Analysis of M/M^k/1 queuing model in AVS real-time transmission system

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Abstract

With the development of the video encoding standard AVS, it becomes urgently to transport AVS video stream over IP network for IPTV. This paper utilizing RTP/RTCP protocol to design and realize a real-time video transport system based on AVS standard, The workflow of video player is modelled as $M/M^k/1$ queuing system, using the states information feed-back from receiver buffer to adjust the sender rate and the buffer strategy. It provides important guidelines and insights on the design of streaming systems

Keywords: RTP; RTCP; Queuing Model; AVS;

1 Introduction

With the growth and popularity of Internet, the demand for multimedia increases dramatically, which motivates the emergence of network video service. With independent intellectual property rights, the domestic AVS [1-3] (Advanced audio video coding standard) qualify as national video coding standard and has been widely used in digital television or network video service. This paper focuses on real-time network transmission of AVS network video service.

Advanced Audio and Video Coding Standard (AVS) represents china's national standard for compression, which is developed by Audio and Video Coding Standard Working Group of China. Comparing to the H.264/AVC video coding standard, AVS provides a better trade off between complexity and coding efficiency for digital broadcast and digital storage media [4]. This is mainly due to the fact that many new techniques have been adopted in AVS, such as entropy coding, intra–frame prediction, transform and quantization de-blocking filter and so on.

In this paper, AVS real-time transmission system is realized on the basis of RTP/RTCP [5-8]. In this system, the service velocity of client is feedback by RTCP packet. As described by [9] high values of the jitter are usually caused by inadequate queue disciplines and/or the network congestion [10]. Proper queue discipline can play an important role in the efficiency of real-time transmission. The queuing theory is incorporated into the AVS real-time network transmission system in following section. The $M/M^k/1$ model is employed as the queue-waiting model before service, which motivates computation method of buffer size in the client of this AVS real-time network transmission system.

2 The structure of AVS real-time transmission system

The real-time transmission system consist of the server and the client, the two parts are connected by a local area network. At the server, RTP encapsulation for AVS Video Streams and send it to the client by using the RTP module, while sends the RTCP data packet on schedule. The client receives data of RTP data packets to unpack, and remove the AVS stream data, which is sent to the decoder for decode and display, the statistical analysis of received data is also send to the server.

The structure of our video real-time transmission system is shown in Figure 1, the server packed the AVS stream and transmitted it to the client, and the RTCP packet is used for Synchronization feedback of statistical analysis in the system and used UDP/IP to control the bottom protocol. Socket is used for interface of network communicating programs, which is common standard for developing network communicating programs.

3 Queuing theory-based intelligent buffer control strategy

The law of data packet can be studied based on queuing theory, time intervals are independent random variables can be described by Poisson stochastic processes, which are completely determined by some factors such as negative exponential distribution of service time, Poisson distribution of arrival interval. The queuing behavior of the system is investigated.

Buffer model is depicted in Figure 2. In order to ensure the decoding quality, we need to design a proper buffer strategy, if the buffer size is too long, will lead to unnecessary waste of resources, when the buffer size is too short, will lead to overflow or underflow. The average queue size is approximately derived as follows.

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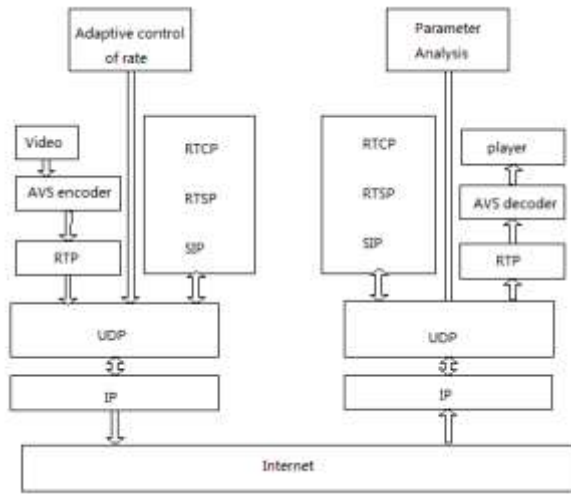
Zhang Qian , Huang Jifeng

FIGURE 1 The platform of AVS Real-Time Transmission System

As we know that the time intervals of data packet can be considered to be a Poisson distribution. On the other hand, the decoder need to receive a full frame data (including *k* data packets) for decoding and playing as is shown in Figure 2, time intervals of service are independent and identically distributed variables, this is
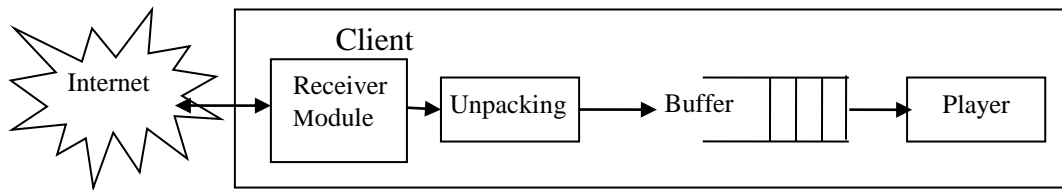
equivalent to the classical queuing model $M / M^k / 1$ [11],so we use it for the buffer size design.

Supposes the system in the time t, $N(t) = j$,if $j < k$ represent the server is free and *j* data packets are waiting for the services. Conversely, it represents that *j-k* data packets are waiting for the services.

Let be $P_j(t) = P\{N(t) = j\}$ $(j = 0, 1, 2, \cdots)$, we can get the state transition equation as follows:

$$P_0(t + \Delta t) = \mu \Delta t P_k(t) + (1 - \lambda \Delta t) P_0(t) + o(\Delta t)$$
$$P_j(t + \Delta t) = (1 - \lambda \Delta t) P_j(t) + \lambda \Delta t P_{j-1}(t) + \mu \Delta t P_{k+j}(t) + o(t), \quad 1 \le j \le k-1$$ (1)
$$P_j(t + \Delta t) = (1 - \lambda \Delta t - \mu \Delta t) P_j(t) + \lambda \Delta t P_{j-1}(t) + \mu \Delta t P_{k+j}(t) + o(t), \quad j \ge k$$

Let (1) divided by Δt , when resolving limit $\Delta t \to 0$, we Can obtain the difference equations:

$$\begin{cases} p_0'(t) = -\lambda p_0(t) + \mu p_k(t) \\ p_j'(t) = \lambda p_{j-1}(t) - \lambda p_j(t) + \mu p_{k+j}(t) \quad 1 \le j \le k-1 \\ p_j'(t) = \lambda p_{j-1}(t) - (\lambda + \mu) p_j(t) + \mu p_{k+j}(t) \quad j \ge k \end{cases}$$ (2)

FIGURE 2 The client model

Let $t \to \infty$, we can get the equations:

$$\begin{cases} \lambda p_0 = \mu p_k \\ \lambda p_j = \lambda p_{j-1} + \mu p_{k+j} \quad 1 \ge j \le k-1 \\ (\lambda + \mu) p_j = \lambda p_{j-1} + \mu p_{k+j} \quad j \ge k \end{cases}$$ (3)

If, $t \to \infty$,define $\rho = \dfrac{\lambda}{\mu * k} < 1$ as system load, thus

$p_j > 0$.

Introducing the function of queue

size $P(z) = \sum\limits_{j=0}^{\infty} z^j p_j, |z| \le 1$ distribution, we can get:

$$P(z) = \sum_{j=0}^{\infty} z^j p_j = \frac{\mu(1 - z^k) \sum\limits_{j=0}^{k-1} z^j p_j}{\lambda z^{k+1} - (\lambda + \mu) z^k + \mu}, \quad |z| \le 1$$ (4)

Let $\tilde{f}(z) = \lambda z^{k+1} - (\lambda + \mu) z^k + \mu$. (5)

As *P(z)* convergence in the space $|z| \le 1$, obviously the zero points $z_1, ..., z_{k-1}, z_k = 1$ (in $|z| \le 1$) of denominator are also numerator's zero points, we can derive that

$z_1, ..., z_{k-1}$

are zero points of $\sum\limits_{j \sim 0}^{k=1} p_j z^j$

from formula (2), so

$$\sum_{j \sim 0}^{k=1} p_j z^j = A \cdot \prod_{j=1}^{k-1} (z - z_j) ,$$ (6)

where, A is a undetermined constant.

As the formula (5) is the *k+1th* order polynomials, there must be a zero point z_0 in the space |z|>1,so we can rewrite formula (5) as follows:

$$\tilde{f}(z) = \lambda z^{k+1} - (\lambda + \mu) z^k + \mu = \lambda \prod_{j=0}^{k} (z - z_j) .$$ (7)

And then inserting formula (6) and formula (7) in Formula (4) combined with $z_k = 1$

$$P(z) = \frac{A\mu \sum_{j=0}^{k-1} z^j}{\lambda(z_0 - z)} \; . \tag{8}$$

Inserting z=1 in formula (8), we can get:

$$1 = P(1) = \frac{Ak\mu}{\lambda(z_0 - 1)} .$$

Thus,

$$P(z) = \sum_{n=0}^{\infty} \frac{(z_0 - 1)(1 - z^k) z^n}{k z_0^{n+1}(1 - z)} \; , \tag{9}$$

where, z_0 is the zero point exterior unit one , so we can get the coefficient of z^n is as follows:

$$p_n = \begin{cases} \dfrac{z_0^{n+1} - 1}{k z_0^{n+1}} & n = 0,1,...,k-1 \\[2ex] \dfrac{z_0^k - 1}{k z_0^{n+1}} & n = k, k+1,... \end{cases} \tag{10}$$

The conditional queue size is calculated as follows:

$$N = \sum_{n=0}^{\infty} nP_n = \frac{(k-1)(z_0 - 1) + 2}{2(z_0 - 1)} \quad . \tag{11}$$

However,in $M / M^k / 1$ queuing system, the client need k data packet for once decoding, so the Buffer size N_q is set as:

$$N_q = \max(\lceil N \rceil, k) . \tag{12}$$

If $k=1$, the $M / M^k / 1$ queuing system degenerate into $M / M / 1$ queuing system, state transition diagram is shown in Figure 3.
By using probability theory, we can get the equilibrium equations:

$$p_n'(t) = \lambda p_{n-1}(t) + \lambda p_{n+1}(t) - (\lambda + \mu) p_n(t) \; , \tag{13}$$

$$p_0'(t) = \mu p_1(t) - \lambda p_0(t) , \tag{14}$$

where, the average arrival rate of the event in stochastic process is λ ; $\mu = 1/T_s$ is the average severing rate

If $t \to \infty$, $\rho < 1$,then $p_n = \rho^n(1 - \rho)$ we can get the queue size N under equilibrium condition:

$$N = \sum_{n=0}^{\infty} n p_n = \sum_{n=0}^{\infty} n p^n (1 - \rho) = \frac{\rho}{1 - \rho} . \tag{15}$$
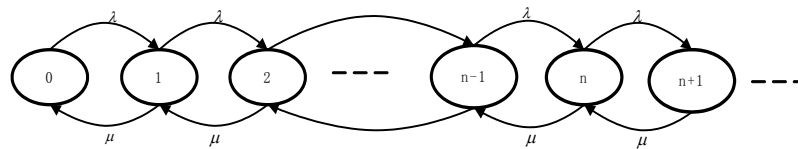

FIGURE 3 State transition diagram of M/M/1

4 Experimental results

Set *k*=5/4/3, the curve diagram: system load-buffer size is made accordingly, and depicted in Fig.4, Fig.5, and Fig.6. The horizontal coordinate represents system load $\rho = \lambda T_s / k$, the longitudinal coordinate represents buffer size, the dashed line is the steady-state queue size *N*. The solid line is the queue size after the adjustment. We can get the different queue size by using $M / M^k / 1$ queuing model, as shown in Figures 4-6.
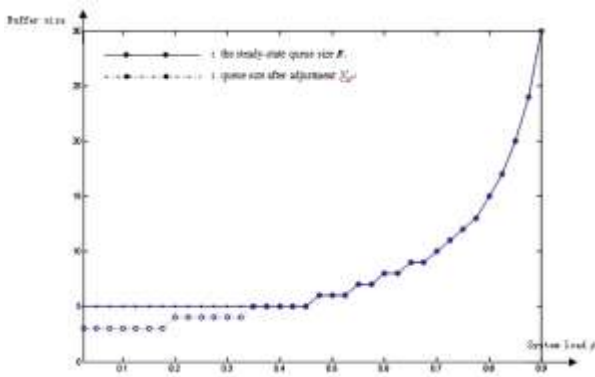

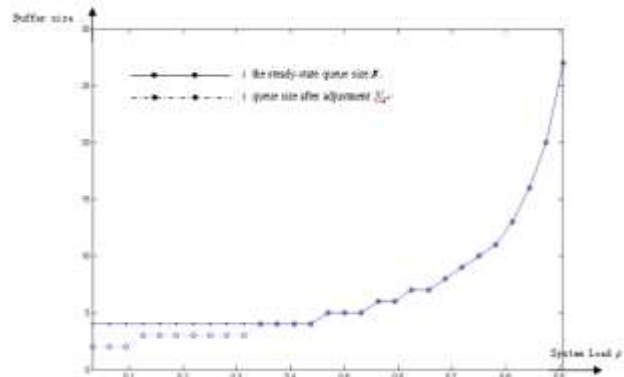FIGURE 4 The system load-buffer size curve of *M/M⁵/1* system

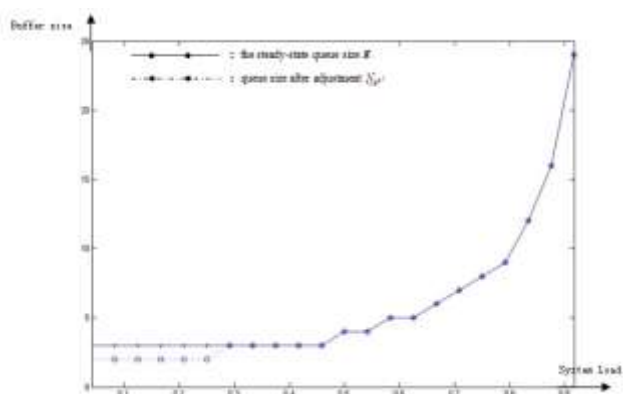FIGURE 5 The system load-buffer size curve of *M/M⁴/1* system

FIGURE 6 The system load-buffer size curve of $M/M^{3/}1$ system

5 Conclusions

In this paper we focuses on the process of video transmission for AVS video stream and provide a new buffer control strategy, queuing theory is applied to the real-time transmission system based on RTP/RTCP protocol architecture. The results could assist researchers and engineer set the buffer size. Though AVS is taken as an example in this study, the proposed algorithm can also be applied to H.264.

Acknowledgement

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