A Measurement-Based Resource Allocation for QoS Provisioning over IEEE 802.11 Wireless LAN

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Abstract— **In this paper, we propose a measurement-based VBR control that adjusts the coding rates according to the channel characteristics of the wireless-to-wired network. This method can reduce substantially the abrupt quality degradation caused by the change of network condition and scene characteristics.**

I. Introduction

With rapid growth of emerging demand and deployment of wireless LAN (WLAN), much of IP traffic including multimedia traffic is forced to travel from wireless networks to the various networks [1]. Such a change in networking environments brings us a necessity to refine conventional rate control schemes. The conventional TCP-friendly end-to-end congestion control such as additive increase & multiplicative decrease (AIMD) control or TCP-friendly rate control (TFRC) evaluates the congestion degree and match the transmission to the available bandwidth in a wired IP network [3]. In a wireless-to-wired network packet losses can be caused by not only network congestion and unreliable error-prone wireless links. Therefore, AIMD and TFRC schemes can not be directly applicable to a wireless network because there is no way to distinguish congestion losses from wireless losses [4].

In this paper, we present a method to estimate the congestion packet loss ratio (PLR) for the TCP-friendly rate control in the wireless-to-wired network, and also introduce an enhanced frame skipping (EFS) to reduce the abrupt video quality degradation caused by abrupt scene change.

II. Proposed Rate Control System over WLAN

Our wireless streaming system consists of channel-adaptive TCP-friendly congestion control and EFS.

A. Channel-Adaptive TCP-Friendly Control

Fig. 1 shows that our wireless video streaming system transmits video to heterogeneous clients over time varying communication links. We assume that the bandwidth between the sender and wireless access point (AP) is reserved by the bandwidth allocation policy of AP to enhance the wireless link utilization. In the wireless hop, both the video server and AP have the additional module so called wireless adaptation layer (WAL) that is useful in estimating wireless channel condition. The WAL function in AP monitors the incoming traffics of all video sessions and sends the feedback messages including the received number of packets to senders periodically. Thus, wireless video server can estimate the wireless PLR $\widetilde{P}^i_{WAL}$ because the sender is aware of the number of packets transmitted before the arrival of the ith WAL packet. On the other hand, another packet loss information $\widetilde{P}^i_{RTCP}$ is obtained from the returned ith RTCP packets. When WAL and RTCP feedback packets are received, two PLRs are smoothed by

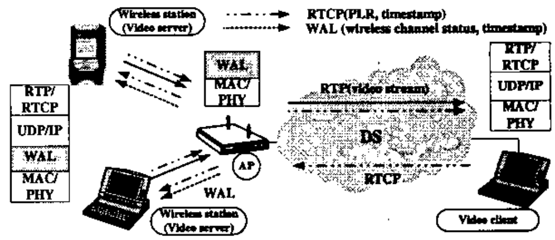

Fig. 1. Proposed wireless video streaming system over IEEE 802.11 WLAN.

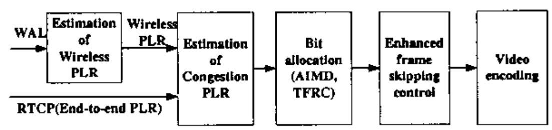

Fig. 2. Bit allocation procedure for the wireless video server.

$$\widetilde{P}^i_{WAL} = \alpha\widetilde{P}^{i-1}_{WAL} + (1-\alpha)P_{WAL} \quad (1)$$

$$\widetilde{P}^i_{RTCP} = \alpha\widetilde{P}^{i-1}_{RTCP} + (1-\alpha)P_{RTCP}, \quad (2)$$

where P_{WAL} and P_{RTCP}, respectively, are the newly obtained wireless and end-to-end PLR values and α is the weight parameter which is set to 0.9 currently.

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Once we obtain $\widetilde{P}^i_{WAL}$ and $\widetilde{P}^i_{RTCP}$, we determine the congestion PLR $\widetilde{P}^i_{CON}$ occurred by traffic congestion in the wired IP network.

$$\widetilde{P}^i_{CON} = max(\widetilde{P}^i_{RTCP} - \widetilde{P}^{i-k}_{WAL}, 0) \qquad (3)$$

where k is the difference between the two round trip time (RTT) values obtained from the latest feedback messages of AP and video client. Now, the wireless video server can calculate TCP-friendly rates by replacing end-to-end PLR by $\widetilde{P}^i_{CON}$. Thus, the target rates calculated using $\widetilde{P}^i_{CON}$ does not depend on the wireless losses. The throughput degradation of a multimedia flow due to wireless loss can be avoided as shown in Fig. 3.

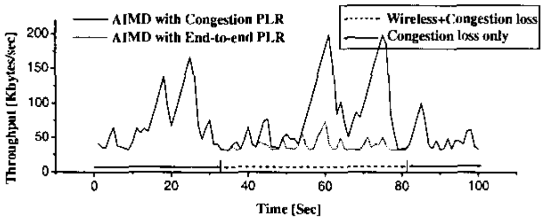

Fig. 3. Throughput comparison between congestion loss and end-to-end loss-based AIMD controls.

Additionally, the quality degradation due to the packet losses can be tolerated using forward error correction (FEC) at the application layer. While system faces to packet losses, the redundant rate is increased within given TCP-friendly rate. To determine redundant rate we should take into account the future status of the end-to-end network links. Since the video streaming system is located in the first-hop wireless link, FEC can promptly react to the wireless losses. However the wireless channel is unreliable and unstable to predict the future status. Thus, we need a robust PLR estimation method using received signal strength indication (RSSI) that is a measure of the RF energy received by the physical layer. $\widetilde{P}^i_{WAL}$ can be smoothed using its relation with RSSI measured at the wireless video server. Fig. 4 shows that $\widetilde{P}^i_{WAL}$ can be estimated approximately over measured RSSI using the linear regression method [2]. Redundant rates can be determined using the estimated wireless PLR and congestion PLR. Our experimental results show that the measurement-based rate control with FEC can enhance the video quality by 1-3 dB with the average PLR of 0.1.

B. Enhanced Frame Skipping

After allocating the target bit rates, the TMN8 encoder performs frame-layer rate control. But the allocated bits are not sufficient to encode a frame when the sequence characteristics significantly change. When scene changes abruptly, frame skipping is occurred since the statistical characteristics of the current picture are very different from that of the previous one. In our rate control, we use the pre-analysis method in which some frames are initially buffered before encoding to detect the scene change. The number of frames to be skipped in future is obtained by the quadratic rate model [2]. Then, we disperse the skipped frames widely using the predicted number of skipped frames as shown in Fig. 5. The proposed rate control can reduce the quality fluctuation caused by the abrupt scene change, resulting in 1-2 dB better than the conventional frame skipping method [2] in the period of scene change.

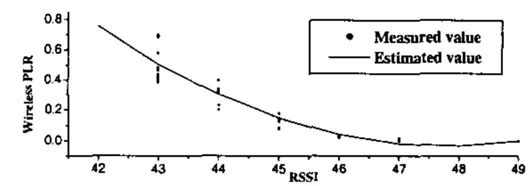

Fig. 4. Wireless PLR estimation over the measured RSSI (location : hallway in a building).

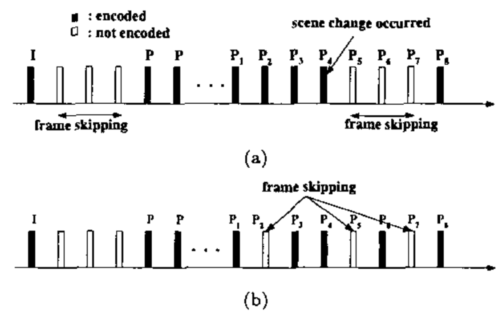

Fig. 5. Frame skipping methods. (a) Conventional frame skipping by the abrupt scene change. (b) New arrangement of frames to be skipped by EFS.

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