An Adaptive Rate Control Algorithm for RCBR Transmission of Streaming Video

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ABSTRACT

This paper presents an adaptive H.263+ rate control algorithm for streaming video applications under the networks supporting bandwidth renegotiation, which can communicate with end-users to accommodate their time-varying bandwidth requests during the data transmission. That is, the requests of end-users can be supported adaptively according to the availability of the network resources, and thus the overall network utilization can be improved simultaneously. They are especially suitable for the transmission of non-stationary video traffics. The proposed rate control algorithm communicates with the network to renegotiate the required bandwidth for the underlying video which are measured based on the motion change information, and choose their control strategies according to the renegotiation results. Unlike most conventional algorithms that control only the spatial quality by adjusting quantization parameters, the proposed algorithm treats both the spatial and temporal qualities at the same time to enhance human visual perceptual quality. Experimental results are provided to demonstrate that the proposed rate control algorithm can achieve superior performance to theconventional ones with low computational complexity under the networks supporting bandwidth renegotiation.

I. Introduction

Recently, the demands and interests on video communication are growing rapidly, and the video data is expectedto be the most significant component among the multimedia traffics over the network. However, it is not a simple problem to transmit video traffics reliably through the network because the video requires a large amount of data compared to other multimedia such as speech, audio and text. Furthermore, the generic characteristics of video traffics are very burst, which makes the problem more difficult and challenging.

The digital communication and network technologies have been developed very fast during the last decade, and both ATM (asynchronous transfer mode) and Internet are widely adopted for the transmission of video data. Since it can support VBR (variable bit rate) and QoS (quality of service), ATM is known to be more suitable for the transmission of video traffics, and, so far, considerable efforts have been devoted to increase the ATM network utilization and improve the video quality at the decoder simultaneously [1,2,3]. However, since the network resources are limited and their utilization can be decreased substantially, it is almost impossible to accommodate the unconstrained VBR video traffics for multiple users. In recent years,the role of Internet for video/multimedia communication becomes more and more important as the number of the Internet users (or servers/clients) increases exponentially. RTP/RTCP based on UDP/IP can be efficiently used to treat real-time applications such as Internet video/audio streaming and video conferencing [4,5,6], while TCP/IP is widely employed for non-real-time applications such as FTP (file

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transfer protocol), SMTP (simple message transfer protocol) and HTTP (hyper-text transfer protocol), etc. Note that Internet does not provide the guaranteed QoS, but just the best-effort service. Therefore, the end-users must control their output bit-stream to prevent serious packet losses and traffic congestions over the Internet. Moreover, every output bit-stream must be adjusted to share the available channel bandwidth fairly with other TCP and UDP traffics [7,8,9,10].

As mentioned earlier, since the amount of video data is enormous compared to other multimedia data, it is indispensable to employ effective video compression algorithms for the video/multimedia systems. Recently, digital video coding techniques have advanced rapidly. International standards such as MPEG-1, 2 and 4, H.261, H.263/+/++ and H.26L have been established or under developmentto accommodate different needs by ISO/IEC and ITU-T, respectively. Among various video-coding standards, H.263+ [14] is the emerging state-of-the-art low bit video compression technique for Internet video transmission, of which core ingredients include the block-based motion compensation and the block-based DCT coding. And also, the rate control algorithm plays a crucial role in videotransmission. It regulates the output bit-stream to meet certain given conditions, as well as enhances the quality of coded video. However, the rate control algorithms are not standardized since they are independent on the decoder structure.

In general, suitable communications between the network and video encoders can increase the network utilization and enhance video quality at the same time [11]. In recent years, thus some bandwidth-renegotiation approaches have been proposed to handle the non-stationary video traffics efficiently over the network. Under the networks supporting bandwidth renegotiation, the video encoder can renegotiate the allowable bandwidth during the transmission. RCBR (renegotiated CBR) [12] is proposed as a simple but quite effective approach to support the renegotiations, while it still needs to be improved to work reliably in real network environments. It can be implemented in both ATM network and the Integrated Service Internet by using the resource management (RM) cell mechanism,and the refreshment of the network reservation state using the RSVP (resource reservation protocol) signaling protocol, respectively [12,13]. Actually, the bandwidth renegotiations can be interpreted as a compromise of ABR (available bit rate) and VBR. Before requesting renegotiation, the video encoder has to estimate the amount of the required channel bandwidth for the underlying video, and also has to decide when to renegotiate. Then, a signaling message requesting an increase or decrease of the channel bandwidth is sent from thevideo encoder to the network. Finally, the network resource management system decides whether the request can be accepted or not according to the current network situations. Note that, in general, more renegotiations can increase the network utilization, however they may cause larger signaling overhead.

In this work, we present a new adaptive H.263+ rate control algorithm for streaming video applications under the networks supporting bandwidth renegotiation. In order to explain the structures and features of the proposed H.263+ based rate control algorithms clearly, let us remark the difference between MPEG and H.263+ rate control algorithms. One of the major differences between the two is the encoding frame structure. The GOP (group of pictures) of MPEG is composed of an I-frame and several predictive P- and B-frames, and this GOP pattern is repeated periodically, so that it is widely used as a basic control unit. In contrast, since H.263+ is mainly intended for low bit rate applications, the number of I-frames is relatively very small, so that the number of P-frames between adjacent I-frames becomes very large, resulting the default encoding frame structure of H.263+ to be ***IPPPPP--*** even though PB-mode is supported as an annex. Thus, HHGOP is not suitable to be used as a basic control unit for H.263+ due to the high computational complexity and long time-delay. Thus, most of the existing H.263+ rate control algorithms including TMN8 [14, 15] focus on macroblock-layer rate control forthe predictive frames [25, 26, 27, 28, 29, 30]. However, we cannot treat the videos efficiently with only macroblock-layer rate control. Thus the proposed algorithm includes the required bandwidth estimation scheme, channel bandwidth renegotiating process, encoding frame rate adjustment algorithm and frame-layer rate control algorithm. One unique feature of the proposed rate control algorithm is that they can renegotiate the bandwidth with the network if needed, and also control the spatial and temporal qualities simultaneously according to the renegotiation results. And TMN8 is employed as a component for the macroblock-layer rate control. In fact the proposed algorithm can be compatible with the above existing macroblock-layer rate control.

The paper is organized as follow. Rate control algorithm for streaming video applications is proposed in Section 2. The required channel bandwidth estimation, encoding frame rate adjustment and frame-layer rate control algorithms are discussed in these sections. Then, experimental results are presented in Section 4. Finally, concluding remarks are given in Section 5.

Ⅱ. Proposed Rate Control Algorithm for Streaming Video Applications

For streaming video applications, the renegotiation time schedule and the required bandwidth can be determined in advance to the video transmission since the video information is available at the encoder side. As mentioned earlier, since the basic frame structure of H.263+ is *IPPP---*, the characteristics of H.263+-coded traffic are relatively independent of the encoding frame types compared with MPEG, and the output bit-stream fluctuation of H.263+ is mainly resulted from the motion changes of underlying video. Among many quantitative measures of the motion change information [24], including *MAD* (mean of absolute difference), *SAD* (sum of absolute difference), *DOH* (difference of histograms), *HOD* (histogram of difference), *BH* (block histogram difference) and *BV* (block variance difference), *MAD* and *SAD* are employed in this work.

Basically, the proposed rate control algorithm consists of three parts: the required bandwidth estimation scheme based on motion changes, the encoding frame rate adjustment scheme, and the frame layer rate control scheme with low computational complexity

1. Required Bandwidth Estimation

A motion-based required bandwidth estimation scheme is presented in this section. Video encoder estimates the required bandwidth to keep the temporal and spatial qualities of the underlying video above a tolerable bound, and attempts bandwidth renegotiations with the network periodically. The renegotiating time interval is dependent on several factors including the encoding time-delay, the network utilization and signaling overhead. That is, as the renegotiating time interval becomes smaller, the encoding time delay is reduced and the network utilization can be improved, while the signaling overhead is increased, and vice versa. The relation between the network efficiency and the renegotiating time interval is shown in [12, 13]. Let us definethe bundle of frames during the renegotiating time interval as *the temporal frame segment*. Since the characteristic of the rate-distortion of predictive frames is greatly related to the motion changes of the underlying video, the required bandwidth for current video can be reasonably estimated by using the motion change information in the temporal frame segment as follow.

$$BW_{k,req}^{SEG} = \left(1 + \frac{MC_k - MC_{ref}}{MC_{ref}}\right) \cdot BW_{ref}, \quad (1)$$

where BW_{ref} and $BW_{k,req}^{SEG}$ are the bandwidth of the reference window and the required bandwidth of the k_{th} temporal frame segment, respectively, and

$$MC_{ref} = \frac{1}{T_{ref}} \sum_{i=0}^{N_{ref}-2} SAD(f_{ref,i} - f_{ref,i+1}),$$

$$MC_k = \frac{1}{T_k^{SEG}} \sum_{i=0}^{N_k^{SEG}-2} SAD(f_{k,i} - f_{k,i+1}),$$

where $f_{ref,i}$ and $f_{k,i}$ are the i_{th} frames of the reference window and the k_{th} temporal segment, T_{ref} and T_k^{SEG} are the time intervals of the reference window and k_{th} temporal segment, and N_{ref} and N_k^{SEG} are the encoding frame numbers in the reference window and k_{th} temporal segment. The above reference window is just a time interval, and the ratio of the motion change and the bandwidth in the interval is kept same during the whole encoding process to make the video quality almost same. Let us assume that if the network is able to accommodate the required bandwidth of the current video, the request is accepted by the network, otherwise it is rejected, so that the bandwidth is not changed. In order to enhance the network utilization, we also assume that the renegotiation requests are always accepted if the required bandwidth is less than the current bandwidth, which is the case when the bandwidth is decreasing due to slow motion changes in video. The negotiation rule can be summarized as follow.

$$BW_k^{SEG} = \begin{cases} BW_{k,req}^{SEG} & \text{if } BW_{k-1}^{SEG} > BW_{k,req}^{SEG}, \\ BW_{k,req}^{SEG} & \text{if } BW_{k-1}^{SEG} < BW_{k,req}^{SEG} \text{ and the request is accepted,} \\ BW_{k-1}^{SEG} & \text{otherwise,} \end{cases} \qquad (2)$$

where BW_{k-1}^{SEG} and BW_k^{SEG} are the bandwidth of the $(k-1)_{th}$ and k_{th} temporal segments.

2. Encoding Frame Rate Adjustment Scheme

Note that when the renegotiations fail, the quality of each frame in the spatial domain can be deteriorated below a certain tolerable bound. Moreover, since the previous P-frames are used as the reference frame for the following P-frames, the degradation is propagated to the following frames. As a result, the overall human visual perceptual quality of thevideo can be degraded substantially. In such case, by utilizing the trade-off between the spatial and temporal qualities in a video adequately, we can prevent the extreme degradation of the spatial quality and maintain the perceptual visual quality within the tolerable range. Thus, in this work, we have employed following encoding frame rate adjustment scheme to control the spatial quality adaptively.

$$N_k^{SEG} = N_{ref} - \left\lceil \left(\frac{BW_{k,req}^{SEG}}{BW_{ref}} - 1 \right) \cdot \alpha \cdot N_{ref} \right\rceil, \qquad (3)$$

where α is a weighting factor and $\lceil x \rceil$ means the smallest integer greater than x. Note also that the time intervals between adjacent encoded frames must be same to minimize the motion unsmoothness in each temporal frame segment. However, human visual perceptual quality can be a little improved by finely adjusting the encoded frame positions in the boundary between temporal frame segments with different the encoded frames numbers [18, 23].

3. Frame-layer Rate Control with Low Computational Complexity

In this Section, we propose a frame-layer rate control algorithm with low computational complexity. The temporal framesegment is used as a basic unit for the control of the frame-layer rate. By assuming that the temporal frame segments are independent each other like the GOPs of MPEG, we can reduce both the encoding time delay and the computational complexity. We can formulate the frame-layer rate control problem as the following optimization problem.

$$\text{Determine } \bar{q}_i, i = 1,2,\cdots,N_k^{SEG} \text{ to minimize } \sum_{i=1}^{N_k^{SEG}} D_i(\bar{q}_i), \qquad (6)$$

$$\text{subject to } \sum_{i=1}^{N_k^{SEG}} R_i \le BW_k^{SEG} \cdot T_k^{SEG},$$

The above optimization problem with constraint can be solved by the Lagrangian multiplier method [20,21], in which the cost function and the constraint are combined by a Lagrange multiplier. However, a high computational complexity and long encoding time delay is needed to find the optimal solution. In order to solve this problem, we propose an efficient iterative algorithm that guarantees a low computational complexity, in which the two steps, i.e. the optimization process and the Lagrange multiplier adaptation process work iteratively to meet the bit rate constraint. The required computational complexity of the proposed iterative scheme is very low since the rate and distortion models are adopted for the optimization process, and no repeated process is required to find the optimal Lagrange multiplier in the Lagrange multiplier adaptation process. The frame-layer R-D (rate-distortion) model and the proposed fast frame-layer rate control algorithm based on the R-D model are described in the following Sections.

1) Frame-layer R-D modeling scheme

The R-D modeling techniques are essential for developing fast rate control algorithms. These can be categorized into two approaches: statistical modeling techniques and empirical databased modeling techniques. In this work, we employ an empirical databased frame-layer R-D model using quadratic ratemodel [19] and affine distortion model with respect to the average QP (quantization parameter) in a frame, which is given by

$$\hat{R}(\overline{q}_i) = (a\overline{q}_i^{-1} + b\overline{q}_i^{-2}) \cdot MAD(\hat{f}_{ref}, f_{cur}),$$
$$\hat{D}(\overline{q}_i) = a'\overline{q}_i + b', \tag{4}$$

where a, b, a' and b' are the model coefficients, $\hat{f}_{ref}$ is the reconstructed reference frame at the previous time instant, f_{cur} is the uncompressed image at the current time instant, $MAD(\cdot,\cdot)$ is the mean of absolute difference between two frames and $\overline{q}_i$ is the average QP of all macroblocks in the i_{th} frame. The model coefficients are determined by using the rate-distortion table obtained from the previous encoding results as follow.

$$b = \frac{N\left(\sum_{i=1}^{N} R_i\right) - \left(\sum_{i=1}^{N} R_i\overline{q}_i\right)\left(\sum_{i=1}^{N} \overline{q}_i^{-1}\right)}{N\left(\sum_{i=1}^{N} \overline{q}_i^{-2}\right) - \left(\sum_{i=1}^{N} \overline{q}_i^{-1}\right)^2}, \qquad a = \frac{\sum_{i=1}^{N}\left(R_i\overline{q}_i - b\overline{q}_i^{-1}\right)}{N},$$
$$a' = \frac{\left(\sum_{i=1}^{N} D_i\right)\left(\sum_{i=1}^{N} \overline{q}_i\right) - N\left(\sum_{i=1}^{N} D_i\overline{q}_i\right)}{\left(\sum_{i=1}^{N} \overline{q}_i\right)^2 - N\left(\sum_{i=1}^{N} \overline{q}_i^{2}\right)}, \qquad b' = \frac{\sum_{i=1}^{N} D_i - a\sum_{i=1}^{N} \overline{q}_i}{N}, \tag{5}$$

where N is the size of the rate-distortion table, D_i and R_i are the actual distortion and bit rates of the encoded i_{th} frame. In order to increase the accuracy of the R-D model, an outlier-removing algorithm is also adopted: If the difference between the estimated value by the models and a datum of the rate-distortion table is greater than a threshold, the datum is discarded, and then based on the refined data, the coefficients are re-calculated by the same method.

2) Optimization Process and Lagrange Multiplier Adaptation Scheme

As mentioned in the above, the proposed frame layer rate control algorithm consists of two iterative processes: optimization process and Lagrange multiplier adaptation scheme. The detailed description of the steps for the proposed frame-layer rate control algorithm is described as follow.

■ **Step 1: Optimization process based on the rate and distortion models**

$$P_i(\overline{q}_i) = \hat{D}_i(\overline{q}_i) + \lambda_i \cdot \max\{\hat{B}_i^{res}, 0\},$$
$$\hat{B}_i^{res} = \sum_{j=1}^{i-1} R_j + \hat{R}_i(\overline{q}_i) - \left(1 + \frac{MC_k^i - MC_k}{MC_k}\right) \cdot \frac{BW_k^{SEG} \cdot T_k^{SEG}}{N_k^{SEG}}, \tag{7}$$

where $P_i(\overline{q}_i)$ and λ_i are the cost function and the Lagrange multiplier for the i_{th} frame, respectively, R_j is the bit rate for the j_{th} frame, and MC_k^i is the motion change between $(i-1)_{th}$ and i_{th} frames of the k_{th} temporal frame segment. Note that as shown in Eq. 7, the target bit budget for each frame is established based on the motion change since the frames with faster motion change need more bit rates than those with slower one. The optimal average QP for the *ith* frame is then determined by

$$\overline{q}_i^* = \arg\min_{\overline{q}_i} P_i(\overline{q}_i). \quad (8)$$

■ **Step 2: Lagrange multiplier adaptation process**

Since the Lagrange multiplier should be updated on the fly, only a sub-optimal solution with a low computational complexity is considered. Thus, in this work, the adaptive adjustment rule proposed by T. Wiegand et al. is employed [22] which can be written by

$$\lambda_{i+1} = \lambda_i + \Delta\lambda, \qquad \Delta\lambda = \frac{B_{used,i}}{B_{target,i}} - 1, \quad (9)$$

where λ_i is the Lagrange multiplier for the i_{th} frame and

$$B_{used,i} = \sum_{j=1}^{i} R_j, \quad B_{target,i} = \sum_{j=1}^{i}\left(1 + \frac{MC_k^i - MC_k}{MC_k}\right)\cdot\frac{B_k^{SEG}\cdot T_k^{SEG}}{N_k^{SEG}}$$

The target bit rates are then assigned to macroblocks in a frame by macroblock layer rate control of TMN8.

Ⅲ. Experimental Results

In this experiment, the UBC H.263+ source codes [16] and the macroblock layer rate control of TMN8 is used for the implementation of the proposed algorithm. The performance evaluation has been made based on the subjective as well as the objective tests.

Note that the proposed rate control algorithm does not treat I-frames. However, since the H.26L Evaluation Delay Model User Guide recommends that the bit rate for the I-frame must not be greater than one second worth of bit transmission at the assumed channel bit rate, in this experiment, we have encoded the I-frames with QP=15 to meet this condition.

As mentioned earlier, unlike the MPEG case, since we cannotuse GOP as a unit of renegotiations in H.263+, a fixed renegotiating time interval is examined for the streaming applications, while various minimum values of the renegotiating time interval are examined for the interactive video applications. The test configuration of the experimental system is shown in Fig. 1, and the simple resource management rules of RCBR network are given in the following.

■ **Simple Resource Management Rules of RCBR network**

(1) If networks can accommodate the requested channel bandwidth under the current network situation, the renegotiation request is accepted. Otherwise, the renegotiation request is rejected and the bandwidth is not changed.

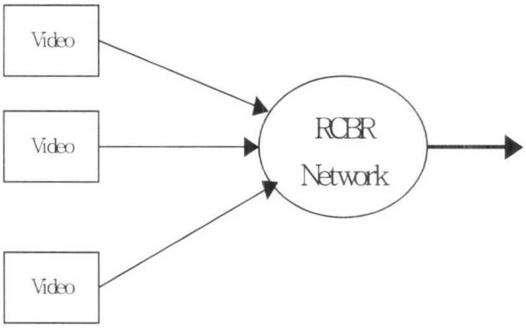

Fig. 1 Test configuration of the experimental system

Table 1. Performance comparison between TMN8 and the proposed rate control algorithm under RCBR. (Foreman)

Rate Control Method	Average PSNR	STDEV of PSNR
TMN8 under CBR	29.602	1.257
Prop. Algorithm under RCBR	30.015	0.906

Table 2. Performance comparison between TMN8 and the proposed rate control algorithm under RCBR. (Silent Voice)

Rate Control Method	Average PSNR	STDEV of PSNR
TMN8 under CBR	30.194	0.644
Prop. Algorithm under RCBR	30.166	0.602

Table 3. Performance comparison between the fixed encoding frame interval and the proposed encoding frame rate adjustment algorithm (Foreman)

Rate Control Method	Average PSNR	STDEV of PSNR	No. of Encoded frm
TMN8	29.574	1.093	144
Proposed Algorithm	29.925	1.029	128

Table 4. Performance comparison between the fixed encoding frame interval and the proposed encoding frame rate adjustment algorithm (Silent Voice)

Rate Control Method	Average PSNR	STDEV of PSNR	No. of Encoded frm
TMN8	30.112	0.629	139
Proposed Algorithm	30.401	0.525	121

(2) The renegotiation request is always accepted if the requested bandwidth is less than the current bandwidth.

(3) The minimum renegotiating time interval is set to control the signaling overhead.

"Foreman (QCIF)", and "Silent Voice (QCIF)" videos are employed for the test, whose average target rates are 48 kbps and 24 kbps, respectively. .In this experiment, by considering the signaling overhead as well as bandwidth utilization [13], we have set the fixed renegotiating time interval to be 1 second. α is set to 1. During the experiment, the proposed modeling works reasonably well. The bit-rate and PSNR plots of Foreman and Silent Voice encoded by TMN8 are shown in Fig. 2 and Fig. 6, respectively. We have estimated the required bandwidth by analyzing themotion change. The output bit rate plot and PSNR plot for the case when all renegotiation requests are accepted are given in Fig. 3 and Fig. 7, and their statistical data are summarized in Table 1 and 2. We note that, in this case, the proposed rate control algorithm reduces PSNR fluctuation by about 7% ~ 30% compared with TMN8.Moreover, the human visual perceptual quality is also clearly improved, especially when the motion changes are heavy.

When the traffic over the network is heavy, the renegotiation requests can be rejected. The experimental results forthis case are presented in Figs. 4, 5, 8 and 9. Note that the proposed encoding frame interval adjustment scheme works in this case. The available channel bandwidth is shown in Fig. 4 and 8, and the output bit rate plot and the corresponding PSNR plot aregiven in Fig. 5 and 9. And their statistical data are also given in Table 3 and 4. Form these results, we can observe that the average PSNR is improved a little bit, while the standard deviation of PSNR is reduced by about 6% ~ 10%, and the numbers of encoded frames is decreased by about 10% ~ 13%. The motion unsmoothnesscaused by the abrupt change of encoded frame number can be minimized by adjusting the time intervals near the boundaries of adjacent frames [18].

As a result, the proposed algorithm can improve the spatial quality and reduce the flickering artifact by sacrificing the motion smoothness unnoticeably.

Ⅳ. Conclusion

In this work, we have presented a new adaptive H.263+ rate control algorithm under the network supporting bandwidth renegotiation. The proposed algorithm estimates the required bandwidth by analyzing the motion change during a time interval and preformed the frame layer rate control based on the motion change of each frames. By the experimentalresults, it is observed that the human visual perceptual quality is kept almost constant when the bandwidth request is accepted, and its degradation can be reduced by adjusting the encoding frame number (or interval) when the request is rejected. The experimental results with real scenes show that the proposed bandwidth renegotiation and adaptive H.263+ rate control algorithm can serve as powerful techniques to transmit the non-stationary video traffics over the network. Compared with the conventional TMN8, the proposed algorithm improve the quality of the compressed video significantly in both objective and subjective tests. The performance analysis including RCBR network efficiency and the amount of control-signaling overhead will be our further research topics.

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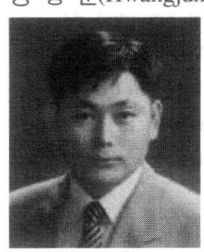
송 황 준(Hwangjun Song)



1990. 2 : 서울대학교 공과대학 제어계측과 졸업(공학사)
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<주관심 분야> Multimedia communication/signal processing, Packet video, Network protocols necessary to implement a functional real-time image/video application.

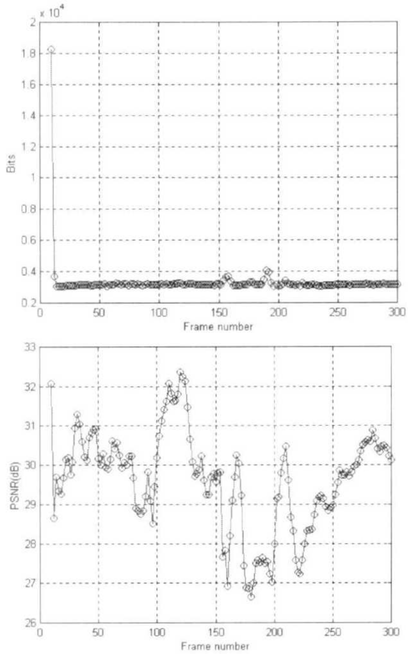

Fig. 2 Bit-rate and PSNR plots of Foreman encoded by TMN8 with 48 kbps(CBR). (Top) Bit-rate plot and (Down) PSNR plot

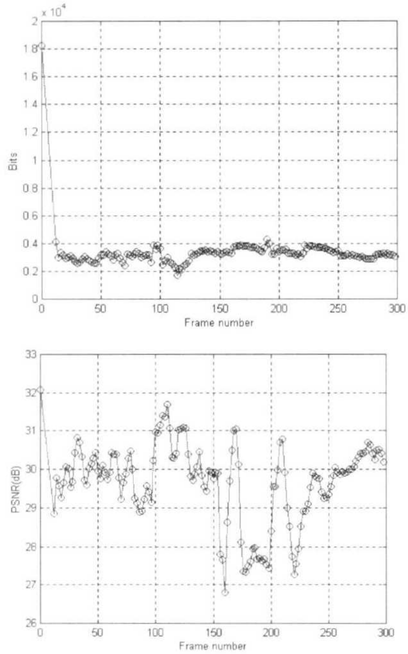

Fig. 3 Rate and PSNR plot when renegotiations always accepted. (Foreman)
(Top)Bit-rate plot and (Down) PSNR plot

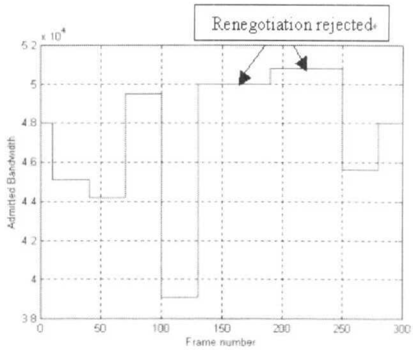

Fig. 4 Admitted bandwidth when renegotiations are rejected (Foreman)

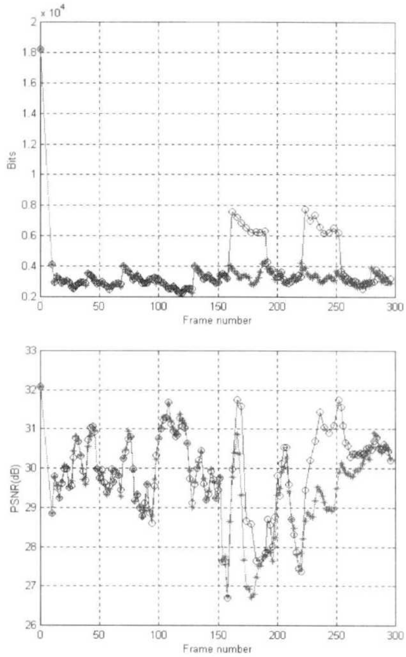

Fig. 5 Bit-rate and PSNR plots when renegotiations are rejected (Foreman).
(Top)Bit-rate plot and (Down) PSNR plot

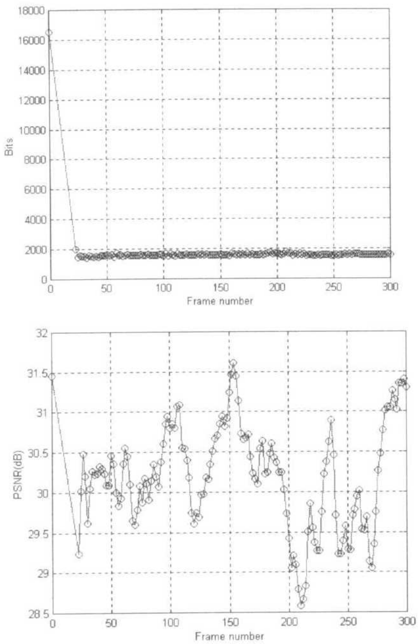

Fig. 6 Bit-rate and PSNR plots of Silent Voice encoded by TMN8 with 24 kbps(CBR). (Top) Bit-rate plot and (Down) PSNR plot

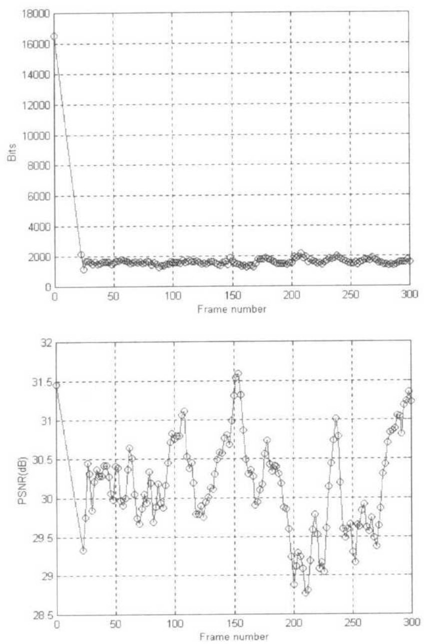

Fig. 7 Rate and PSNR plot when renegotiations are always accepted.(Silent Voice) (Top) Bit-rate plot and (Down) PSNR plot

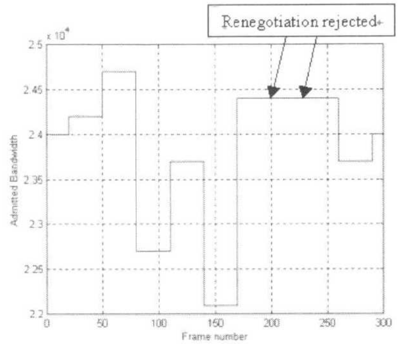

Fig. 8 Admitted bandwidth when renegotiations are rejected (Silent Voice)

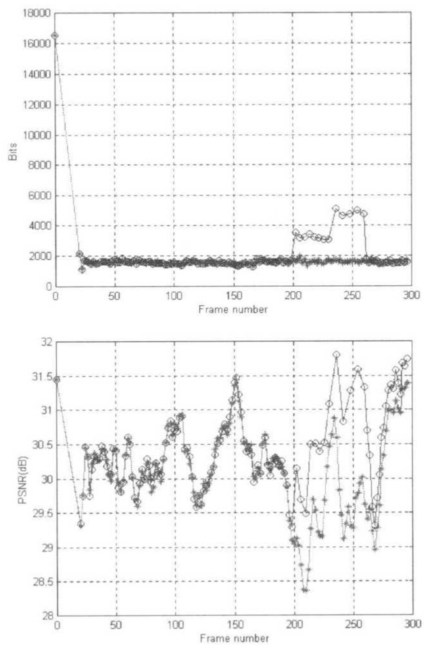

Fig. 9 Bit-rate and PSNR plots when renegotiations are rejected (Silent Voice). (Top) Bit-rate plot and (Down) PSNR plot