Multimedia Specific Scheduling Algorithm for Wireless Networks

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Abstract: This paper studies on multimedia specific scheduling algorithm for wireless networks. Unlike other scheduling algorithms, instead of packet length, it uses playing time as bounded delay and compensation unit for the application-level QoS. It also considers property of encoding scheme for multimedia in order of priority. It studies the trade-off between QoS improvement and fairness through a simulation and then specifies an optimal relation with them. From simulation results, we know that the proposed algorithm improves PSNR about 43%, while the fairness is 89% of IWFQ's one.

1. Introduction

Multimedia applications become popular not only in wired networks, but also in wireless networks due to emerging high performance wireless networking technologies. Real-time multimedia applications require effective network quality of service (QoS) support in terms of throughput, delay, jitter and loss ratio. Due to the delay-sensitive nature of these applications, delay-sensitive scheduling is needed to meet their stringent delay requirements in packet transmission [1].

In wired networks, a popular model for packet scheduling over a link is the fluid fair queuing (FFQ) [2]. It would seem that the FFQ model is applicable to scheduling over a wireless channel, and several packet-level algorithmic implementations of this model (WFQ, WF2Q, SCFQ, STFQ, etc.), will work just as well for wireless channels [2]-[5]. However, there are two key characteristics of shared wireless channels which render the FFQ model inapplicable: bursty channel errors and location-dependent channel capacity and errors. This implies that at any time, it may happen that some flows can transmit but other flows cannot due to channel errors.

Many researchers have proposed various packet scheduling algorithms that provide packet-level throughput and delay bounds over the error-prone wireless channel. In the virtual clock model [6], when a flow has nothing to transmit during a time window, it can reclaim its missed share of the channel capacity at a later time. IWFQ only compensates if the flow has packets to transmit but is unable to do so because of channel error and bounds the amount of compensation [7]. WPS compensates a back-logged flow that is unable to transmit a packet during its scheduled slot only if some other flow transmits a packet during this slot [8]. In the wireless scheduling algorithms, compensation unit depends on packet size that was unable to transmit.

In the multimedia stream, delayed and bounded times are more important than packet length of them. In generally, frame lengths are different from each other, because typical multimedia data, MPEG, allows a variable length coding. In other words, same packet size does not mean same playing time any more. However, to measure delay and bound, all of the above algorithms use not time unit, but packet length.

2. Multimedia specific scheduling

All of the above algorithms provide flow isolation among different flows while supporting network-level QoS in the presence of location-dependent channel errors. However, for multimedia applications, the performance of theses scheduling algorithms should be evaluated in terms of application-level QoS, which is the users' perceived satisfaction. The application-level QoS is heavily dependent on the encoding scheme being employed by the multimedia applications [1]. The proposed algorithm is based on MPEG-4 which is the most popular multimedia encoding format.

We consider not only finish tag, but also playing time and delayed time for service tag. Unlike other wireless scheduling algorithms, the proposed algorithm uses delayed time as threshold of delay and compensation unit for the application-level QoS instead of delayed packet size. In the proposed algorithm, there are two operation mode, urgent mode and normal mode. In the urgent mode, schedule gives hightest priority to delayed multimedia flow. In the normal mode, it chooses a flow based on start tag and delayed time because it does not classify data type. Thus, multimedia flow is served by urgent mode when delayed time of multimedia flow is longer than threshold. The following equation (1) shows service tag calculation method in normal mode.

$$T_s = S(p_i^k) + L(p_i^k)/r_i - d_i \cdot \beta + R(0) \qquad (1)$$

In equation (1), first two parts are same as final tag of WFQ. $S(p_i^k)$ is start tag which is maximum value between k^{th} packet's arrival time of i flow and finish time for $(k\text{-}1)^{th}$ packet of i flow. $L(p_i^k)/r_i$ is transmission time of k^{th} packet where $L(p_i^k)$ is packet length for k^{th} packet of i flow and r_i is weighted value for i flow. d_i is delayed time of i flow based on playing time for multimedia packet and waited time in queue for non-multimedia packet. β is a weighted factor for delayed time and $R(0)$ is constant.

$$T_s = S(p_i^k) + L(p_i^k)/r_i + B_i^k - d_i + R(T(p_i^k)) \qquad (2)$$

Equation (2) shows service tag calculation method in urgent mode. In the urgent mode, we consider B_i^k -d_i instead of d_i in the normal mode, where B_i^k is bounded time (tolerable delayed time) for *i* flow. In the urgent mode, we choose packet with minimum remained tolerable delayed time. $R(T(p_i^k))$ is constant based on type of frames for k^{th} packet of *i* flow. Some flow in the urgent mode is prioir to flows in the normal mode because of constant relationship, R(0) » R(B) » R(P) » R(I), where I, P, B are I-,P-,B-frame of MPEG 4 stream, respectively.

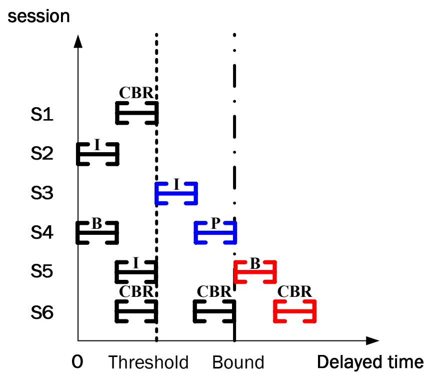

Figure 1. packet type and delayed time

Figure 1 shows packet type and delayed time of each flow. In the figure, CBR means non-multimedia type packet and I, P, B are I-, P-, B- frame's packets. B frame packet of S5 and CBR packet of S6 are discarded from the queue because their delayed times are longer than bounded time. For non-multimedia flow, bounded time means affordable waited time at the queue. It same as buffer overflow time at the allocated queue for corresponding flow.

In this section, we assume that packet length and weighted value of every flow are same. S6 is allways served by normal mode if it does not exceed bounded time, because of non-multimedia data. I frame packet of S3 and P frame packet are served by the urgent mode. In the urgent mode, scheduler chooses based on priority of frame type without delayed time. We know that I frame is the most important and P frame is more important than B frame in the MPEG 4. Thus, the first scheduled packet is I frame of S3, second one is P frame of S4, and then the last one is CBR of S6 among packets exceeding threadhold. In the normal mode, scheduling order is based on delayed time and weighted value for channel utilization. Figure 2 and 3 shows packet scheduling order in IWFQ and the proposed algorithm, respectively.

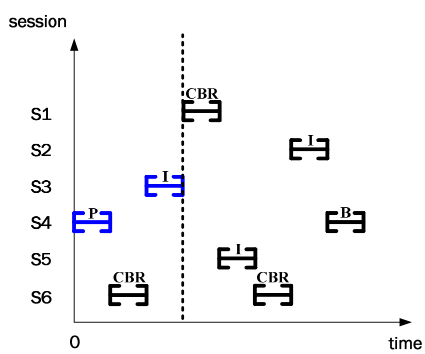

Figure 2. Packet schudleing order at the IWFQ algorithm

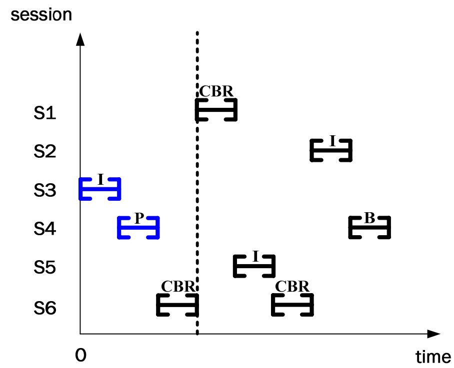

Figure 3. Packet schuduling order at the proposed algorithm

From the figures, we know that the proposed algorithm in the normal mode is same as IWFQ, while it is prior to scheduling multimedia flow in urgent mode where delayed time is longer than threshold. In the normal mode, among packets with same delayed time for S6, S1, and S5, S6 has least priority because it allocated channel for other packet previously. CBR packet of S1 and I frame packet of S5 have common priority because delayed times and weighted value for them are equal. I frame packet of S2 goes ahead because S4 used channel previously, although I frame packet of S2 and B frame packet of S4 have equal delayed time.

3. Simulations

To evaluate the performance of the proposed algorithm, we compare a PSNR of reconstruction images at the mobile node and fairness which is ratio of channel utilization.

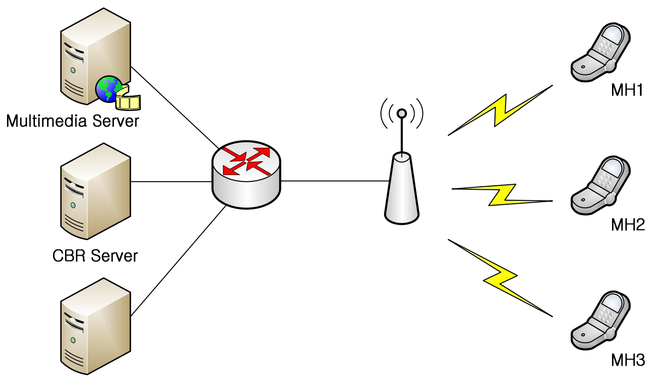

Figure 4. Network model for simulations

Figure 4 shows network model for simulations used in this paper. CBR server provides contant bits rate service of 1.5Mbps to MH 1, while multimedia server serves MPEG-4 video to MH 2. The base station is a scheduler which decides the transmission order of packets in the wireless channel. In the simulation studies, we assume wireless channel is as following.

- Each flow has perfect knowledge on channel state and transmitted packets are never lost in transit
- Channel state is not change during transmition after scheduling the packet.

As a simulation tool, we use the ns-2 based on C. H. Ke's work [9]. To classify multiple flows, we design a queue composed of multiple packet queues for each flow. Bounded time is 350ms and weighted values for MPEG4 video and CBR are 1.5Mbps, although average bit rate of video is 2.7Mbps. For evaluating degradation of QoS about multimedia service and fairness between multimedia and CBR service, we restricted bandwidth to 3Mbps that is less than required resources. Table 1 shows parameters for simulations and table 2 shows information of video source for simulation.

Table 1. Parameters for wireless multimedia scheduling simulation

Simulation Parameters	Value
Tolerable delay for play	350ms
Recoverable delay at BS	150ms
Bandwidth	3Mbps
Maximum data packet size	1024bytes
Weight for Video	1.5Mbps
Weight for CBR	1.5Mbps
Queue limits	50
β	1

Table 2. Information about sample video: Foreman

File Name	Foreman
Number of frame per second	30(frames/sec)
Running time	10sec
Total number of frames	300
Total number of GOPs	20
Number of frames per GOP	15
I / P / B	1: 4 : 10
Image size	352 X 288
Average size of I frames	31976
Average size of P frames	16543
Average size of B frames	3058
Average bit rate	2.7Mbps

3.1 Multimedia QoS (PSNR)

In this simulation, compared to conventional scheduling methods (FCFS and IWFQ), we measured average PSNR between original source frame and reconstruction frame at the mobile node. Figure 5 shows average PSNR of each GOP for the proposed algorithm, FCFS, and IWFQ. The figure shows that the proposed scheduler fairly improves PSNR because it considers a priority of packet type and delay sensitivity, unlikely to FCFS and IWFQ.

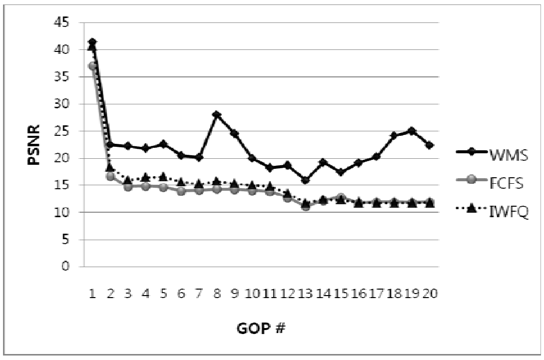

Figure 5. Average PSNR of each GOP

3.2 Fairness

As a coordinator, scheduler should fairly allocate wireless channel for flows. Most important role of the scheduler keeps up fairness. Thus, we evaluate receiving ratio of CBR to video. Figure 6 shows average unfairness. The figure shows that the proposed algorithm is similar to IWFQ for fairness.

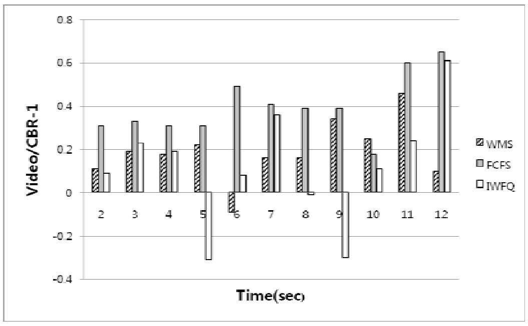

Figure 6. Average unfairness according to time variation

Table 3 summarizes simulation results of average PSNR and fairness. From the table, we know that the proposed algorithm improves PSNR about 43%, while the fairness is 89% of IWFQ's one.

Table 1. Simulation results of average PSNR and Fairness

	WMS	IWFQ	FCFS
PSNR	22.01	15.39	14.46
Fairness	1.19	1.06	1.38

4. Conclusion

We propose the wireless multimedia scheduling that improves the application-level QoS in error prone wireless networks while keeping fairness among flows. It focuses on delayed time as a factor for improving QoS of multimedia service. When some multimedia packets are expected late in normal scheduling, it transits an urgent mode that they can go first according to MPEG frame's priority. From simulation results, we know that WMS improves PSNR about 43%, while the fairness is 89% of IWFQ's one.

There is trade-off between improvement of QoS for multimedia service and fairness. However, we not only improve a PSNR, but also keep up fairness. It is possible to solve the problem because we rearranged a channel allocation time using a variable length coding property of MPEG-4 and minimized a waste of channel by removing irrecoverable packets in advance.

Acknowledgement

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