An Adaptive Optimal Multimedia Network Transmission Control Scheme

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Abstract. In this paper, we present a sender-driven adaptive optimal multimedia network transmission control scheme, which takes into account the buffer occupancy and network delay to maximize the utilization of network resources. For this purpose, adaptive network resource optimization with quadratic costs is used in the proposed scheme to provide the minimal allocation of the bandwidth and achieve the maximal utilization of the client buffer. Simulation results show that the transmission rate can be dynamically adjusted at the server according to the changing network delays and buffer packet sizes at the client to avoid the loss of packets, and at the same time to achieve the minimal bandwidth allocation at maximal utilization of the client buffer.

1 Introduction

The development and use of distributed multimedia applications are growing rapidly. Some applications are video-conferencing, video-on-demand, and digital library. To provide cost-effective multimedia services and satisfy the quality-of-service (QoS) of different applications, efficient utilization of network resources is essential. Some dominating parameters for QoS are reliability, bandwidth, packet loss, and jitter [2][5][10]. In general, multimedia applications have highly time-varying bandwidth requirements. These are the requirements that the special characteristics of multimedia applications place on the network.

There are several approaches to address the requirements of multimedia transmission. One approach is the static resource reservation [3][6] schemes based on fixed resource allocation at the connection stage. With large variations in bandwidth requirements, static allocation usually results in considerable wastage of network resources. Another approach is rate adaptation that adjusts the bandwidth used by a transmission connection according to the existing network conditions [11]. Compared with resource reservation, the adaptive approach can better utilize the available network resource that changes with time. There are several types of adaptive control schemes, namely sender-driven, receiver-driven and transcoder-based. Sender-driven schemes require the server to adjust the transmission rate according to changes of the network resources. The most commonly used sender-driven schemes are buffer-based adaptation schemes and loss-based adaptation schemes [8]. Receiver-driven schemes require the receiver to select the transmission according to the network condition [9]. Transcoder-based mechanism requires a gateway at suitable locations to perform different types of transmission [1]. Though much work has been done in the area of adaptive rate control mechanisms [7], the provision of adaptive transmission scheme from the point of view of providing optimal network resource utilization is still a challenge.

In this paper, a sender-driven adaptive optimal multimedia network transmission control scheme is designed. The server can dynamically change the transmission rate according to the occupancy of the buffer along the transmission path. Here we only consider the buffer at the client. At the same time, the bandwidth allocation can be minimized and the utilization of client buffer can be maximized.

The organization of this paper is as follows. Section 2 describes the proposed adaptive network framework. Simulation results are given in Section 3. Conclusions are presented in Section 4.

2 The Proposed Transmission Control Scheme

Fig. 1 gives the proposed adaptive network framework. Let k be the time interval, $Q(k)$ be the packet size in buffer at time interval k, $R(k)$ be the packets transmitted from the server at time interval k, $P(k)$ be the packets arriving at the client buffer at time interval k, $L(k)$ be the packets used for playback at time interval k, and Q_r be the allocated client buffer size at the setup of the connection. Because of the network delay, it is obvious that $P(k)$ is not equal to $R(k)$.

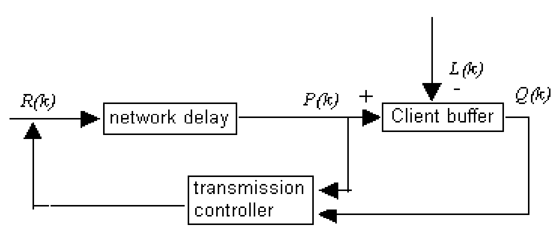

Fig. 1. The proposed adaptive network framework

In order to maximize buffer utilization, *Q(k)* should be close to Q_r. For simplification, *R(k)* is referred to as the transmission rate and *L(k)* as the playback rate at interval *k*. The optimization goals for the proposed framework are to minimize Q_r-*Q(k)* and the bandwidth requirements (i.e., to minimize the transmission rate *R(k)*).

Our approach is to design a transmission controller that provides high network resource utilization and good QoS. In this paper, packet loss and delay parameters for QoS are considered. Transmission rates can be changed automatically according to the existing network conditions such as the occupancy of client buffers and network delays, to maximize the buffer utilization and minimize the bandwidth allocation. Because the network delay is irregular due to the unpredictable network traffic, it is not possible to describe it with an exact mathematical model. Therefore, the modeling of network delays is considered as a stochastic process, and adaptive control for quadratic costs is used to design the transmission controller to obtain the optimal transmission rate.

2.1 Optimal Control for Quadratic Costs

For stochastic adaptive control, an ARMAX (autoregressive-moving average with exogenous input) process [4] that models the stochastic input-output feedback control system in discrete time is mainly considered. We can describe its multidimensional version as follows.

Let $A_1, A_2, \ldots, A_p, C_1, C_2, \ldots, C_r$ be m by m matrices and $B_1, B_2, \ldots, B_q$ be m by l matrices. We denote by y_k the m-dimensional output, u_k the l-dimensional control (input), and w_k the m-dimensional driven noise. The ARMAX system is in fact a stochastic difference equation:

$$\begin{cases} y_n + A_1 y_{n-1} + \ldots + A_p y_{n-p} & \\ = B_1 u_{n-d} + B_2 u_{n-d-1} + \ldots + B_q u_{n-d-q+1} & \\ \quad + w_n + C_1 w_{n-1} + \ldots + C_r w_{n-r} & n \geq 0 \\ y_n = w_n = 0, u_n = 0 & n < 0 \end{cases} \tag{1}$$

where $p \geq 0$, $r \geq 0$, and $d \geq 1$. The above equation can also be written in a compact form as follows.

$$A(z) y_n = B(z) u_{n-d} + C(z) w_n \tag{2}$$

where $A(z) = I + A_1 z + \ldots + A_p z^p$, $B(z) = B_1 + B_2 z + \ldots + B_q z^{q-1}$, $C(z) = I + C_1 z + \ldots + C_r z^r$, and z denotes the shift-back operator such that $z y_n = y_{n-1}$.

Consider the following quadratic loss function

$$J(u) = \limsup_{n \to \infty} \frac{1}{n} \sum_{i=0}^{n-1} [(y_i - y_i^*)^\tau Q_1 (y_i - y_i^*) + u_i^\tau Q_2 u_i] \tag{3}$$

where $Q_1 \geq 0$ and $Q_2 \geq 0$ are the weighting matrices and $\{y_i^*\}$ is a bounded deterministic reference signal. The control $\{u_i\}$ is designed to minimize $J(u)$.

2.2 Adaptive Optimal Transmission Controller

Consider the relationships among *Q(k)*, *R(k)*, *P(k)* and *L(k)* (defined earlier). Let *Q(k+1)* denote the buffer packet at the time interval *k+1*. We have the following buffer equation

$$Q(k+1) = Q(k) + P(k) - L(k) \tag{4}$$

P(k) can be represented as a function of the transmission rate *R(k-i)* of the source.

$$P(k) = B_1 R(k) + B_2 R(k-1) + \ldots + B_d R(k-d+1) \tag{5}$$

where $0 \leq B_i \leq 1$. The value of B_i depends on the percentage of those packets arriving at the kth interval to the packets transmitted at *k-i* time interval. The value for d can be determined by the timeout limit set by the transmission protocol. In order to catch the unknown network delay, the following model is introduced. With the knowledge of the incoming packets, the parameters of the model are updated at each time interval. That is, the vector $B = [B_1 B_2 \ldots B_d]$ is updated. So we have

$$Q(k+1) = Q(k) + B_1 R(k) + B_2 R(k-1) + \ldots + B_d R(k-d+1) - L(k) \tag{6}$$

In order to maximize the utilization of client buffer and minimize the allocated bandwidth, the function J is set as the optimization index function.

$$J(u) = \limsup_{n \to \infty} \frac{1}{n} \sum_{i=1}^{n} [(Q(i) - Q_r)^2 + R(i)^2] \tag{7}$$

where n is the total number of time intervals. In addition, the proposed transmission controller runs at a discrete time interval of T, which means the feedback information is sent back to the server every T seconds.

3 Simulation Results

In order to evaluate the effectiveness of the proposed framework, different playback rates are generated randomly between 0.1MB per second (MBps) and 1MBps to simulate the actual playback scenario. Fig. 2 shows the changes of the transmission rates, packet sizes in the client buffer, playback rates, and network delays. Assume the allocated client buffer size is 2MB and it is run within the interval [1, 100] with the increment of one interval. Since there is no accurate function to describe the network delays, a function of the vector $B = [B_1 B_2 \ldots B_i \ldots]$ is selected to roughly indicate the network delays. B_i is generated randomly to simulate the changing network delay. When B_i is increasing, more packets are arriving, which indicates the network delay at that time is decreasing. On the other hand, when B_i is decreasing, fewer packets are arriving indicating the network delay is increasing. The network delay displayed in Fig. 2

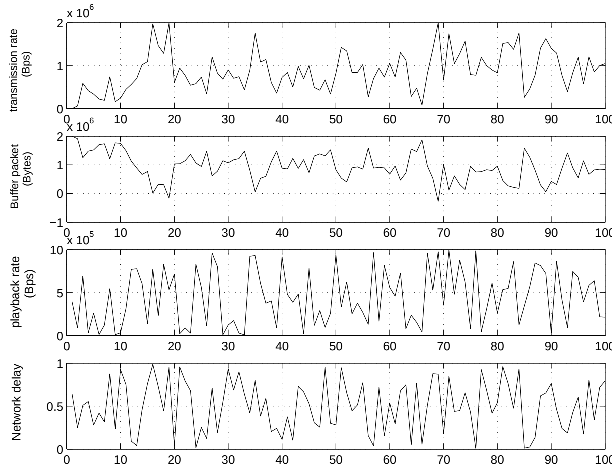

Fig. 2. Various transmission rates with different packet sizes in client buffer, playback rates, and network delays. The network delay is displayed as a weight value in the range of (0,1].

is a weight value in the range of (0,1], and a large value indicates large network delay.

The transmission rate for the next time interval is calculated based on the buffer packet, playback rate, and network delay information at the current time interval. In Fig. 2, the transmission rate displayed at a certain time interval is calculated at the end of this time interval and is actually the rate for the next time interval. The buffer packet displayed is the value when the current time interval is over.

As can be seen from this figure, the transmission rate is adjusted according to the buffer occupancy. When the buffer begins to fill up, the transmission rate is reduced. When the buffer turns to empty, the transmission rate is increased. Moreover, the transmission rate also changes with the network delay. For example, at the time interval 40, the playback rate is relatively high, buffer packet is medium, and the network delay is small, the transmission rate for the next interval (i.e., at time interval 41) is relatively low. The reason is that it takes a relatively short time for the packets to arrive at the client buffer to satisfy the playback requirement and not to overflow the buffer due to the small network delay. At the time interval 90, playback rate is low, buffer packet is low, and network delay is large. Therefore the transmission rate for the next time interval is high so that the packets can still arrive at the client buffer in time after a large network delay to satisfy the playback requirement.

The advantage of the proposed transmission control scheme is that the transmission rate of the server is dynamically adjusted according to the unpredictable network delay. The client buffer can be fully utilized under limited bandwidth requirement to provide jitter free playback. In our approach, the packet size in the buffer will never exceed the allocated buffer size, thus avoiding the overflow in client buffer and the loss of packets. In addition, the proposed transmission control scheme is easy to implement.

4 Conclusions

In this paper, a sender-driven adaptive optimal multimedia network transmission control scheme is presented. Under the proposed scheme, the server can dynamically adjust the transmission rate according to the buffer occupancy and network delay. An adaptive network framework is introduced to capture the changing network delays and adaptive optimal control with quadratic costs is used to achieve the maximal utilization of client buffer and the minimal allocation of bandwidth. Instead of giving a fixed transmission rate, the transmission rate can be determined dynamically to achieve the optimal utilization of network resources. The simulation results show that the proposed transmission control scheme can maintain high buffer utilization and minimum bandwidth allocation, and at the same time avoid buffer overflow.

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