Performance Enhancement of Multimedia Data Transmission by Adjusting Compression Rate*

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Abstract. The rapid growth of wireless communication technology has spurred various mobile applications. In this paper we propose a scheme which can improve the PSNR of multimedia data transmission using the RTP/RTCP. It is achieved by adjusting the compression rate according to the packet loss rate, delay, or jitter in the network. The NS-2 simulation reveals that the proposed approach shows a significant improvement over the conventional scheme of fixed compression rate. Particularly, it turns out that adjusting based on jitter is more effective than with the other two factors.

Keywords: Compression rate, multimedia data, PSNR, RTP/RTCP, transmission interval.

1 Introduction

Resulting from rapid growth of wireless internet, various services are available in mobile communication environment [1]. Because of the mobility of hosts, surrounding buildings, and variation of terrain, the mobile communication channel suffers from doppler shifts in the carrier due to relative motion between the terminals. It also suffers from fast spatial fading due to regions of constructive and destructive interference between the signals arriving along different propagation paths. These problems may cause variation of signal attenuation or channel fading. During a long deep fade, data packets could be severely corrupted or completely lost [2]. Consequently, there is a need for an efficient data transmission mechanism which improves the quality of services with wireless networks [3,4].

HTTP and FTP are not the protocols designed for speed but stability using TCP layer. However, the TCP has several disadvantages with respect to multimedia data transmission; high processing overhead, network transmission delay, and lack of multimedia functionalities. Because of such disadvantages, the real-time transport

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protocol (RTP) was designed operating in the UDP layer [5]. It provides end-to-end network transport functions suitable for the applications transmitting real-time data such as audio, video, or simulation data over multicast or unicast network. The RTP does not address resource reservation nor guarantee quality-of-service for real-time applications. The application transporting multimedia data by the RTP uses two ports. One is for data transportation and the other is for control. The protocol used for control is called as real-time control protocol (RTCP), showing how many packets have been transported and received.

Currently, data are transmitted to mobile handheld devices at a fixed compression rate without considering operation condition such as error rate, delay, and variation in bandwidth. Therefore, it is difficult to provide the service of high quality [6]. In this paper we propose a scheme solving the problem by adjusting the compression rate based on either packet loss rate, delay, or jitter with the RTP/RTCP. It significantly improves the PSNR of the transmission compared to the conventional scheme. NS-2 simulation verifies the efficiency of the proposed approach. Particularly, it turns out that adjustment based on jitter is more effective than the other two factors.

2 The Proposed Scheme

There exist various obstacles interfering with data transmission in wireless network environment. Therefore, an efficient transmission mechanism which can dynamically cope with harsh environment is needed to provide high quality service. For this we develop a scheme which transmits multimedia data using the RTP/RTCP, while properly adjusting the compression rate according to the network condition. Figure 1 shows the flow of data in the proposed scheme.

Fig. 1. The flow of data in the proposed scheme

For data transmission, the data compression rate or transmission interval can be adjusted. By checking the sequence number and timestamp of the packets arriving at the receiver, packet loss, delay, and jitter can be identified. The compression rate is then

decided accordingly. When the packet loss rate, delay, and jitter are high, the receiver orders the sender to raise the data compression rate. The quality of motion picture can be improved by adopting this approach. Also, packet loss can be prevented by increasing the transmission interval. Through proper adjustment of the operation parameters according to the network condition, users can receive better services. Packet loss will increase if the bandwidth is not enough. Then the sender needs to reduce the amount of data transmitted at a time and raise the compression rate by increasing the quantization level of the transcoder. Table 1 summarizes the adjustment of compression rate and transmission interval according to the network condition represented by packet loss, delay, and jitter.

Table 1. Adjustment of compression rate (CR) and transmission interval (TI)

delay, jitter Packet loss	No change	Increase	Decrease
	No adjustment	TI++	TI--
No change	No adjustment	TI++	TI--
Increase	CR--	CR --/ TI ++	CR --/ TI --
Decrease	CR++	CR ++/TI ++	CR ++/ TI --

For simulation we choose an about 100Mbyte motion picture with frequent movement of receiver node to recognize picture collapse and frame loss. We implement and evaluate the proposed scheme using ns-2. We adopt the link error model given by ns-2 and assume that the channel error rate is 10%.

The operation flow is as follows. The raw data generated in the application layer are encoded to exclude the influence of PSNR of the compression. A timestamp and sequence number are inserted in the encoded data in the RTP/RTCP layer. The data are encapsulated by the UDP header for real-time transmission, and then sent through the network. The data received are passed through the UDP layer up to the RTP/RTCP layer and then the application layer. It orders the sender to adjust the transmission interval and compression rate using the timestamp and sequence number obtained from the RTP by sending an RTCP packet every 0.1 second.

Fig. 2. The comparison of the PSNR values with changed compression rate

The simulation measures the average PSNR values while varying the compression rate as the error rate, delay, and jitter value change. Note that varying transmission interval may not be suitable for real-time data transmission. Therefore, we consider varying only the compression rate.

Figure 2 shows the simulation results for each of the three cases of compression rate adjustment according to error rate, delay, and jitter, respectively. Observe from Figure 2(a) that the PSNR with changing compression rate based on error rate is statistically higher than with a fixed rate. Figure 2(b) shows that changing the rate based on jitter allows higher PSNR compared with the delay-based change. The average PSNR of the fixed compression rate, changing compression rate according to the error rate, delay, and the jitter are 26.7dB, 29dB, 31dB, and 34.4dB, respectively. The result indicates that adjusting compression rate based on jitter allows maximum throughput.

3 Conclusion

In this paper we have proposed a scheme for multimedia data transmission that effectively copes with unstable environment causing packet loss, delay, and variation in the bandwidth by adjusting the compression rate and transmission interval. The simulation using ns-2 reveals that the proposed approach significantly improves the efficiency of transmission of motion picture compared with the conventional data transmission employing a uniform compression rate. Especially, adjusting the compression rate based on jitter is found to be more effective than error rate or delay. Varying transmission interval may not be suitable for real-time data transmission.

Future work will include the investigation of controlling transmission interval along with data compression rate for achieving a better performance. We will also consider the channel error rate, delay, and variation in bandwidth not separately but in a unified way for better services in wireless mobile network environment.

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