JOINT PACKET PRIORITIZATION AND QOS MAPPING FOR SVC OVER WLANS

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ABSTRACT

The emerging H.264/AVC extension, SVC encoding standard facilitates the truncation of bitstreams at certain points to fit in with wireless network variations. In this paper, we propose a joint packet prioritization and QoS mapping scheme based on priority analysis of SVC packets and the exploration of the service differentiations among IEEE 802.11e EDCA access categories. The proposed scheme enables the interaction among different layers, providing differentiated services for scalable video packets. The cross-layer optimization is performed based on SVC packet information at application layer, differentiated access categories at MAC layer and interface queue (IFQ) control at link layer. Based on these cross-layer information, the proposed joint packet prioritization and QoS mapping strategy minimizes the packet loss impact of visual quality. The proposed approach shows significant performance improvement when compared to conventional schemes for SVC streaming over 802.11e wireless networks.

Index Terms— SVC, Packet Prioritization, QoS Mapping.

1. INTRODUCTION

Wireless video finds its pervasive applications with the rapid progress in media coding technology, network reliability, and reduction of hardware cost. The scalable extension of the legacy H.264/AVC video coding standard, known as Scalable Video Coding (SVC) [1], is ideal in accommodating rigorous needs of video streaming in time-varying wireless channel. SVC has full compatibility of base layer which can be decoded with AVC decoders. It offers temporal, spatial and quality scalabilities with availability of part or all enhance layers packets. Resolution and quality can be improved with more enhance layers reached. These functionalities enable the regulation of bitrate according to the variation of networks and special needs from the end users.

In this work, we are interested in SVC delivery over 802.11e WLAN. The IEEE 802.11e standard [2] introduces two medium access mechanisms: contention-based channel access and controlled channel access, which greatly improve the convenience of multimedia communications. In particular, the contention-based channel access scheme, Enhanced Distributed Channel Access (EDCA), extends the legacy distributed coordination function by providing differentiated traffic prioritization by means of MAC layer access categories (ACs). For instance, delay-sensitive video and VoIP traffics should be given higher priority than the background traffic. The differentiated service capabilities among ACs are conducive for allocating packets with distinct priorities so to maximize the system performance.

Recently, some studies [3, 4, 5] on SVC streaming over lossy networks have been conducted. In [3], the authors proposed a traffic prioritization scheme for SVC packets. It resorted to encoding structure to decide packet priority, but packet length and throughput differentiation among ACs were ignored. A cross-layer design for SVC over IEEE 802.11e WLAN is presented in [4], the authors proposed a packet scheduling method according to SVC packet syntax. However, the throughput differentiation among ACs were not under consideration. In [5], the authors proposed an optimized cross-layer framework for motion compensated temporal filtering (MCTF)-based scalable video coding. A fixed differentiated ratio of bitstream distribution between AC2 and AC1 was adopted, which is not optimal for time-varying wireless environment.

In this paper, we propose an accurate packet prioritization scheme to evaluate SVC packet priorities and present an adaptive QoS mapping algorithm accordingly. The packet prioritization considers both which layer the data belongs to and what dependencies in the coding structure. The QoS mapping algorithm allocates video packets according to their priority indices obtained from application layer and simultaneously considers both the services differentiation among different ACs in MAC layer and interface queue occupation information from the link layer. The rest of this paper is organized as follows: Section 2 describes the details of our joint packet prioritization and QoS mapping algorithm. Simulation results are presented in Section 3. Finally, Section 4 concludes the paper.

2. PACKET PRIORITIZATION AND QOS MAPPING

In a time varying wireless environment, intensive competition and heavy traffic congestions bring difficulties to multimedia delivery. Meanwhile, large size SVC packets aggravate competition among both internal and external of EDCA queues, and the accumulative effects make the queue overflow non-ignorable. The scalable video provides high flexibility for packets prioritization, which is helpful in designing an adaptive QoS mapping algorithm in wireless networks to offer differentiated transmission capacities. The application layer decides an optimized packet allocation strategy based on packet information acquired from incoming bitstreams and network conditions from lower layers. Objective AC, sending order and timing of every packet are interactively decided and timely regulated based on the information of packets length, packet distortion indices, interface queues occupation conditions, and differentiation of ACs’ transmission capabilities.

2.1. Packet Prioritization

Priority of individual packet depends on its frame type, as well as its position in the coding structure. Due to the use of hierarchical
structure in SVC, the error propagation is confined mostly within one GOP. We resort to two objective video quality measures, mean square error (MSE) and flickering to estimate quality degradation introduced by individual packet loss. When packet $k$ belongs to frame $j$ in one GOP is lost, it will introduce distortion not only on frame $j$ itself but also on those predicted from $j$. Overall MSE induced by loss of $k$ is

$$\Delta_k = MSE_{k,j} + \sum_{h<j} MSE_{k,h} \tag{1}$$

where $MSE_{k,j}$ is MSE value between fully reconstructed frame $j$ and the one with packet $k$ lost, $h < j$ means frame $h$ is predicted from frame $j$. The last item expresses MSE summation of all enhance layers and subsequent frames that are contaminated.

Flickering is another important quality measure. Fan et al introduce a simple flickering metric for video coding [6]. In this paper, we make improvement on the measure to accommodate its efficiency for wireless video. Let $O(j,z)$ indicate $z$th macro block of the $j$th frame in fully reconstructed video, and $P(j,z)$ be the reconstructed one where packet $k$ is lost. Suppose $SAD(O(j,z), O(j+1, z))$ across all macro blocks from first to second-to-last frame as samples of a folded normal distribution, unbiased estimator of mean and variance could be derived and so the cumulative distribution function $F(x)$. The flickering introduced by loss of $k$ is

$$\Delta_k = \frac{AVG}{SAD(P(j+1, z) - P(j, z), O(j+1, z) - O(j, z))} \tag{2}$$

where $\gamma$ is percentage threshold to ensure certain ratio of collocated macro block differences that are extended after reconstruction is under consideration. Regardless of the motions of video sequence, a certain percentage will be covered therefore this measure is content-independent.

SVC facilitates the calculation of $\Delta_k$, for both MSE and flickering, due to the use of hierarchical encoding structure. Generally, the first encoded frame within every GOP is encoded as I or P frames, the remaining are encoded as B frames. The loss of packet $k$ which belongs to I or P frames introduces quality degradation to all subsequent frames from current GOP to the next intra coded frame. These in-between frames should all be measured to acquire $\Delta_k$. If lost packet belongs to B frames, only frames residing in the same GOP predicted from current frame should be measured. Therefore for SVC only a few frames will be employed for comparison to measure the distortion, extra computation introduced is limited so the measure can be additionally acquired after the encoding process.

### 2.2. EDCA Model Based Packet Allocation

We describe an analytical model of the EDCA under saturation traffic condition in this subsection. Based on the analyses on parameters of EDCA, differentiations of service capacity among different ACs are measured. The model is developed based on Markov Chain model constructed in [7]. We consider service ratios of ACs as functions of wireless nodes number in the medium so time-varying nature of wireless channel could be accurately followed. In this work, AC2, which is designed for video transmission in standard and AC1, which performs in a best-effort fashion, are used.

Let $N$ denote the number of nodes in WLAN. For a given station, its AC2 and AC1 are entitled with high and low priorities respectively. $CW_{\min}[i]$ indicates the minimal contention window size of AC$i$, and $CW_{\max}[i]$ is maximal contention window size. Denote $RL[i]$ the maximum retry limit for AC$i$. Based on EDCA backoff increase scheme, window size $W_{i,r}$ for AC$i$ in $r$th retransmission is

$$W_{i,r} = \min\{PF^* \cdot CW_{\min}[i], CW_{\max}[i]\} \tag{3}$$

where $PF$ is persistence factor. Parameter set $\{AIFS, CW_{\min}, CW_{\max}, PF, RL\}$ for AC2 in this work is $\{2, 15, 31, 2, 7\}$, and is $\{3, 31, 1023, 2, 4\}$ for AC1.

Let $\tau_i$ be the probability that AC$i$ transmitting packets in a generic time slot and $p_i$ the probability that AC$i$ senses the wireless medium busy due to both internal and external collisions. According to Markov chain model described in [7], we have:

$$\tau_i = \sum_{r=0}^{RL[i]} b_{i,r,0} = \frac{1 - \frac{1}{1 + RL[i]}}{1 - \frac{1}{p_i}} \tag{4}$$

where $b_{i,r,s}$ is stationary distribution of Markov chain, $i$ stands for AC$i$, and $r$ is the backoff stage having values in the region $\{0, 1, ..., RL[i]\}$, $s$ is the backoff delay taking values from set $\{0, 1, ..., W_{i,r} - 1\}$ in time slots. So here $b_{i,r,s}$ is probability that AC$i$ stays in backoff stage $r$ with backoff delay $s$. And the initial value $b_{i,0,0}$ based on stationary distribution of Markov model is

$$b_{i,r,0} = \frac{1}{\sum_{r=0}^{RL[i]} \left(1 + \frac{1}{p_i}\right)^{W_{i,1} - r} \left(\frac{1}{p_i}\right)^r} \tag{5}$$

The probability that AC$i$ senses the medium busy is

$$p_i(N) = 1 - \prod_{r=1}^{2} \left(1 - \tau_i\right)^N \tag{6}$$

Equations from (3) to (6) form a nonlinear equation set, which can be solved through numerical method. Based on derived $p_i(N)$, the probability $\tau_i$ that a wireless station gets access to the medium during a generic time slot is $\tau_i = 1 - (1 - \tau_i)(1 - \tau_j)$. The probability that AC$i$ suffers internal and external collision is

$$p_{i,N}(N) = 1 - (1 - \tau_i)^{N+1} \prod_{1 < s \leq 2} (1 - \tau_i) \tag{7}$$

Since ACs of all nodes in the medium compete for the channel access, collisions will encumber transmission from some ACs. The probability that AC$i$ in one node can successfully transmit is

$$p_{i,N}(N) = \tau_i(N - p_{i,N}) \tag{8}$$

From the view of a wireless node, the probability that it suffers collision should be $p_{i,N}(N) = \sum_{r=1}^{2} p_{i,r}(N)$, the probability that this node can successfully transmit is $p_i(N) = \sum_{r=1}^{2} p_{i,r}(N)$. Correspondingly, the probability that one node is busy is $p_{i,N}(N) = 1 - (1 - \tau_i)^{N+1}$. Based on the derived parameters for each AC, we have normalized throughput of AC$i$ as

$$S_i(N) = \frac{E(P_i)}{E(L_i)} = \frac{\sum_{p_i} p_{i,N}(N)E[p]}{1 - p_{i,N}(N)\delta + p_{i,N}(NT_s + p_{i,N}(NT_c))} \tag{9}$$

where $E(P_i)$ is the average payload transmission time in a slot time for the AC$i$, and $E(L_i)$ is expected length of a slot time. For basic access mode of 802.11e, $\delta = 20\mu s$, $T_s = E[p] + 131.8182\mu s$ and $T_c = E[p] + 93.1818\mu s$ are duration of an idle slot, successful transmission and collision, respectively. $E[p]$ is average duration of frame payload, please refer to [8] for its calculation.

Let $R(N)$ denote the ratio of normalized throughput between AC2 and AC1, which varies on wireless nodes number as

$$R(N) = \frac{S_2(N)}{S_1(N)} \tag{10}$$
Packets scheduled into AC_i will have a probability of p_{e,i} = R_{l}(|+1)_{l} to be dropped due to exceeding of the maximum retry limit.

For video packet k, its length is tagged as l_k. Packets in current GOP should be classified exclusively into two sets, \( \Phi_1 \) and \( \Phi_2 \), which are distributed into AC1 and AC2 respectively. The QoS mapping strategy to minimize the packets dropping can be formulated as

\[
\min \left( \sum_{a \in \Phi_1} \Delta_a p_{e,1} + \sum_{b \in \Phi_2} \Delta_b p_{e,2} \right) \text{ s.t. } \sum_{b \in \Phi_2} l_b - R(N) \sum_{a \in \Phi_1} l_a \leq \zeta
\]

where \( \zeta = \max \{l_1, l_2, \ldots, l_l\} \) is the largest packet size within current GOP. We solve this minimization problem using greedy algorithm, which is inspired by knapsack problem. First for all packets within the coming GOP, we deduce distortion to size ratio for packet k as \( r_k = \frac{\Delta_k}{l_k} \). Packets are arranged with \( r_k \) in descending orders, then they are classified into two sets at the point where the preceding packets that are going to be scheduled into AC2 have a total length that is \( R(N) \) times of the remaining ones which are for AC1. This mapping strategy at the application layer is combined with cross-layer control in next subsection to form the entire solution of our proposed QoS mapping algorithm.

2.3. Adaptive Regulation Based on Cross-layer Information

The previously derived allocation strategy is optimal from EDCA MAC view. We move further to minimize packets dropping due to interface queue (IFQ) overflow which is not contained in EDCA model to decide the final QoS mapping strategy. For packets scheduled into AC_i (i = 1, 2) within time slot \( T \), proposal of arranging sending order and timing for individual video packet k in order to minimize frame dropping due to IFQ overflow is generalized as

\[
\min \sum_{k} \Delta_k p_{d,k} \text{ s.t. } \sum_{k} t_k \leq T
\]

where \( \Delta_k, p_{d,k} \) and \( t_k \) are distortion once loss of \( k \), packet dropping probability and time interval for video packet \( k \), respectively. Once allocation strategy is determined, IFQ dropping probability \( p_{d,k} \) can be derived based on packet arrival and departure models.

We propose a simple and efficient solution. Denotes summation of average departure rate of both queues for AC2 and AC1 within last GOP as \( d \), where \( d \) is measured as UDP packets departed in unit time. This value is acquired as cross-layer information and is updated GOP by GOP. We assume network service rate approximately stable in two subsequent GOP. There are \( \text{ceil}(T/d) + m \) UDP packets scheduled for transmission in current GOP. An increment \( m \) is added to refrain from under utilization of network capacity.

The descending ordered sequence is truncated such that the preceding packets have \( \text{ceil}(T/d) + m \) UDP packets and the others are discarded beforehand. The reserved packets for transmission are allocated into two ACs according to the ratio \( R(N) \). Here we consider two ACs together to allocate packets for further improvement. This is because with descending ordered sequence derived according to \( r_k \), discarding packets at the end of the whole sequence then allocate with ratio \( R(N) \) is superior to allocating first then discarding at individual sequence end of two ACs. Amount of UDP packets kept in each queue is equal but more important packets are reserved for transmission. For each packet, allocating a waiting interval \( T_k \) subsequent to the previous sent video packet as

\[
T_k = \frac{T l_k}{l_{AC_i}}
\]

where \( l_{AC_i} \) is the the summation of length of all packets deserved in the AC in which packet \( k \) resides. In addition, least important video packets are discarded only when IFQ has been occupied by half or more.

3. SIMULATION RESULTS

In order to investigate the performance of our scheme, we use twelve nodes, six sources and six sinks. In each node, AC2 and AC1 are employed to transmit video. For these six source nodes, four of them transmit constant bit rate (CBR) traffic, the other two are for SVC bitstreams. The simulation is conducted on NS-2.33, and the implementation of EDCA of 802.11e for NS-2 is based on the model developed by Telecommunication Networks Group at Technical University of Berlin [9]. This model is modified to add in the retry limit differentiation among ACs to support high priority for AC2 transmitting more important video packets. Wireless nodes are placed in a 250 * 250m² area to communicate with random movements. The IEEE 802.11e settings follow the parameters listed in subsection 2.2. Two video sequences, one minute Bus and Foreman are encoded with frame rate 30 frames per second. SVC node pair 0 transmits and receives sequence Bus and node pair 1 is for Foreman. CBR nodes start transmission several seconds before SVC nodes and keep until video transmissions finish.

3.1. PSNR Validation

In this subsection, we make comparisons with conventional schemes where MSE-based packet prioritization is adopted. As the MSE measurement and PSNR are equivalent in illustrating video quality, PSNR results are given for MSE based packet prioritization simulations to enhance readability. First we compare our QoS mapping algorithm with the structure proposed in [3]. In the reference model, base spatial layer packets are ranked above enhance ones. Within one spatial layer, hierarchical structure and predicted relations decide priorities, the predicted one are ordered behind those it is derived. For EDCA, four ACs are equally used for transmission without considering their capacity differences. Two sequences are encoded with temporal and spatial scalabilities supported. We present average PSNR comparison in Table 1.

<table>
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<th>Ref</th>
<th>Our</th>
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<tbody>
<tr>
<td>Bus</td>
<td>26.12</td>
<td>32.31</td>
</tr>
<tr>
<td>Foreman</td>
<td>30.11</td>
<td>38.42</td>
</tr>
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</table>

As shown in Table 1, our scheme presents its advantage in handling packets allocation compared to equal distribution. In our scheme, discarding some least important packets beforehand leads to slight PSNR reduction in some frames, but this loss is only confined within frame itself or small number of subsequent ones. The most important ones are well delivered and overall quality is preserved. More critically, equal allocation in the reference [3] highly underutilize AC2, loss of excessive packets in AC1 which have less opportunity to acquire medium leads to severe packets loss. One thing needs to be supplemented is the excessive PSNR reduction in reference comes from the temporal and spatial encoding structure. Loss of individual packet in enhance spatial layer will lead to one frame totally unavailable, so PSNR degradation in [3] is severe.

Next we make a comparison between our algorithm with strategy adopted in [5]. The authors there adopted another scalable video
coding scheme based on MCTF. The encoded streams were allocated into AC2 and AC1 with a constant ratio 2:1, and an adaptive QoS mapping algorithm with micro control was proposed. We cannot make straightforward comparison on the whole structure since the CODEC is different so interaction between CODEC and network evaluation in their structure is not used in our scheme. We compare specific aspects proposed in [5] and validate the improvement of our QoS mapping strategy on packet allocation. Here, two sequences are encoded with all three types of scalabilities supported. We implement the same packet prioritization and allocation ratio is set to 2:1 in Ref-1 and Ref-2. QoS mapping strategy is implemented in Ref-2 but not Ref-1. Table 2 show the PSNR results.

<table>
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<th>Ref-1</th>
<th>Ref-2</th>
<th>Our</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bus</td>
<td>27.61</td>
<td>28.71</td>
<td>31.26</td>
</tr>
<tr>
<td>Foreman</td>
<td>31.75</td>
<td>32.75</td>
<td>37.08</td>
</tr>
</tbody>
</table>

Improvements could be seen when adaptive QoS mapping strategy is adopted when comparing Ref-2 with Ref-1, scheduling packets into two ACs according to their departure rate fully utilizes the overall network capacity. In addition, preserving more important packets maximizes the overall quality. However, when comparing Ref-2 with our scheme, we see the constant 2:1 allocation strategy is inferior to the allocation with accurate throughput ratio. Practical service ratio between two AC2 and AC1 should be greater than 2:1 and be adjusted based on wireless nodes number variation. Keeping 2:1 ratio does not fully exploit capacity of AC2, part of important packets are scheduled in AC1, which has lower priority in competing for wireless medium and make these packets easily exceed maximum retry limit thus be discarded. In above comparisons, the increments of PSNR are different, this is because different encoding settings are used for both videos and the packets length distribution among SVC layers is also content-dependent.

We also conduct comparison with [4], where SVC packet layer ID was employed for allocation. The throughput differentiation among ACs and packet length were not considered and there was no adaptive regulation mechanism. Our strategy also have a PSNR gain around 3 dB. Due to the limited space, details are not presented.

3.2. Flickering Validation

In this subsection, we validate the performance of our algorithm when flickering measure is adopted. Those lead to larger flicker measure values are given higher priority in ranking. We show the comparison between proposed scheme with no packet prioritization and QoS mapping strategy, allocation ratio between two ACs keeps \( R(N) \). Fig. 1 shows the results.

As illustrated in flickering measure curves, our QoS mapping scheme presents great advantage in handling packets allocation, flickers maintain at a low level in our scheme through the whole simulation. Allocating a waiting interval at application layer for incoming packets according to their length is helpful in reducing IFQ overflow, and packet prioritization ensure only the least important packets are discarded in case of network congestion. Without consideration of the bursty packet size, importance of video packets, and buffering status, the reference scheme results in large flickering artifacts and significant quality degradation.

For SVC video delivery over time-varying wireless network, we find the cross layers interactions outperforms reference algorithms presented in [3] and [4]. We also find the cross-layer control and adaptive QoS mapping are better than the constant ratio based approach proposed in [5]. Significant reduction in flickering is also validated when the proposed strategy is adopted.

4. CONCLUSIONS

In this paper, we propose an efficient scheme of SVC streaming over IEEE 802.11e WLAN based on joint packet prioritization and QoS mapping. From an accurate Markov model for IEEE 802.11e MAC protocol, we design a QoS mapping strategy based on three important factors: video packet priority information, capacity differentiations of AC at MAC layer, and interface queue regulation at link layer, for the SVC streaming. The improvements of the proposed scheme on visual quality compared to conventional strategies are demonstrated in various simulations.

5. REFERENCES


