Optimal Video Stream Transmission Control over Wireless Network

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ABSTRACT
Wireless networks makes us to expect the realization of new applications using distributed multimedia like the advanced traffic system, the disaster prevention system, and the adhoc network system. However, since the network bandwidth over wireless network is not stable and limited because of higher packet loss rate inherently compared with wired networks, more careful QOS guaranteed function have to be introduced when video streaming service is provided. In this paper, we introduce the optimal control method to determine how to transmit the compressed video packets from server to client so that the required network bandwidth is minimized and network input/out buffers are optimized. Client server model for video streaming is proposed and numerical equations are formulated. The optimal sending data rate from server is derived and the temporal buffer occupancy is calculated. Computational result using the actual video data indicates the effectiveness of the suggested video data transmission.

Keyword: video stream, packet rate control, multimedia communication, traffic engineering

1. Introduction

Recently the distributed multimedia applications over wireless networks have been growing in various fields, such as advanced traffic system, the disaster prevention system and adhoc network. Those applications impose new challenges in networking efficient utilization of network bandwidth and computing resource where quality of service (QoS)[4][8] requirement of video and audio is met. However, since the network bandwidth over wireless network is not stable and limited because of higher packet loss rate compared with wired networks, more careful QOS guaranteed function have to be introduced when video streaming service is provided from server to client.

In this paper, we consider the end-to-end transmission of multimedia streams over current available wireless LAN, such as IEEE 802.11x. While video frame data are captured at the server (sender) side and divided into multiple packets and sent to client (receiver) through wireless network. The received packets are temporarily stored in the client buffer. Then the packets may be randomly lost on wireless LAN depending on the inherent environmental condition such as weather condition, antenna installation condition and wireless signal noise interference in addition to the network traffic condition. In order to deal with the packet loss, two methods including, Forward Error Correction (FEC) control [1][2] and Automatic Retransmission reQuest (ARQ) [3], are usually considered. In FEC control, the redundant packets are added to the data packet at sender side and sent to the noisy network. The arrived packets are received at the receiver side and checked whether packet error occurred or not. When packet error was detected, then this packet is corrected by decoder, such as Reed Solomon coding. There is no packet retransmission process in this method. However, the number of the packets transmitted increase depending on how the packet loss rate is. So far, we have developed the QoS control method for video stream over wireless network where the number of the redundant packets are dynamically control depending the observed actual packet loss[1][2]. As a result, the packet loss rate is maintained within the admissible range and the QoS of the video stream could be improved [9].

On the contrary, in the ARQ method, the packet loss is detected at received and feedbacked to the sender side to simply retransmit the equivalent packet. TCP protocol uses is based on this method although more sophisticated control mechanism using window control. However, this packet retransmission process leads extra packet traffic, eventually actual arrival video rate will decrease. In this case, in order to maintain the video play-out rate on the client side, the packet transmission rate must be increased. However, the packet loss randomly occurred and arrival packet rate also varied randomly in time. In addition, since each frame size of compressed video, such as JPEG, MPEG-1/2/4, varies in time depending on the movement and compression rate within a frame and inter-frame. Thus, the arrival video packet rate dynamically and largely changed. To absorb the change of this arrival video packet rate, buffer is prepared. However, if the receiving buffer size is small, it will be easily full and packet will be overflowed. On the other hand, if the receiving buffer size is large, video display will be delayed.

We try to minimize the utilization of buffer and at the same time, avoid sending too much data to overflow the client buffer because of the fixed client buffer size. The client buffer acts as a reservoir to regulate the difference between the transmission rate and the display rate. It is an important resource for users to prevent display jitter, i.e. buffer overflow and underflow. Usually the conventional network service uses constant transmission rate which is determined by the peak data rate. Hence, it is not efficient to use the peak bandwidth because of the varied display.

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throughput at the client.

Also since the transmission rate on the wireless network varies in time, more dynamic rate control method must be introduced. Furthermore, the required bandwidth should be minimized as much as possible because, the maximum bandwidth provided on the wireless is quite limited.

In this paper, a transmission schedule that dynamically changes the transmission rate designed, which can minimize the peak bandwidth, maximize the number of the packets stored in buffer at every time instant to keep the network utilization as much as possible. A network model is introduced to capture the relation of transmission rate, buffer packets and display rates. The optimal packet transmission rate is derived to minimize the quadratic cost function.

In the followings, in section 2, network model and system architecture of our video communication is defined. Section 3 deals with the optimal input control to minimize the quadratic performance index function under state vector equations. Section 4 describes network model for video stream with packet loss over simple wireless LAN and formulates the simulation model. Section 5 explains the simulation result for the model for the both cases with and without packet loss to evaluate the effectiveness of the suggested control method. Finally section 6 derived our conclusion and future works.

2. Network Model

2.1 Network Architecture

The system configuration and architecture of our suggested video communication system is based on the peer-to-peer model as shown in Fig. 1 and is organized by the media coordination system (MCS) [5][6][7] which consists of three layers, including the synchronization layer, the data transform layer, and the media flow control layer between the application layer and the transport layer in the OSI reference model to realize end-to-end QoS guarantee as depicted in Fig. 2.

The synchronization layer performs inter/intra media synchronizations in between audio and video frames. The media transform layer performs transcoding functions including transformation of a video coding to another one, such as from/to Motion JPEG to/from MPEG-1, 2, 4, changing Q-factor, frame rate, color depth and pixel resolution at the sender side. The controls of Q-factor, frame rate, and the transform of video format are dynamically fitted according to change of the user resource environment, such as CPU load and memory buffer occupancy.

The media flow control layer performs variable packet flow control and packet error detection and correction functions according to dynamic change of traffic load conditions of the computers and networks resource environment. MCS is furthermore vertically divided into four planes such as the user plane, the QoS maintenance plane, the control plane, and the stream management plane by referring ATM architecture and adopted QoS-A model of Lancaster university.

In the user plane, synchronization function between different media, such as audio and video streams, data transform function between different media attributes, media flow control for both constant and variable bit rate transmission for video/audio streams are performed.

In the control plane, connection establishment/release of the media streams and QoS renegotiation are maintained. In the QoS maintenance plane, each entity for video/audio services is responsible for the fine-grained monitoring and maintenance of their associated protocol entities. In the stream management plane, the most suitable QoS parameter values on each protocol values on each protocol layer are determined according to the users QoS requirements, characteristics of the source media data, output device attributes, and available computing and network resources. Fig. 4 indicates Video Stream Transmission Model for the system architecture in Figure 2.

3. Network model

The network model we proposed in Figure 3 is given in Figure 4. As can be seen from the Figure 3, \( R_{in}(t) \) and \( R_{out}(t) \) are the incoming/outgoing transmission rates from/to the transmission network, respectively. The relations between \( R_{in}(t) \) and \( R_{out}(t) \) are expressed as,

\[
R_{out}(t + \tau) = R_{in}(t) - W(t) \quad (3-1)
\]
where $\tau$ means the network delay of data packet transmission in the network. $W(t)$ is the disturbance by packet loss rate on the wireless network. Let $Q$ be the allocated client buffer size at the setup connection, $Q(t)$ be the size of packet in the client buffer, $L(t)$ be the display rate, and $e(t)$ be the difference between $Q_{\text{max}}$ and $Q_{\text{in}}(t)$. In order to maximize buffer utilization, $Q_{\text{in}}(t)$ should be close to $Q_{\text{max}}$. So we try to minimize $e(t)$ (the difference between $Q_{\text{max}}$ and $Q(t)$) and at the same time, to minimize the bandwidth requirement (i.e., to minimize the transmission rate $R(t)$). The difference between the packet size in buffer and the allocated buffer, $e(t)$ can be used as the feedback to control the transmission rate at the server side.

Instead of transmitting the packets in a fixed rate, transmission rate can be changed automatically according to the buffer packet and display rate. In order to maximize the buffer utilization and minimize the bandwidth allocation, Linear Quadratic (LQ) tracker is used to design the transmission schedule to get the optimal transmission rate.

Since the change rate of the buffer packets in the difference between the transmission rate and the display rate, the transmission schedule model is defined as

$$Q_{\text{out}}(t) = R_{\text{out}}(t) \cdot L(t)$$  \hspace{1cm} (3-2)

and $e(t)$ be the difference between the tracked $Q_{\text{out}}(t)$ and the allocated $Q_{\text{max}}$, and expressed as

$$e(t) = Q_{\text{max}} - Q_{\text{out}}(t)$$  \hspace{1cm} (3-3)

While $e(t)$ should be minimized, at the same time, the transmission rate $R_{\text{in}}(t)$ should be minimized to save the bandwidth. The performance index $J$ that needs to be minimized is defined as

$$J = \frac{1}{2} \int_0^{t_f} \left( q \cdot (Q_{\text{max}} - Q_{\text{out}}(t))^2 + k \cdot R_{\text{in}}(t)^2 \right) dt$$  \hspace{1cm} (3-4)

where $q$ and $k$ are weighting coefficients determined depending which is more important with buffer occupancy or required bandwidth. Since the playback time is quite long, $t_f$ approaches to $\infty$. According to the control theory[10], the optimal transmission rate can be expressed as

$$R'(t) = m \cdot \{Q_{\text{max}} - Q_{\text{out}}(t)\} + L(t)$$  \hspace{1cm} (3-5)

$$Q_{\text{out}}(t) = \{Q_{\text{max}} - Q_{\text{out}}(t)\} e^{-mt} + \int_0^t W(\theta)d\theta$$  \hspace{1cm} (3-6)

where $m = \sqrt{q/k}$ and $t_i$ is a time internal for observation of $Q_{\text{out}}(t)$. Thus, observing the $Q_{\text{out}}(t)$ at the client and sending back $Q_{\text{max}} - Q_{\text{out}}(t)$ to the server, the optimal transmission rate $R'(t)$ can be determined. In our method, in addition to the performance index function $J$, $R_{\text{max}}$ the maximal transmission rate due to the network bandwidth limitation, needs to be considered. When the optimal transmission rate obtained from Equation (3-5) is larger than $R_{\text{max}}$, $R_{\text{max}}$, will be used as the transmission rate and remaining packets in the buffer will be used for display to compensate the upper bound of the transmission rate.

4. Numerical Analysis and Evaluation

In the simulation, the actual video data was examined to evaluate the effectiveness of our suggested method. IEEE802.11b wireless LAN (2.4GHz, DSSS, 11Mbps) was used as the wireless network. Then, the real time video data with iStar Wars by Motion JPEG with 3.98 Mbps in average data rate (average compression rate = 1/15, 320x240x30fps) was transmitted from the server to the client. The numerical values of video source are shown in Table 1. Assuming the network environment of wireless LAN dynamically changes, the packet error rate was randomly changed from 0 to 30%. This packet loss lead performance degradation of video transmission rate from 0~1.2Mbps. In this condition, we would like to transmit the video packet so as to minimize the performance index function (3-4).

<table>
<thead>
<tr>
<th>Video Format Coding</th>
<th>Motion JPEG</th>
</tr>
</thead>
<tbody>
<tr>
<td>Frame Size</td>
<td>320x240</td>
</tr>
<tr>
<td>Frame Rate</td>
<td>30fps</td>
</tr>
<tr>
<td>Color Depth</td>
<td>3byte(Full Color)</td>
</tr>
</tbody>
</table>
Table 1 The Numerical Values of Video Source

<table>
<thead>
<tr>
<th>Average Compression Rate</th>
<th>1/15</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video Scene</td>
<td>Star Wars</td>
</tr>
<tr>
<td>Average Playback Rate</td>
<td>3.68 Mbps</td>
</tr>
<tr>
<td>Observed Packet Error Rate</td>
<td>0 – 30%</td>
</tr>
<tr>
<td>Network speed</td>
<td>11 Mbytes</td>
</tr>
<tr>
<td>Max. Buffer Size at Client</td>
<td>1 Mbps</td>
</tr>
<tr>
<td>Weight functions m</td>
<td>0.1, 0.3, 1.0</td>
</tr>
</tbody>
</table>

In the simulation, both cases with and without optical control method were examined for different weighting function values. Figure 5 shows optimal input rate $R_{in}(t)$ for both with and without random packet loss varied from 0~30% when $m$=0.2, and 1/30 sec for time increment. Figure 6 shows the packets in buffer $Q_{out}(t)$ As can be seen from Figure 5, in the case without noise, $R_{in}(t)$ initially increased to the max but gradually reduced and approached to the playback rate. On there hand, from Figure 6, the packets in the buffer $Q_{out}(t)$ gradually approached to the $Q_{max}$ namely, 8Mbits(=1MByte). Thus, both $R_{in}(t)$ and $(Q_{max}-Q_{out}(t))$ were properly controlled. When packet loss is existed, $R_{in}(t)$ was influenced by the $W(t)$ did not approached to the playback rate. This is because the packet loss existed, the actual input decreased, the feedback signal $e(t)$ reflected to the $R_{in}(t)$ , eventually, $R_{in}(t)$ increased.

5. Conclusion

In this paper, the optimal rate control for video stream transmission to minimize both the client buffer utilization and the video transmission rate under packet loss over wireless network is introduced. Compared with by giving the fixed transmission rate, the our suggested method can determines the video rate dynamically to achieve the maximum utilization of the client buffer and minimal allocation of bandwidth even the random packet loss or traffic load variation occurred over wireless network. The simulation result based on the actual video source data show that the proposed method can minimize the transmission rate whole the packet buffer in the client buffer can avoid the buffer overflow and underflow. Therefore, the proposed method is shown in practical, efficient and reliable to support variable bitrate video transmission.

7. Reference