PAPER

Adaptive Video Quality Control Based on Connection Status over ATM Networks

Pao-Chi CHANG†, **Jong-Tzy WANG**†*, *and* **Yu-Cheng LIN**†, *Nonmembers*

SUMMARY The MPEG video coding is the most widely used video coding standard which usually generates variable bitrate (VBR) data streams. Although ATM can deliver VBR traffic, the burst traffic still has the possibility to be dropped due to network congestion. The cell loss can be minimized by using an enforced rate control method. However, the quality of the reproduced video may be sacrificed due to insufficient peak rate available. In this work, we propose an end-to-end quality adaptation mechanism for MPEG traffic over ATM. The adaptive quality control (AQC) scheme allocates a certain number of coding bits to each video frame based on the network condition and the type of next frame. More bits may be allocated if the network condition, represented by the connection-level, is good or the next frame is B-frame that usually consumes fewer bits. A high connection-level allows a relatively large number of tagged cells, which are non-guaranteed in delivery, for video frames with high peak rates. The connection-level adjustment unit at the encoder end adjusts the connection-level based on the message of the network condition from the quality monitoring unit at decoder. The simulation results show that the AQC system can effectively utilize the channel bandwidth as well as maintain satisfactory video quality in various network conditions.
key words: *rate control, quality control, quality of service, ATM, MPEG*

1. Introduction

With the development of high speed network technology and highly efficient video coding standards, it shows great potential to deliver video applications over high speed networks. Asynchronous transfer mode (ATM) networks have been proposed to provide a unified transport structure for Broadband-ISDN [1]–[4]. The MPEG video coding is the most widely used video coding standard which usually generates variable bitrate (VBR) data streams [5]–[7]. Although ATM can deliver VBR traffic, the burst traffic still has the possibility to be dropped due to network congestion. The cell loss problem is an important issue to MPEG video coding because MPEG video is very sensitive to the channel disturbance [8], [9]. Each coding unit, e.g. a macroblock (MB) or a frame, may be referred by other coding units in the decoding process. Thus a single error may cause serious error propagation in both the spatial and the temporal domains, and degrades the video quality substantially.

It is necessary to control the peak cell rate (PCR) of the VBR traffic to avoid the cell loss. Depending on if the encoder is affected, in general, there are two approaches to the PCR control. The first approach is the traffic shaping or the traffic smoothing which uses a buffer to store the data temporally and smooth out the traffic [10], [11]. The encoding process is not affected in this approach. However, it may need a large buffer to completely solve the problem. This increases the hardware cost and the end-to-end delay. The second approach is the source rate control which changes the encoding parameters to generate output streams at a pre-determined rate [12], [13]. It is able to generate a smoother traffic. However, some of the video frames that require high peak rates may be coarsely quantized. Thus it produces streams with inconsistent video quality that may be perceptually annoying. To make the problem even worse, those high peak rate frames which are forced to reduce quality usually are important, e.g. I-frames, and may be referred by other frames. As a result, the poor quality may propagate to the succeeding frames.

In this work, we propose an adaptive quality control (AQC) mechanism for MPEG traffic over ATM networks. It utilizes the concepts of both approaches. However, neither one is heavily used to minimize the disadvantages. In addition, the information of the network condition and the type of next frame are used to better control the reproduced video quality. More bits may be allocated if the network condition is good or the next frame is B-frame that usually consumes fewer bits.

The AQC system consists of three parts, the encoder with adaptive rate control, the decoder with quality monitoring, and the connection-level adjustment unit at the encoding end. The adaptive rate control unit determines the target rate for each frame based on the buffer state. To take the advantage of statistical multiplexing, tagged cells, i.e., cells with non-guaranteed delivery, are permitted. However, the number of allowed tagged cells is determined based on the network status, represented by the connection-level. A high connection-level allows a relatively large number of tagged cells for video frames with high bitrates. As a result, nearly constant video quality can be maintained. On the other hand, no tagged cells are desirable at a

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low connection-level that reflects a poor network status, because any cell loss may cause serious error propagation.

The network status is monitored by the quality monitoring unit at the decoder. The monitoring object could be either the cell loss ratio from the perspective of the network, or the video slice errors from the perspective of the video. Both measurements reflect the condition of the connections. The connection-level adjustment unit at the encoder end adjusts the connection-level based on the message of the network condition from the quality monitoring unit at decoder. When the network connection is good, it increments the connection-level by one. On the contrary, once a poor network connection occurs, the connection-level jumps back to the lowest level which represents the poorest connection and no tagged cells are allowed. The reason for this reaction is to minimize the number of lost cells because any lost cell may cause serious damage to the video quality.

This paper is organized as the following. In Sect. 2, we describe several traffic shaping and source rate control methods. In Sect. 3, the structure of the adaptive quality control system that we propose is described in detail. The simulation results and conclusion are presented in Sect. 4 and Sect. 5, respectively.

2. Traffic Shaping and Source Rate Control

In this section, we first describe how ATM networks provide quality of service (QoS) and the reactions when the traffic exceeds the QoS granted. Then we discuss the methods to regulate VBR video, including traffic shaping with buffers and the non-tagging source rate control.

2.1 ATM Quality of Service (QoS)

The concept of ATM QoS can simply be described by two steps.

- Connection admission control (CAC)—A user submits a traffic contract. The ATM network then replies whether to accept this connection or not corresponding to this contract.
- Usage parameter control (UPC)—The ATM network must ensure the connection to be no violation to the traffic contract.

The principle of CAC is that any new connection should not affect the QoS of the existing connections [14], [15]. A new connection request should be rejected if it requires more resources than the network can provide and may degrade the service quality of existing connections. Otherwise, a new connection is granted. Once a connection is established it must follow the initial traffic contract to avoid affecting the service quality of other connections.

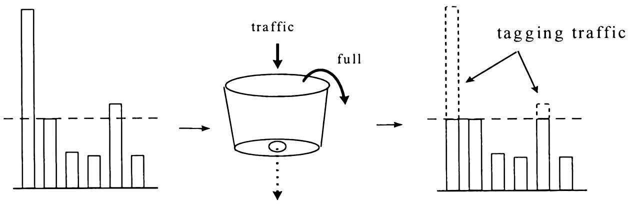


Fig. 1 Leaky bucket conformance.

2.2 Leaky Bucket Traffic Conformance

The leaky bucket model, shown as in Fig. 1, is generally used in ATM networks to verify if the traffic of a connection obeys the contract. The UPC of ATM guarantees the service quality as long as the traffic passes the conformance test of the leaky bucket. Otherwise, the service is not guaranteed.

In the case that the traffic exceeds the QoS requested, UPC might process the exceeding traffic with four policies of discarding, tagging, monitoring, or no action. In this work, we consider the tagging method which sets the cell loss priority (CLP) bit in the ATM header to be 1 for the exceeding cells. If some cells must be discarded due to network congestion, those cells with CLP=1 will be discarded first.

The tagging policy can take the advantage of statistical multiplexing to use the remaining bandwidth of other connections to deliver tagged cells. The probability that all connections working at peak rates is low. Thus tagged cells are not necessarily to be discarded in actual transmission. Note that since the tagged cells are not guaranteed to be delivered successfully, the tagged traffic is not considered to be included in the contract.

One way to reduce the tagged cells is to add a buffer to the leaky bucket. If the buffer size is sufficiently large, all cells will not be tagged as long as the average input rate is less than the token rate. As a result, the original VBR data becomes nearly CBR traffic.

2.3 Source Rate Control

In this section, we discuss two rate control methods. The first is MPEG-2 Test Model 5 (TM5) which utilizes the characteristics of the video sequence to allocate the bitrate. The second is the non-tagging rate control method which modifies TM5 toward the goal such that all frames obey the traffic contract and no tagged cells exist.

2.3.1 TM5 Rate Control

In general, MPEG-2 coding generates VBR bit streams for constant video quality. Similar to the layering structure of video coding, TM5 rate control also operates based on layers. It allocates the bitrates for each group-of-picture (GOP), frame, and MB in video sequences. Basically, it consists of three major steps.

Step 1. Target bit allocation

Before the actual encoding, it determines the number of bits for each GOP according to three user defined parameters: the bitrate, the picture rate (frames/second), and the number of frames in a GOP. The remaining unused bits from the previous GOP are added to the current GOP. Then the target bit allocation for each frame is calculated based on its frame type. I-frames generally get the highest rate.

Step 2. Rate control

The next step after the bitrate for a frame is determined is to calculate the reference quantization parameter for each MB. The reference quantization parameter Q_j for j-th MB is directly proportional to the fullness of the virtual buffer. If the number of encoded bits of previous j MBs is larger than the average rate, then the fullness of the virtual buffer increases which results in a large Q_j. A large Q_j represents a coarse quantization, and hence a low bitrate.

Step 3. Adaptive quantization

The reference quantization parameter calculated from the step 2 is further fine-tuned based on the spatial activity of each MB. The spatial activity is represented by the variance of pixel values. A high activity value results in coarse quantization and low bitrate.

Note that TM5 is just an estimation process. The true bitrate may be different from the estimation result. However, the difference of the bitrate is added to the bit allocation of the next GOP. Hence the average bitrate approaches to the target rate in the long run.

2.3.2 Non-Tagging Rate Control

Cell loss may seriously degrade the reconstructed video quality. To avoid cell loss in using TM5, we can either reserve a high bandwidth for a peak rate or use a large buffer. However, it needs extra cost to implement these ideas. For a reasonably large buffer size, the burstiness of video traffic must be properly controlled to avoid the generation of tagged cells [11], [12]. Non-tagging rate control is modified by Luo from MPEG-2 TM5 rate control [13]. The objective is to enforce the bitstream to have no tagged cells at all. It modifies the step 1 of TM5 by reassigning the target bit allocation for the current frame [16]. The new target bit allocation must be less than or equal to the maximum transmission rate $R_{\max}$, which is calculated as in Eq. (1)

$$R_{\max} = (B - B_{n-1}) + R \tag{1}$$

where B is the buffer size, B_{n-1} is the part of buffer space which has been used to store data at the end of frame $n-1$, and R is the token rate. Thus $(B - B_{n-1})$ is the remaining space of buffer.

As long as the target bit allocation of each frame is less than or equal to the maximum transmission rate, the buffer overflow will not happen, and no cells will be tagged. However, the video quality may be deteriorated due to insufficient peak rates, and the degraded quality may propagate to other frames in a GOP. In other words, although non-tagging rate control is almost free from cell loss, the video quality degradation may still be serious.

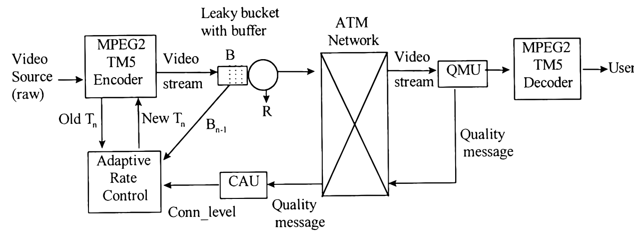


Fig. 2 AQC system model.

3. Adaptive Quality Control System

We propose an adaptive quality control (AQC) system for MPEG video in this section. In addition to the traditional source rate control, the connection status monitored at the decoding end is also used to control the quality and the rate. We first give an overview of the system, then describe each major part of the system in the rest of this section.

3.1 System Architecture

Figure 2 shows the block diagram of the AQC system architecture. At the encoding end, the TM5 video encoder compresses the video source and generates VBR video streams. The leaky bucket with buffer performs the traffic conformance and smoothing. The adaptive rate control calculates the bit allocation (New T_n) for each frame based on the information of the target bit allocation (Old T_n) for the frame, the buffer state (B_{n-1}), and the connection-level (Conn_level) generated by the connection-level adjustment unit (CAU). At the decoding end, in addition to the TM5 video decoder, the quality monitoring unit (QMU) counts the number of errors in the received data stream and sends the quality status back to the CAU at the encoding end. The CAU dynamically adjusts the connection-level based on the quality message that reflects the network status. The connection-level is raised if it receives a good quality message. Otherwise, it is lowered.

The video quality is affected by the network status significantly. If the network is congested, tagged cells may be discarded. As a result, the video quality will be deteriorated. The issue of congestion control is not in the scope of this paper. The focus of this paper is the rate control at the encoder. When congestion exists, the tagged cells must be reduced to avoid cell loss.

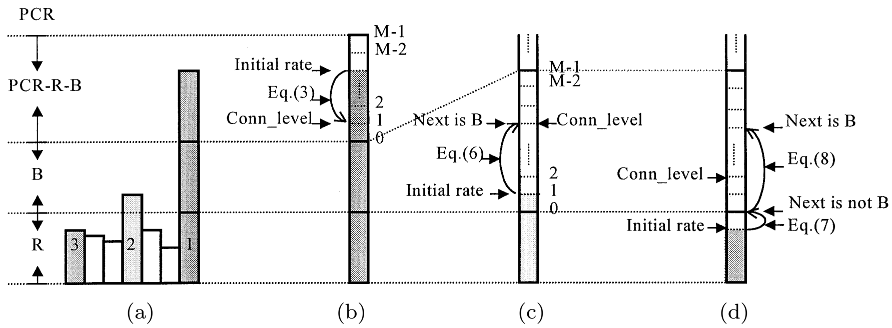


Fig. 3 Adaptive rate control in different buffer conditions. (a) Three buffer conditions (b) Case 1: overflow (c) Case 2: partial full (d) Case 3: underflow.

3.2 Adaptive Rate Control

From the discussion in the last section, TM-5 may generate a large amount of tagged cells, which may be dropped in congested conditions. On the other hand, non-tagging rate control may have significant quality degradation due to insufficient peak rates. Particularly, the rate reduction occurs most significantly in I-frames and results in serious error propagation. Thus, both methods may leave rooms for improvements.

In this work, we propose an adaptive rate control (ARC) algorithm. It modifies TM5 rate control method by considering more factors that include the fullness of the virtual buffer, the next frame type, and the network status. A virtual buffer is used as an indicator for bit allocation. More bits will be allocated to a frame if the buffer is closer to empty. In addition, the ARC method also allows cells to be tagged when the network condition is good. This results in consistent video quality regardless the frame type. Note that any overflow of the virtual buffer does not necessarily result in cell drops in the actual transmission because some of the cells will be tagged and transmitted in an interleaving order if the overflow happens. When the network status is poor, i.e. congestion exists, any tagged cells may be dropped and the video quality will be seriously degraded. Hence, we define the connection-level to represent the network status and use the connection-level as the indicator of allowance of tagged cells. A high connection-level reflects good network status and allows a large number of tagged cells.

In MPEG, I, P, and B frames appear in a predefined pattern periodically. Different frame types require different bitrates. We take the next frame type into the consideration. If the next frame is a B-frame, then extra bits can be allocated to the current frame since these extra bits can be delivered in the next frame period. However, if the next frame is an I or P frame, the bit allocation for the current frame should not be increased. The bit allocation strategy for each frame of ARC is discussed in three cases based on the buffer status, shown as in Fig. 3.

Case 1. Buffer overflow
Figure 3(b) shows an example of buffer overflow. In this case, the target bit allocation T_n of TM5, indicated as the initial rate in the figure, for the current frame is too large to be transmitted in this frame period. The untransmitted data can not be completely stored in the buffer since the buffer is full. This situation can be expressed by $T_n \geq R+B-B_{n-1}$, where is R the token rate which is the bandwidth that the network allocates, B is the buffer size, and B_{n-1} which is not plotted in the figures for simplicity is the buffer space occupied at the end of frame $n-1$. With the tagging policy, the overflowed data are then tagged and transmitted with no guarantee.

The ARC allows tagged cell to exist when the network is in a good condition in which the tagged cells are not likely to be dropped. Depending on the network status, represented by the connection-level, the number of allowed tagged cells, $Cell_{CLP}$, is calculated as in Eq. (2)

$$Cell_{CLP} = (PCR - R - B) * (Conn\_level/M) \quad (2)$$

where is the peak cell rate defined in the traffic contract, $Conn\_level$ is the connection-level, and M is the total number of connection-levels. Thus the maximum number of allowed tagged cells $PCR-R-B$ is divided into M units, and a higher connection-level results in more allowed tagged cells. The new target bit allocation $T_n^{'}$ for frame n is then calculated as in Eq. (3)

$$T_n = \min(T_n, (B - B_{n-1} + R + Cell_{CLP})) \quad \text{for all types of frames} \quad (3)$$

where T_n is the original target bit allocation for frame n, and $(B - B_{n-1}) + R$ is the maximum transmission rate without tagging. In other words, if the difference between the target bit allocation and the maximum transmission rate is less than the maximum number of allowed tagged cells, the target bit allocation is not changed. Otherwise, the target bit allocation is reduced so that the number of tagged cells is within the allowable range.

Case 2. Buffer partial full
Figure 3(c) shows the case of buffer partial full, which

can also be described by Eq. (4).

$$R - B_{n-1} < T_n \leqslant R + B - B_{n-1} \quad (4)$$

In this case, the target bit allocation T_n is already smaller than the maximum transmission rate without tagging $R + B - B_{n-1}$. Thus all data in this frame are guaranteed to be delivered without tagging. However, some of them will be buffered and delayed.

To utilize the bandwidth more efficiently, we may increase the bit allocation T_n, if the next frame is expected to have a low bitrate, e.g., B-frame, and the connection status is good. The new target rate is calculated by Eq. (5)

$$T_n = T_n + c * B * (Conn\_level/M) \quad (5)$$

where c is a constant, $0 \leqslant c \leqslant 1$, which regulates the ratio of the number of allowed tagged cells to the buffer size. In addition, $T_n^{'}$ may exceed the maximum transmission rate without tagging, hence the final bit allocation $T_n^{''}$ should satisfy Eq. (6) to avoid cell loss.

$$T_n^{''} = \begin{cases} T_n, \text{ if next frame is not B frame} \\ \min(T_n^{'}, R + B - B_{n-1}), \\ \text{if next frame is B frame} \end{cases} \quad (6)$$

Case 3. Buffer underflow
Figure 3(d) shows the case of buffer underflow. If the target bit allocation T_n is less than the guaranteed transmission rate without buffering, i.e., $T_n < R - B_{n-1}$, all the data in the current frame as well as accumulated from previous frames can be delivered within the frame period. However, the channel capacity reserved for this connection may not be fully utilized in this case. To raise the bandwidth utilization, the target bit allocation is modified by Eq. (7) regardless the next frame is a B-frame or not

$$T_n^{'} = R - B_{n-1} \quad (7)$$

where $T_n^{'}$ is the new target bit allocation.

In this situation, the bandwidth is fully utilized and the buffer will be empty when the current frame is transmitted. Moreover, the target bitrate can be further increased safely if the next frame is a B-frame. In other words, some of the cells in the current frame may be stored in the buffer temporally. The extra data rate also depends on the network status because a poor network condition may still have the possibility that results in cell loss in some B-frames. The new target bit allocation $T_n^{''}$ is calculated as in Eq. (8), which is similar to Eq. (5).

$$T_n^{''} = \begin{cases} T_n^{'}, \text{ if next frame is not B frame} \\ T_n^{'} + c * B * (Conn\_level/M), \\ \text{if next frame is B frame} \end{cases} \quad (8)$$

Since $0 \leqslant c \leqslant 1$, $T_n^{''}$ is bounded by the maximum transmission rate without tagging $R + B - B_{n-1}$. Namely, all data in this frame can be delivered safely.

3.3 Quality Monitoring

The connection status is monitored by the quality monitoring unit (QMU) at the decoding end. The QMU collects and sends back the data of the connection quality to the encoder for adaptive rate control. The basic processing unit in time is a frame period. The connection status can be represented by two measurements, the cell loss ratio (CLR) from the perspective of the network, and the slice error rate from the perspective of the video.

3.3.1 Cell Loss Ratio (CLR) Monitoring

CLR is an important parameter in the evaluation of the network quality. It can be measured either in the ATM adaptation (AAL) layer or in the ATM management plane. We use AAL-5 to deliver video streams in this work. The length in the trailer of an AAL-5 packet indicates the data length. Any cell loss could be easily detected by comparing the transmission length and the number of actually received cells.

The performance management function in ATM management plane is also able to measure the CLR. The total user cell count field in OAM&P (operation, administration, maintenance, and provisioning) cells records the number of transmitted cells since the last OAM&P cell. The cell loss can be calculated by comparing this with the number of actually received cells.

3.3.2 Slice Error Rate Monitoring

The network status is also reflected in the video quality. Considering the spatial error propagation, the slice is chosen as the unit in measuring the errors. In most cases, the decoder is able to detect the errors in the received stream, e.g. undecodable VLC, incorrect number of DCT coefficients, etc.

3.4 Connection-Level Adjustment

The connection-level adjustment unit (CAU) at the encoding end dynamically adjusts the connection-level based on the error rate measured by the QMU at the decoder. The CAU could be combined with the QMU at the decoder. However, a separate scheme is more robust because it also works in a multicasting environment.

The delay in the network and the video processing may cause a problem that the connection-level is not able to reflect the current network status and results in incorrect rate control. To reduce the impact of the instantaneous change, the connection-levels are adjusted once for n consecutive frames, where n is a parameter determined by the network size and the delay.

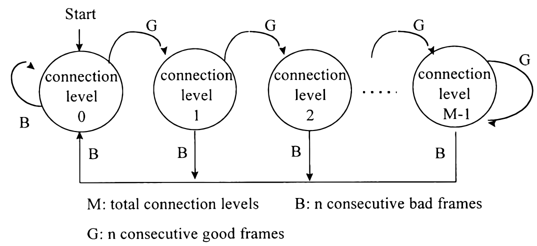


Fig. 4 Quality adjustment state diagram.

The state diagram of the connection-level adjustment is shown in Fig. 4. A good frame means the CLR or slice error rate in a frame is less than a threshold. If there exist n consecutive good frames, a state transition to the next higher connection-level occurs. On the contrary, n consecutive bad frames may result in a big transition to the lowest connection-level to avoid cell loss.

4. Simulation Results

The simulations are carried out under the conditions that the untagged cells will not be dropped and the peak signal-to-noise ratio (PSNR) between the reconstructed and original images is used as an objective image quality measure. The video sequences "Football" and "Garden" with CIF sequence format (24 fps, 352 × 240 pels, 4:2:0 chrominance format, 15 slices per picture) are MPEG coded with 12 pictures per GOP and one slice per MB row. The feedback channel is assumed to be error-free in the simulations. The cell rate used in simulations are calculated based on the capacity of 48 bytes per ATM cell. In the actual transmission, each frame will form a Packetized Elementary Stream (PES) packet which consists of Transport Stream (TS) packets with 188 bytes in length. The TS packet sequence is then AAL-5 segmented and transmitted. Thus, an overhead of 2.4% to 3.2% in rate should be added to accommodate the header and the stuffing bits in the actual transmission. The parameters of the rate control system for simulations are determined as follows.

- Sustained cell rate (SCR) or the average rate which is used for TM5 rate control. It is set to be 120 cells/frame period.
- Peak cell rate (PCR). It is typically two to three times of SCR for video signals. We choose two times, i.e., 240 cells/frame period.
- Effective bandwidth. It is the bandwidth the ATM network should reserve for this connection, usually between PCR and SCR. It is mainly affected by SCR, PCR, and the maximum burst length. In the extreme case, the effective bandwidth is equal to SCR if the traffic is CBR. In simulations, it is set to be 150 cells/frame period.
- Buffer size. A large buffer size may generate a smooth traffic but with the cost of large delay. For a given average delay and a maximum point-to-point delay, the buffer size B of the leaky bucket is calculated as in Eq. (9) [15].

$$B = (D_{\max} - D) * r \tag{9}$$

where $D_{\max}$ is the maximum point-to-point delay, D is point-to-point delay, and r is the average bits/frame. In simulations, the buffer size is set to be 30 cells.

- Delay from the QMU to the CAU is one frame.
- Total number of connection-levels M is set to be five.
- Network status is represented by the cell loss ratio (CLR) for tagged cells. The threshold of CLR to be a good frame and n in Fig. 4 are both set to be 1. Namely, a single cell loss will result in a bad frame and a bad frame will reduce the connection level to the lowest.

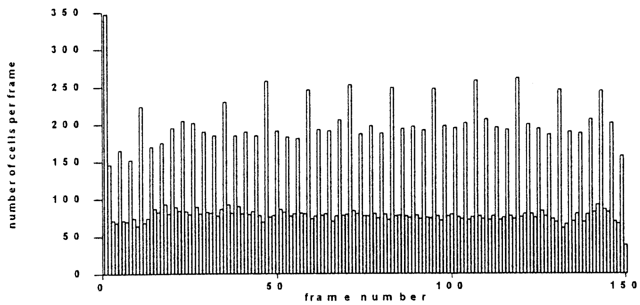
4.1 Comparisons of Rate Control Methods

We compare the output bitrates of three rate control methods, TM5, non-tagging, and adaptive rate control used in this system. Figures 5(a), (b), (c), and (d) show the bitrates, represented by the number of cells per frame, before the buffer of TM5, non-tagging, ARC with connection-level 0 (ARC-0), and ARC with connection-level 4 (ARC-4), respectively.

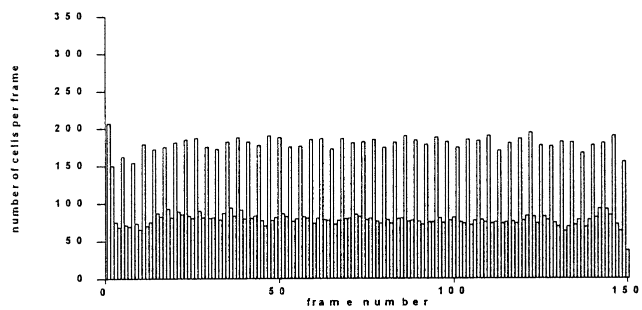
The traffic is further smoothed by the buffer. The bitrates of the buffer output are shown in Figs. 6(a), (b), (c), and (d), respectively. In these four cases, TM5 shows serious burstiness, even after the buffer. The bandwidth utilization of TM5 is low at B-frames. The non-tagging method controls the rates of I and B frames to be less than the effective bandwidth. At B-frames, the bandwidth is still under-utilized. The adaptive rate control method increases the rate of B-frames to raise the utilization and regulates the rate of I-frames based on the connection-levels. In the case of ARC-0, which is used in a poor connection status, similar to the non-tagging method, it is controlled to have no tagged cells. While in ARC-4, which represents a good connection, some tagged cells are allowed in I-frames for delivering video with better quality. Thus ARC has the highest bandwidth utilization while the non-tagging method has the lowest utilization. A VBR traffic with high burstiness generally needs a high effective bandwidth to deliver the traffic with QoS guatantee. Here AQC intends to generate guaranteed CBR. Although the average rate is higher than the non-tagging method, the effective bandwidth that the network should provide is still the same.

4.2 Performance of AQC System

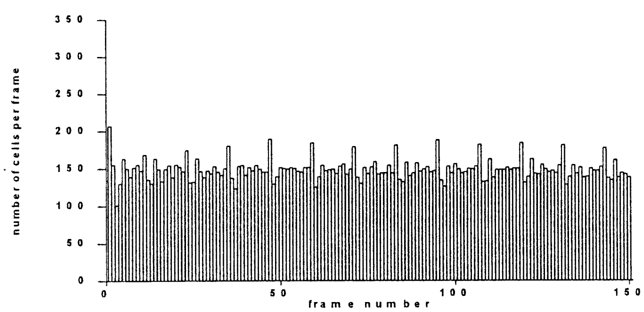
The performance of the adaptive quality control system is compared with TM5 and non-tagging methods. In



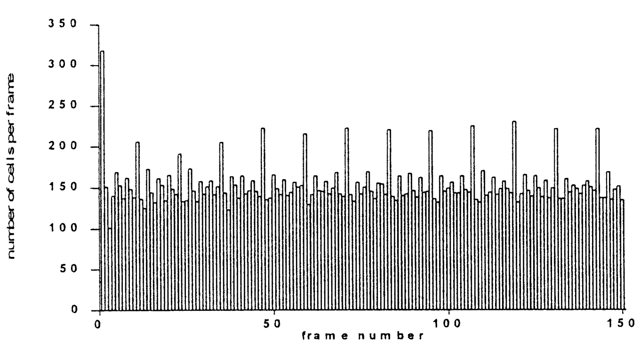
(a) TM5



(b) Non-tagging

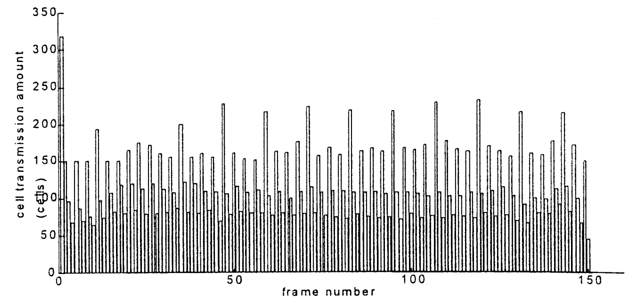


(c) Adaptive rate control (level 0)

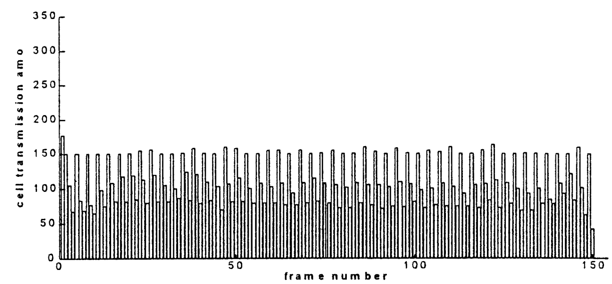


(d) Adaptive rate control (level 4)

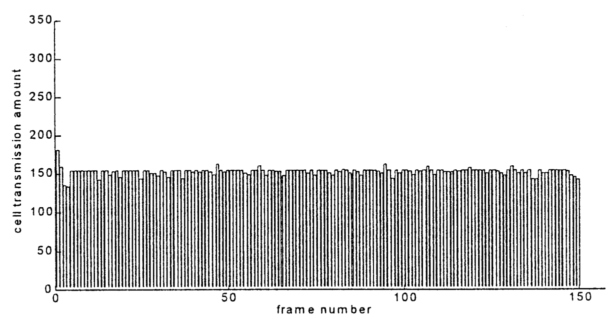
Fig. 5 Coding rates (before the buffer).



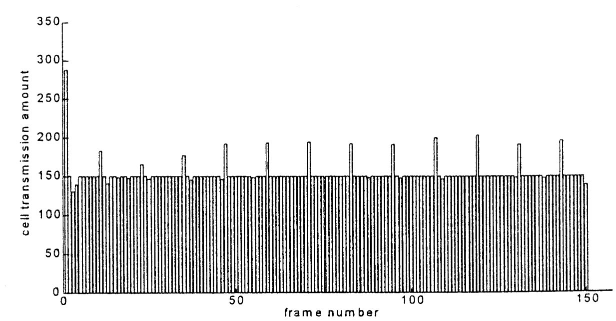
(a) TM5



(b) Non-tagging



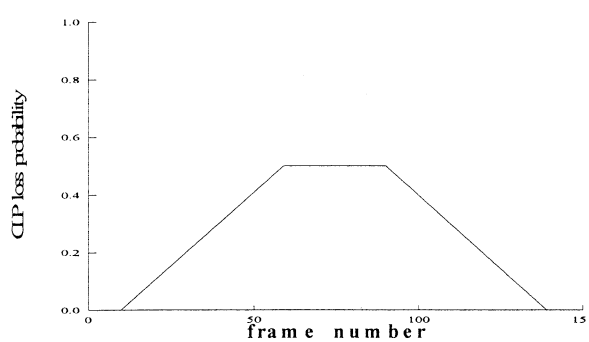
(c) Adaptive rate control (level 0)



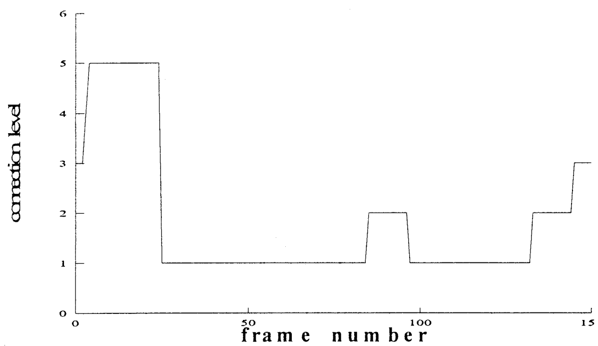
(d) Adaptive rate control (level 4)

Fig. 6 Transmission rates (after the buffer).

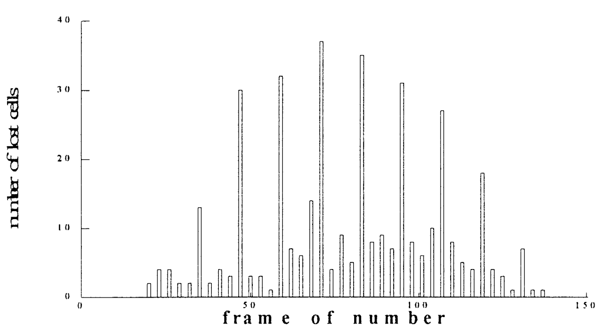
AQC system, the connection-level is adjusted automatically based on the cell loss ratio. Figure 7(a) simulates the cell loss probability of tagged cell in a congested network condition. In the period from frame 60 to frame 90, 50% of tagged cells are assumed to be dropped. Figure 7(b) shows the change of the connection-level corresponding to the CLP variation. A light congestion results in a sudden downgrade of the connection-level. Figures 7(c), (e), and (g) show the number of lost cells in each frame for the three rate control methods, respectively. The PSNRs of three methods are depicted in Figs. 7(d), (f), and (h), respectively. It clearly shows that any cell loss results in serious video



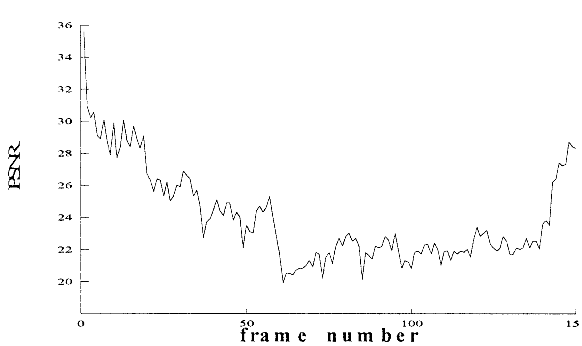
(a) Cell loss probability



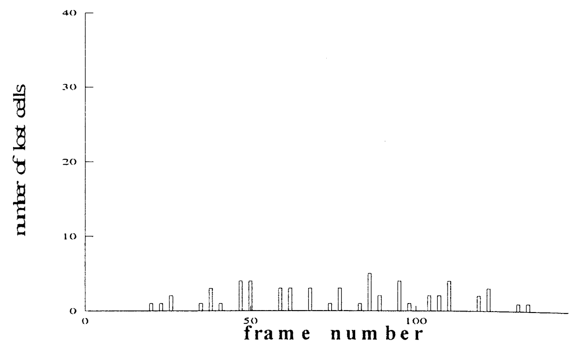
(b) Connection level of AQC



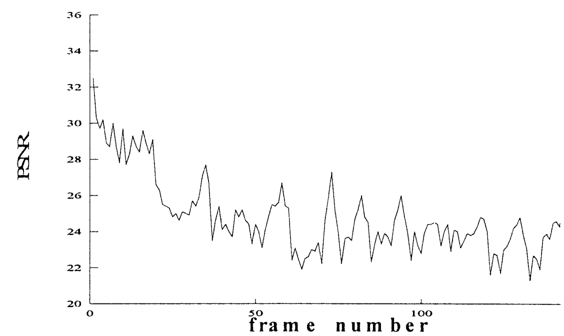
(c) Number of lost cells in TM5



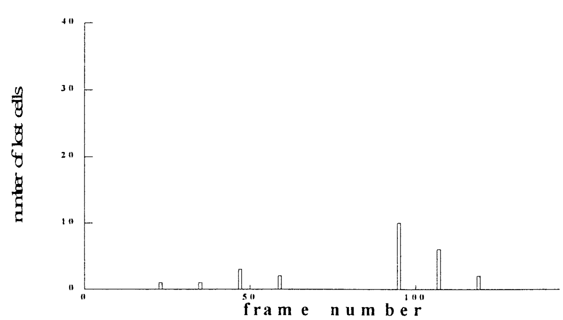
(d) PSNR of TM5



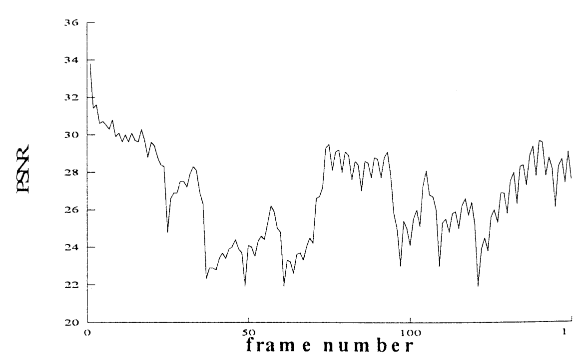
(e) Number of lost cells in Non-tagging



(f) PSNR of Non-tagging



(g) Number of lost cells in AQC



(h) PSNR of AQC

Fig. 7 Cell loss and PSNR in a heavily congested network condition.

Table 1 Overall performance in a heavily congested network condition.

	CLP (cells)	loss (cells)	PSNR (dB)
TM5	1369	380	23.95
Non-tagging	170	58	24.96
AQC	216	25	26.76

quality degradation, and the damage may propagate to the whole GOPs. Table 1 shows the overall performance. The TM5 method has the most serious cell loss problem, which generate substantial PSNR degradation. Note that although the objective of non-tagging method is to have no tagged cells, it may actually generate tagged cells due to incorrect estimation of the coding rate. AQC has the least lost cells due to the adaptive scheme. Thus the overall performance of AQC, represented by the average PSNR (26.76 dB), is significantly better than non-tagging method (24.96 dB) and TM5 (23.95 dB).

5. Conclusion

We have presented an adaptive quality control system, which utilizes the network condition as well as the MPEG coding characteristics. Higher bitrates are allocated to I-frames if the connection status is good. The extra cells are transmitted with tagging. The overall performance of AQC is better than existing rate control methods.

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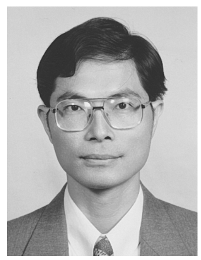

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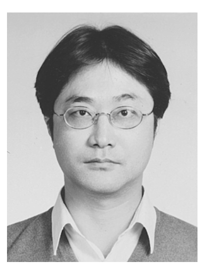

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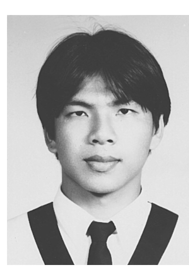

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