study, design, and implement efficient and reliable FEC-based JSCC image transmission schemes.

By noticing that most existing ULP schemes do not consider the minimum quality requirement and usually have high computation complexity, we have developed the L-ULP scheme for progressive image transmission over packet loss channels. In the proposed L-ULP, we have solved the optimal FEC selection problem by gradient search instead of exhaustive iterative search, which greatly reduces the system complexity. We have further extended the L-ULP scheme to the case with delay constraints. In particular, we have proposed a two-step approach to perform the FEC selection among individual packet groups and also within each packet group. Moreover, we have also integrated the L-ULP with the pre-interleaving to further improve the performance. The combined system is able to delay the occurrence of the first unrecoverable loss in the progressive

source bit stream while still keeping the original priorities among different layers. Experimental results have demonstrated that the proposed IL-ULP framework can satisfy the low-complexity and minimum quality requirements with almost no performance loss. Finally, we have applied IL-ULP to transmission of JPEG2000 encoded images in order to understand if pre-interleaving can still function or not for JPEG2000. We have conducted the end-to-end distortion analysis by taking into account the effect of the lost channel packets on the distortion of the reconstructed image. Experimental results demonstrate that IL-ULP can always outperform L-ULP.

4.2 Future Work

4.2.1 Application of Pre-Interleaving to Other Codecs

Traditional interleaving is only designed for improving channel coding performance and does not take the characteristics of source coding into consideration. When there are residual loss/errors remaining after the channel decoding, the residue loss/error patterns may not be preferred by the source decoding. Therefore, if the interleaving design can consider the characteristics of source coding, i.e., generate the residue loss/error pattern that matches the characteristics of source coding, the interleaving can also improve the source coding performance. Our proposed IL-ULP scheme has demonstrated this point.

Applying the pre-interleaving to the SPIHT codec is quite successful. Experimental results show that application of pre-interleaving to JPEG2000 is also very promising. This is because SPIHT and JPEG2000 are progressive image codecs, in which the earlier portions of the progressive bit stream are always more important than the later parts and the bit stream is allowed to be truncated in any position. We could foresee the performance improvement for other progressive image and video codecs such as MPEG-4 fine granularity scalability (FGS) [65] by using pre-interleaving.

The pre-interleaving could also be useful for those non-progressive image and video codecs. For example, for MPEG-4 with re-synchronization words enabled, a slice, i.e., a data segment starting with a re-synchronization word, is the smallest independent unit in the compressed video bit stream. The MPEG-4 decoding performance is very much determined by the slice loss ratio. At the same data loss/error rate, it is easier for the MPEG-4 decoder to handle burst residue loss/errors within one or several consecutive slices than to handle random residue loss/errors in many slices. Therefore, MPEG-4 prefers burst residue loss/error patterns than random loss/error patterns. This is exactly what the pre-interleaving can provide, i.e., reserving the burst residue loss/errors instead of distributing them in the source bit stream.

4.2.2 ROI-Based Image/Video Coding and Transmission

Region of interest (ROI) is a novel feature of JPEG2000 [66]. Basically, a ROI is a part of an image that is encoded earlier and thus always has better quality than the rest of the image. It is especially useful for applications such as X-ray images and image or video retrieval where an image region is particularly interesting.

For ROI-based image coding, a fundamental problem is how to allocate bits between the ROI and the background. If there are multiple ROIs in one image, we need to consider bit allocation among multiple ROIs and the background. When transmitting ROI-based bit streams over packet loss channels, we need to study how to provide protection according to the inherent priorities. A few researchers have done some work in this area [67, 68, 69]. Although we have proposed the novel L-ULP scheme for fast progressive image transmission over packet loss channels, it will be interesting to study how to apply the L-ULP to ROI-based bit streams. In particular, it is essential to jointly consider the FEC allocation and the bit allocation among the ROIs and the background.

In addition, ROI-based media adaptation is another interesting topic that deserves further research effort. It is well-known that multimedia communications tend to have

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more and more heterogeneities. For example, different users have different preference, may access resource-rich Internet through different access networks, and use different devices. The visual information has to adapt to user requirements, network fluctuation, and device capability. Traditional approaches are to change the quantization parameters, frame size, and frame rate of original image sequences. ROI-based image and video coding provides a new approach, i.e., media adaptation can be realized by using different coding parameters for different regions.

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