Asynchronous and Reliable Multimedia Multicast with Heterogeneous QoS Constraints

Wei Sheng, Wai-Yip Chan, Steven D. Blostein and Yu Cao
Queen's University, Kingston, Canada

***Abstract*— We present an asynchronous and reliable multicast framework for a scalable multimedia system. An unequal error protection (UEP) transmission scheme employing a layered packetization structure and rateless codes is proposed. With this scheme, asynchronous data reception and guaranteed quality-of-service (QoS) can be achieved at the same time. Furthermore, a novel allocation algorithm is developed within the layered multicast framework, which can minimize the system cost in terms of the number of transmitted packets. Simulation results show that the proposed rateless-codes-based transmission scheme works well for asynchronous multicasting to heterogeneous receivers, and can be applied to a wide range of multimedia applications.**

I. Introduction

Future QoS-aware communications systems, e.g., Beyond-Third-Generation (B3G), require more than just reliable multicast for scalable multimedia contents. Receivers will wish to receive the multimedia data asynchronously. The concept of asynchronous reception here is akin to the digital fountain concept in [1], i.e., receivers can access the system ('dip' into the fountain) any time within a certain time period. Furthermore, their access speeds, channel error statistics and QoS requirements may vary widely.

A recent approach toward QoS-aware multicast is receiver-driven multicast (RDM) over noisy channels [2], which allows each receiver to subscribe to different multicast groups until a target QoS constraint is satisfied. However, this approach is designed for simultaneous reception, i.e., all the receivers start their reception simultaneously with the transmission of the first packet and remain tuned into the packet transmission, until all receivers' QoS requirements are satisfied, or when the transmission ends. Although in principle, late tune-in receivers can subscribe to multiple parity layers to recover missing packets, the server needs to transmit an amount of data through different layers to cater to the most demanding (worst case) channel and receiver combination, which potentially is wasteful of precious system capacity, especially in wireless communications.

In the literature, some existing multiple description coding (MDC) schemes, e.g., priority encoding transmission (PET) [3], and layered unequal-error-protection (UEP) [4] [5] [6], can provide the digital fountain property of asynchronous reception. However, these schemes can only provide best effort service, so that the quality of the reconstructed signal may vary widely, making it a less satisfying user experience than QoS-enabled multicast in general [7]. Therefore, these schemes cannot be applied to commercial multimedia multicast applications, such as video-on-demand (VoD). Furthermore, most of the above best-effort schemes, e.g., [5], are designed for unicasting over a known channel, and as a result, are not appropriate for multicast.

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In this paper, we consider a general multicast scenario in which the receivers have distinct channel error statistics. A PET-like layered packetization transmission scheme combined with rateless codes is proposed. This scheme has the digital fountain property, and as a result, allows asynchronous reception. Also, with rateless codes, the number of transmitted packets can be adaptively adjusted until all the receivers' QoS are satisfied, so that the quality guarantees can be quite precise for a wide range of channel conditions.

We note that some recent related work, which we call UEP rateless codes, either non-uniformly chooses message bits [8] or employs Expanding Window Fountain (EWF) codes [9] to design novel rateless codes for UEP. In contrast, in this paper, we aim to design a UEP scheme, such that a variety of channel codes can be employed. This provides more system design flexibility and simplicity. Another difference between our proposed scheme and the UEP rateless codes in [8] and [9] is that our scheme is designed to satisfy heterogenous QoS constraints, while the designs in [8] [9] do not consider guaranteed user experience.

A complementary work on optimal QoS-aware rateless code allocation scheme is discussed in [10]. The difference between our paper and the work in [10] is that in [10], the numbers of source bits allocated to different layers are assumed to be known, and the focus is to optimize the interleaving of rateless codes across different source layers. In this paper, the main objective is to allocate source bits among different layers in order to optimize the joint source-channel coding for a given rateless code.

The rest of this paper is organized as follows. In Section II, we describe the system setup. A novel transmission scheme employing rateless codes is then proposed in Section III. Simulation results are presented in Section IV.

II. System Description

We consider a multicast system where a scalable coded image/video stream is transmitted from a server to a number of asynchronous and heterogenous receivers. In the system, there are J classes of receivers, and the number of receivers in class j is denoted by M_j. Each class is characterized by a QoS requirement in terms of a target peak signal-to-noise ratio (PSNR) of the reconstructed signal, which is denoted by γ_j for class j, $j = 1,..,J$. In the proposed scheme, PSNR can be straightforwardly replaced by other measures of source reconstruction quality.

At the transmitter/server, the embedded source bit stream is loaded onto a sequence of packets by a PET-like layered packetization structure, as illustrated in Figure 1. PET [3] is an algorithm that assigns channel codes to source symbols according to priorities specified by the receivers. Our scheme employs this layered structure so that source symbols can be protected unequally according to their significance. As shown in Figure 1, the number of layers, denoted by L, is equal to the packet size, and the number of allocated source symbols for layer i is denoted by K_i, where $i = 1, .., L$. A lower-indexed layer contains more significant source symbols than a higher-indexed layer. Unlike traditional PET, in which classic forward-error-correction (FEC) codes are employed and the number of transmitted packets is decided before transmission, our layered packetization structure employs the benefits provided by rateless codes, so that the total number of transmitted packets can be adjusted online to suit current conditions. There can be as many packets generated as needed until all the receivers' QoS requirements are satisfied, or a cutoff deadline is reached. This cutoff deadline depends on system and application constraints, and is assumed to be given by the system.

In this paper, we investigate a system in which the actual erasure rate for a receiver is not known precisely, but the set of all possible channel erasure rates as well as their probabilities are known at the transmitter. We consider a network with memoryless and independent packet erasure channels from the server to the receivers. Each receiver accesses the server under different channel qualities. The channel erasure rate for receiver i in class j, denoted by $\rho_{j,i}$, where $i = 1, .., M_j$ and $j = 1, .., J$, can be chosen from a set of possible channel erasure rates, $\mathbf{h}_j = \{h_{j,1}, .., h_{j,b_j}\}$, where b_j denotes the size of this set, with probabilities $\mathbf{p}_j = \{p_{j,1}, .., p_{j,b_j}\}$. Once the channel erasure rate is chosen, it remains unchanged during the whole transmission period.

Each receiver keeps receiving packets until its target QoS, in terms of PSNR, is satisfied, and then sends a one-bit acknowledgement (ACK) signal to the server. We remark that the receivers cannot evaluate PSNRs since they do not have access to the original source data. However, the encoder/transmitter can embed PSNR values obtained from the operational rate-distortion function at various breakpoints in the bit stream. The decoders at each receiver can determine the source reconstruction quality from these embedded PSNR values.

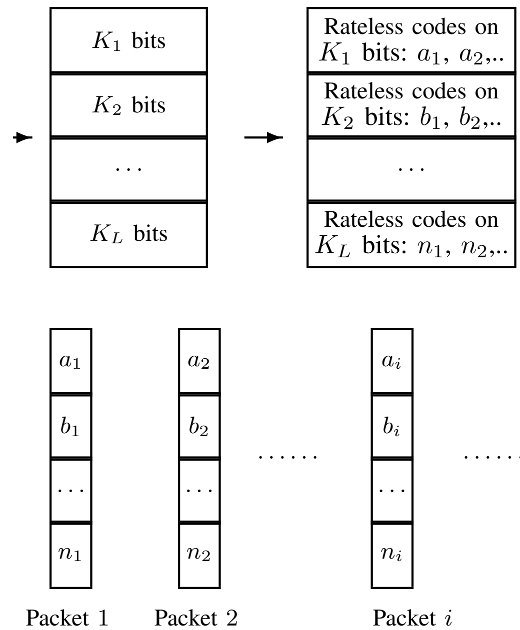

Fig. 1. Layered packetization structure combined with rateless codes.

III. Proposed transmission scheme based on rateless codes

In the previous section, we presented the packetization structure based on PET and rateless codes, but the scheme that allocates $K_1, .., K_L$ was not specified. In the following, we formulate the allocation problem as a constrained optimization problem, and then provide an allocation algorithm to solve the problem.

A. QoS constraints

Without loss of generality, the J classes are ordered with increasing target PSNR, i.e., $\gamma_1 \leq \gamma_2 \leq .. \leq \gamma_J$. In the course of compressing a source, its operational rate-distortion function (e.g., in terms of PSNR versus the number of decoded source bits) can be obtained, denoted by $f(.)$. Therefore, to achieve a target PSNR γ_j, where $j = 1, .., J$, it is required that at least R_j source bits can be successfully decoded,

$$R_j = f^{-1}(\gamma_j) \tag{1}$$

and f^{-1} denotes the inverse of the operational rate-distortion function.

With an embedded source bit stream, the less important bits are not usable if the preceding more important bits are not correctly received. Therefore, the more important part of the source stream should be better protected. Since layer i is useless if layers $1, \ldots, i-1$ are not decoded successfully, layer $i-1$ should have no less error protection than layer i, i.e., the channel code rate of layer $i-1$ is no greater than that of layer i. This is accomplished by setting

$$K_1 \leq K_2 \leq .. \leq K_{L-1} \leq K_L. \tag{2}$$

The proposed scheme allows each user to keep receiving packets, and then decode the source symbols layer by layer. After one layer is successfully decoded, the user evaluates its current PSNR, and then decides if it is necessary to receive more packets. Therefore, for class j, it is the number of successfully decoded layers, denoted by d_j, which decides the received PSNR, denoted by $\text{PSNR}_j(K_1, .., K_L)$. To ensure a target PSNR, d_j should be chosen to satisfy the following condition

$$\begin{aligned} \text{PSNR}_j(K_1, .., K_L) &= f(\sum_{m=1}^{d_j} K_m) \\ &\geq \gamma_j. \end{aligned} \tag{3}$$

B. Cost criterion

Capacity in a multicast system is a valuable resource, especially in wireless environments. To sufficiently utilize system resources, it is desirable to transmit as few packets as possible. For synchronous systems, this is equivalent to minimizing the total number of transmitted packets. For asynchronous systems, however, the total number of transmitted packets depends on the worst channel and late-tune-in receiver combination, and as a result, it is more reasonable to minimize the average number of transmitted packets (including correctly and incorrectly received packets) required for each receiver.

For an arbitrary receiver, e.g., receiver i in class j, where $i = 1, .., M_j$ and $j = 1, .., J$, the number of transmitted packets, denoted by $N^{j,i}$, can be obtained by

$$N^{j,i} = N_C^{j,i} + N_{INC}^{j,i}$$

where $N_C^{j,i}$ denotes the number of correctly received packets required to decode R_j source bits, and $N_{INC}^{j,i}$ denotes the number of incorrectly received packets before $N_C^{j,i}$ packets are successfully collected at the receiver.

With known $K_1, .., K_L$, the number of decoded layers needed for class j, d_j, can be obtained from (3). To successfully decode layer d_j (e.g., to limit the decoding outage probability in this layer within a tolerable level), the receiver needs to collect an average of $\lceil K_{d_j}(1+\epsilon_{i,d_j})\rceil$ successfully received packets, where $\lceil x \rceil$ denotes the smallest integer not less than x, and ϵ_{i,d_j} denotes the average overhead rate needed to recover the K_{d_j} source bits at receiver i, which depends on the rateless code design for these K_{d_j} source bits, the target decoding outage probability as well as the channel condition from the server to receiver i.

Due to the layered packetization structure with $K_1 \leq K_2.. \leq K_L$, it is obvious that if $\lceil K_{d_j}(1+\epsilon_{i,d_j})\rceil$ packets are correctly received, then all the layers prior to layer d_j can be decoded successfully. Therefore, the number of correctly received packets, $N_C^{j,i}$ can be expressed as

$$N_C^{j,i} = K_{d_j}(1+\epsilon_{i,d_j}), \quad (4)$$

where we have removed the integer constraint for convenience.

The cost for class j, in terms of the average number of transmitted packets in class j, denoted as N_j, can be derived as

$$\begin{aligned} N_j &= \frac{1}{M_j}\sum_{i=1}^{M_j} N^{j,i} \\ &= \frac{1}{M_j}\sum_{i=1}^{M_j} K_{d_j}(1+\epsilon_{i,d_j}) + \frac{1}{M_j}\sum_{i=1}^{M_j} N_{INC}^{j,i} \end{aligned}$$

where M_j denotes the number of receivers in class j.

If M_j is moderately large, e.g., $M_j \geq 30$, by the Weak Law of Large Numbers (WLLN), the above equation can be approximately by

$$N_j \approx K_{d_j}(1+\epsilon_{d_j}) + E[N_{INC}^{j,i}] \quad (5)$$

where ϵ_{d_j} denotes the average overhead rate for layer d_j over all receivers in class j; ϵ_{d_j} depends on the rateless code design and is assumed known to the system.

For a given channel erasure rate, $\rho_{j,i}$, for receiver i, packet transmissions to receiver i can be modeled as independent Bernoulli trials, and as a result, it can be shown that $N_{INC}^{j,i}$ in (5) has a negative Binomial distribution with mean $N_C^{j,i}\frac{\rho_{j,i}}{1-\rho_{j,i}}$ [11]. Therefore, $E[N_{INC}^{j,i}]$ in (5) can be obtained as

$$\begin{aligned} E[N_{INC}^{j,i}] &= E_{\rho_{j,i}}[E[N_{INC}^{j,i}|\rho_{j,i}]] \\ &= E_{\rho_{j,i}}[N_C^{j,i}\frac{\rho_{j,i}}{1-\rho_{j,i}}] \\ &= E[N_C^{j,i}]E_{\rho_{j,i}}[\frac{\rho_{j,i}}{1-\rho_{j,i}}] \quad (6) \\ &= K_{d_j}(1+\epsilon_{d_j})\sum_{m=1}^{b_j} p_{j,m}\frac{h_{j,m}}{1-h_{j,m}} \quad (7) \end{aligned}$$

where $h_{j,m}$ and $p_{j,m}$, $m = 1, .., b_j$, denote a possible erasure rate and its corresponding selection probability, respectively, and (6) is obtained since $N_C^{j,i}$ can be considered independent of $\rho_{j,i}$.

Therefore, the cost for class j expressed in (5) becomes

$$N_j = K_{d_j}(1+\epsilon_{d_j})\left(1+\sum_{m=1}^{b_j} p_{j,m}\frac{h_{j,m}}{1-h_{j,m}}\right).$$

The overall cost, in terms of the average number of transmitted packets among all classes, denoted by N_{av}, is defined as

$$\begin{aligned} N_{av} &= \sum_{j=1}^{J}\beta_j N_j \\ &= \sum_{j=1}^{J}\beta_j K_{d_j}(1+\epsilon_{d_j})\left(1+\sum_{m=1}^{b_j} p_{j,m}\frac{h_{j,m}}{1-h_{j,m}}\right) \end{aligned} \quad (8)$$

where $\beta_j > 0$ is a weighting factor set to the fractional significance of class j, and without loss of generality we assume $\sum_{j=1}^{J}\beta_j = 1$.

C. Allocation problem formulation

With the above discussion on QoS and cost criterion, the allocation problem can be formulated as follows:

$$\min_{K_1,..,K_L} N_{av} \quad (9)$$

subject to

$$f(\sum_{m=1}^{d_j} K_m) \geq \gamma_j, \; j = 1, .., J, \quad (10)$$

$$K_1 \leq K_2 ... \leq K_L. \quad (11)$$

D. Proposed allocation algorithm

To solve the optimization problem in (9)-(11), we propose a novel allocation algorithm in Table I. In this algorithm, to minimize the cost, we choose the total number of source bits K as the number of required source bits to guarantee the most demanding QoS. Also, it is obvious that setting $d_J = L$ sufficiently utilizes the packetization structure and thus leads to a lower N_{av} in (8).

The idea behind this allocation algorithm can be summarized as follows:

- Let S be the set of all possible allocation schemes.
- For each allocation scheme $s \in S$, the decoding-layer-set is defined as $d_s = [d_{s,1}, .., d_{s,J}]$, where $d_{s,j}$ denotes the required number of successfully decoded layers needed to satisfy class j's QoS, where $j = 1, .., J$, and $d_{s,i} \leq d_{s,j}$ for $i < j$.
- All the allocation schemes in S are then grouped into different subsets such that all the allocation schemes in a subset have the same values for the decoding-layer-set. The decoding-layer-set for subset g, where g denotes the subset index, is denoted by d^g.
- For each subset g, we choose the allocation scheme which equally (or approximately equally) allocates $R_j - R_{j-1}$ source bits from layers $d_{j-1}+1$ to d_j. This allocation

scheme and its corresponding cost are denoted by $\mathbf{k}_{opt,g}$ and C_g, respectively.

- Minimization is accomplished by checking the cost C_g for each subset, and the optimal scheme, denoted by $\mathbf{k}_{opt}$, can thus be obtained as

$$g_{opt} = \arg\min_g C_g$$
$$\mathbf{k}_{opt} = \mathbf{k}_{opt,g_{opt}}.$$

The above allocation scheme needs to search the cost for each possible decoding-layer-set. With $d_J = L$, and $1 \leq d_j \leq L$ for $j \leq J$, there are approximately $L^{J-1}/2$ possible decoding-layer-sets, corresponding to $L^{J-1}/2$ subsets. For each subset, the cost in (8) is computed, which requires J summations. Therefore, the proposed algorithm has a complexity of $JL^{J-1}/2$ summations, which can be manageable for practical systems. For example, with a packet size of $L = 1250$, and three classes of receivers, the proposed allocation algorithm needs to compute roughly 2.3 million summations.

It is obvious that if an optimal allocation scheme can be found within each subset, then our proposed allocation algorithm provides a globally optimal solution. For $J = 2$, we have proved mathematically that our proposed allocation scheme is globally optimal. Although for $J > 2$, it is still unclear if the proposed scheme is optimal, this scheme provides a means to obtain a near-optimal solution.

E. Remarks

In the above, we have designed a novel allocation scheme, which achieves all the receivers' QoS requirements with minimum cost. This is based on the assumption that ultimately each receiver can reach its target QoS. For a system with a hard limit on the total number of transmitted packets, some late-tune-in receivers or receivers with poor channel conditions may not achieve their target QoS when the cutoff deadline is reached. In this case, an optimal transmission scheme is harder to derive due to asynchronous reception. However, the transmission scheme proposed in this paper provides a suboptimal solution as well as some insights for this more complicated multicasting scenario.

IV. Simulation

A. System Setup

We use image transmission as a vehicle for testing the proposed scheme. Consider a standard 512×512 Lenna image compressed with the set partitioning in hierarchical trees (SPIHT) algorithm [12] at a rate of 0.2 bits/pixel. There are $L = 47$ layers in the packetization structure. Two classes are considered in the system, with target QoS $\gamma_1 = 27$ dB and $\gamma_2 = 30$ dB. From the operational rate-distortion curve of this image, the minimum number of the source bits needed to provide the QoS targets are $R_1 = 11072$ and $R_2 = 24728$, respectively.

For each class, there are a number of receivers, and the channel erasure rate for each receiver can be chosen from a set of possible channel erasure rates, $\mathbf{h}_1 = \mathbf{h}_2 = \{0.01, 0.2\}$, with corresponding probabilities $\mathbf{p}_1 = \mathbf{p}_2 = \{0.8, 0.2\}$. The weighting factors β_j are given by the system to indicate the fractional significance of class j, where $j = 1, 2$.

B. Allocation schemes

The allocation schemes for different values of weighting factor β_1 are shown in Figure 2. Note that $\beta_2 = 1 - \beta_1$. It is observed that if the weighting factor of class 1 is less significant, e.g., $\beta_j \leq 0.4$, the allocation scheme becomes the trivial equal-error-protection (EEP) scheme, which actually is the optimal solution if there is only one class with the higher target PSNR in the system. If β_1 is increased to $\beta_1 = 0.6$, we need to distribute the source bits needed to achieve class 1's target PSNR, γ_1, over more layers, so that the number of correctly received packets needed to achieve γ_1 can be reduced. If β_1 is further increased to 0.8, the bits are spread over even more layers, and in the extreme case, when $\beta_1 = 1$, (i.e, only class 1 receivers can contribute to the cost), the $R_2 - R_1$ source bits in the least significant layers are useless, and as a result, can be removed. In this case, the optimal allocation scheme is to allocate the source bits required by class 1 equally among all layers.

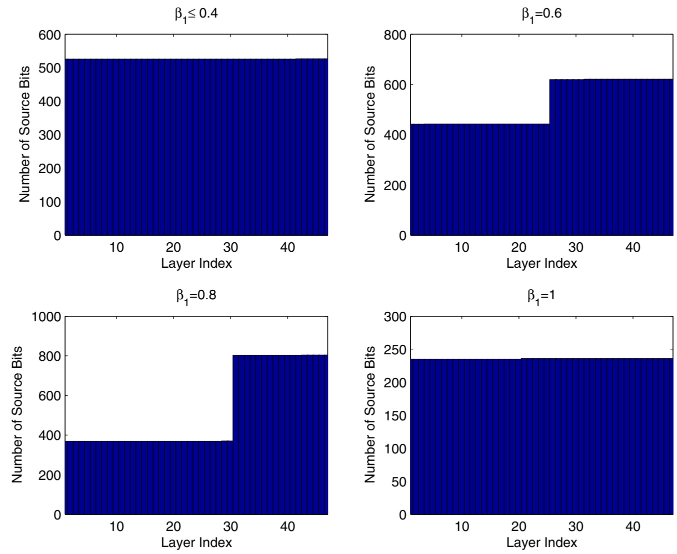

Fig. 2. Allocation of source bits over L layers for various values of class 1 weighting factor β_1. Note the subplots have different vertical scales.

C. Performance of the proposed allocation scheme

We have derived the allocation schemes for different weighting factors. Next we apply these allocation schemes to a multicasting system. On the transmitter side, an LT encoder with a robust soliton degree distribution with parameters $c = 0.03$ and $\delta = 0.5$ [1] is employed at each layer.

We now measure the cost for each class, N_j, where $j = 1, 2$, as well as the overall cost, N_{av}. Measurements are taken at an arbitrary class 1 receiver and an arbitrary class 2 receiver. With measured cost N_j, we can compute the overall cost N_{av} from (8). These results are shown in Table II, which are averaged over 100 realizations. From this table, it can be observed that an advantage of our proposed scheme is to appropriately trade off between the heterogenous classes. For example, with $\beta_1 = 0.3$, the optimal allocation scheme to achieve the minimum N_{av} is to equally allocate all source bits among L layers. When the weighting factor of class 1 (lower QoS class) is increased to $\beta_1 = 0.8$, the proposed scheme sacrifices the cost of class 2 to reduce the cost for class 1. For example, when β_1 is increased from 0.3 to 0.8, the cost of class 2, N_2, is increased from 713 to 1067, in order to reduce the cost of class 1, N_1, from 712 to 525. Note that as the costs change, so are the reception delays, since the average reception delay for class j is proportional to N_j.

TABLE I

A NOVEL ALLOCATION ALGORITHM.

```
Initialization: Best = 10000 and R_0 = 0;
Derive source bit rate R_j from (1);
foreach d_1 = 1 : 1 : L
  foreach d_2 = d_1 : 1 : L
      ....
      foreach d_{J-1} = d_{J-2} : 1 : L

          if d_1 = L;
            allocate R_J bits equally
            among L layers; break;
          end;
          if d_1 = d_2 = .. = d_l, where 1 < l <= L;
            allocate R_l bits equally among
            d_l layers; Goto l + 1;
          end;
1:      foreach i <= d_1;
           allocate R_1 bits equally, or approximately
           equally among d_1 layers;
        end ;

          if d_2 = L;
            allocate R_J - R_1 bits equally
            among L - d_1 layers; break;
          end;
          if d_2 = d_3 = .. = d_l, where 2 < l <= L;
            allocate R_l - R_1 bits equally
            among d_l - d_1 layers; Goto l + 1;
          end;
2:      foreach d_1 < i <= d_2;
            allocate R_2 - R_1 bits equally
            among d_l - d_1 layers;
        end ;

          . . .
          if d_{J-1} = L;
            allocate R_J - R_{J-2} bits equally
            among L - d_{J-2} layers; break;
          end;
J:      foreach d_{J-1} < i <= d_J;
           allocate R_J - R_{J-1} bits equally
           among d_J - d_{J-1} layers;
        end ;

        if k_1 <= k_2.. <= k_L;
           derive Cost according to (8);
           if Cost <= Best
             Best = Cost;
             Optimal allocation scheme is set to
             k_opt = [k_1, .., k_L];
           endif;
        endif;
      end foreach d_{J-1}

      ...
  end foreach d_2
end foreach d_1
```

Fig. 3. The reconstructed Lenna images at an arbitrary class 2 receiver.

TABLE II

COST COMPARISONS FOR DIFFERENT ALLOCATION SCHEMES, WHERE N_j, $j = 1, 2$, AND N_{av} DENOTE THE COST FOR CLASS j AND THE OVERALL COST AMONG ALL CLASSES, RESPECTIVELY.

$[\beta_1, \beta_2]$	N_1	N_2	N_{av}
[0.3 0.7]	712	713	713
[0.5 0.5]	659	765	712
[0.8 0.2]	525	1067	634

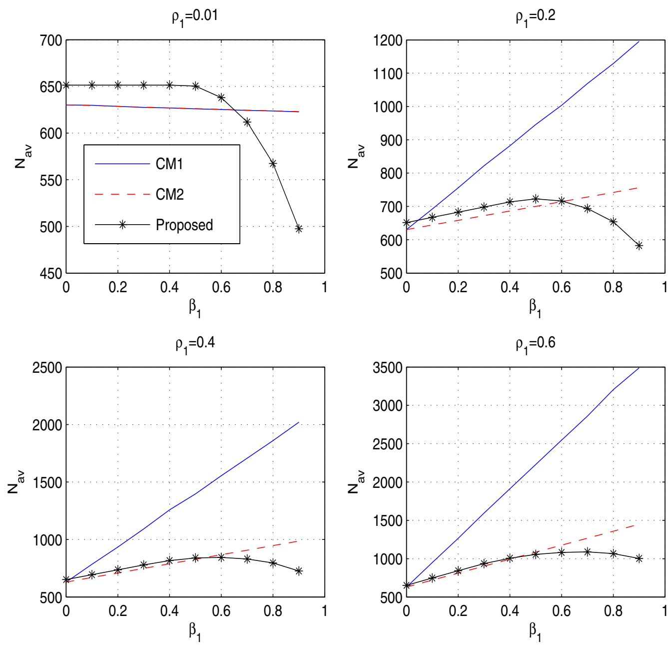

Fig. 4. Comparison between our proposed scheme with the carousel-Mohr schemes for different values of the channel erasure error rate ρ_1 for class 1. The channel erasure rate for class 2 is set to $\rho_2 = 0.01$, and N_{av} and β_1 denote the overall cost and the weighting factor for class 1, respectively. The average overhead rate $\epsilon_{d_1} = \epsilon_{d_2} = 0.05$ is employed.

At last, we present how an image can be progressively reconstructed with an increasing number of correctly received packets. Under the multicast scenario with $\beta_1 = 0.7$ and $\beta_2 = 0.3$, Figure 3 presents the progressively reconstructed images at an arbitrary class 2 receiver which encounters the channel error rate of 0.2. Theoretically, when the allocation profile has only two distinct values for K_m, where $m = 1, .., L$, the achieved PSNR should have only two distinct levels. However, due to the independently distributed random overhead rates for each layer, it is possible to decode the first l layers, where $l < d_1$, or $d_1 < l < L$, leading to extra PSNR levels. In this test, when 569 packets are correctly received, the first l layers, where $l < d_1$, can be decoded, resulting in an extra PSNR level of 21 dB.

D. Comparison with the PET-based UEP scheme in [5]

In this experiment, we compare our proposed rateless-UEP scheme and the scheme presented in [5]. For the proposed UEP scheme, a rateless code with a typical average overhead rate of 0.05 [1] is employed unless specified otherwise.

To apply the UEP scheme in [5] to systems with heterogeneous QoS constraints and multiple channel error statistics, while still maintaining its performance, we can search for the most demanding class, i.e., the class which requires the largest weighted cost $\beta_j N_j$, and then perform the design in [5] for this most demanding class. Instead, in this experiment, we simply choose the class with the highest target QoS, i.e., class J, as the most demanding class. After designing the scheme in [5] for this class, the performance of the design is assessed over a range of channel and weight parameter values. The scheme in [5] leaves the total number of transmitted packets N as a design parameter. We pick N as the minimum number of transmitted packets which accommodate all the source symbols needed to achieve γ_J for class J. To guarantee the QoS requirements for all the classes, the N packets are repeatedly transmitted until all the receivers' QoS requirements are satisfied. In effect, we have extended [5] to a scheme which we call carousel-Mohr (CM1) in order to suit the multicasting scenario.

With carousel transmission, there will be repeated packets at the receiver, leading to more transmitted packets required to achieve a target QoS. To account for the impacts of the above distinctness inefficiency, we also consider an unrealistic case, in which successfully received but repeated packets are not counted. We denote this unrealistic scheme as CM2.

With the proposed scheme and the above CM schemes, we now measure the cost, N_j, where $j = 1, 2$, at arbitrary class 1 and class 2 receivers. With measured cost N_j, we can then compute the overall cost N_{av} from (8). Figure 4 presents a comparison between our proposed scheme and the CM schemes for different combinations of the channel erasure rate ρ_1, and the weighting factor β_1.

From Figure 4, it can be observed that if the channel erasure rate is small, e.g., for the case of $\rho_1 = \rho_2 = 0.01$, the distinctness inefficiency of CM1 is not severe, and the two CM schemes can achieve very similar performance. In this case, the performance gap between the CM schemes and the proposed scheme is due to the zero overhead rate of the Reed-Solomon (RS) codes used in the CM schemes.

Comparison between the proposed scheme and the CM schemes shows that when the weighting factor β_1 is less than a certain level, class 2 becomes the most demanding class, thus matching the CM design, and in this case, the unrealistic CM2 scheme is superior to the proposed scheme, due to its employment of zero-overhead RS codes. We also see from Figure 4 that increasing β_1 and/or ρ_1 renders even the unrealistic CM2 scheme inferior to the proposed scheme. This is because the CM schemes are designed to optimize for the most demanding class, while the proposed scheme is designed to minimize the cost by considering the multiple channel error statistics, the heterogeneous QoS constraints, as well as the weighting factors. Therefore, CM2 is inferior to the proposed scheme when an increased weighting factor β_1 and/or channel erasure rate ρ_1 make class 1 the most demanding class. Overall, the proposed scheme substantially outperforms CM1, and either performs very close to or outperforms the unrealistic CM2.

Figure 4 clearly demonstrates the advantage of using rateless codes: every correctly received packet contributes to decoding. The price paid to obtain this desirable property of 'infinitely innovative' coded packets is expressed in the nonzero overhead rate of rateless codes.

In the above experiment, a typical overhead rate of $\epsilon_{d_j} = 0.05$ is employed for the proposed scheme. If a better rateless code with reduced overhead rate is employed, the performance of the proposed scheme is improved. For example, with $\epsilon_{d_j} = 0.001$, the performance of our proposed scheme is always superior to the unrealistic CM2 scheme for every tested combination of the channel error statistics and the weighting factor.

We remark that the systematic RS codes employed in the CM schemes have high decoding complexity, making them unattractive for time-constrained streaming and power limited wireless applications [13], while in our proposed scheme, due to the employment of sparse-graph rateless codes, the decoding time and the computational complexity can be dramatically reduced.

V. Conclusions

We have presented a novel transmission scheme for asynchronous and reliable multicast of scalable multimedia. This scheme employs a layered packetization structure that originates from PET, in combination with rateless codes. A near-optimal allocation scheme, which effectively assigns channel code rates, is developed within the layered multicast framework. This scheme provides an efficient approach to tradeoff the costs among different clients, so that heterogeneous QoS requirements can be satisfied with minimum overall system cost. The presented transmission scheme in this paper may be attractive to QoS-enabled multimedia business, such as Internet Protocol Television and wireless video on demand. In addition, our presented framework considers a system with uncertain channel statistics, which is more realistic for multicasting systems. As a result, our proposed scheme can be applied to a wide range of channel loss profiles, such as multicasting over ad hoc networks and wireless local area networks.

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