Adaptive MPEG-4 Source Rate Control over Internet for ADSL Subscribers

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Abstract

This paper proposes a mechanism that integrates a MPEG-4 adaptive source rate control algorithm and a bi-directional negotiation scheme over Internet for ADSL (Asymmetric Digital Subscriber Line) subscribers to maintain the quality of real time MPEG-4 applications. We introduce an IP/MPLS/POS (IP / MultiProtocol Label Switching / Packet Over SONET/SDH) network architecture to improve the packetization efficiency for MPEG-4 encoded bit-streams. This architecture can also reduce the delay and loss of feedback control packets. We show that the VOP (Video Object Plane) based packetization scheme has the best performance under the proposed network architecture. According to the network status, the proposed MPEG-4 adaptive source rate control algorithm adjusts the target quantization value instead of reducing the target encoding rate directly. Simulation results show that, with the three MPEG-4 video sequences and four types of Internet traffic conditions, the proposed mechanism always has the best performance. In summary, the higher the video complexity of MPEG-4 sequence, the more PSNR improvement the proposed mechanism; moreover, the higher the traffic load, the more PSNR improvement the proposed mechanism.

I. Introduction

With the emphasis on multimedia services in the next generation Internet, the video communication facilities will support a lot of multimedia applications such as telemedicine, real time news, movie streaming, and remote diagnosis. However, Internet video services encounter a number of technical challenges [1][2][9]. One of the issues is that Internet cannot provide QoS (Quality of Service) guarantee because of the inherent high variable network status during a communication period. Packet loss of a congested path ranges from single packet loss to burst losses or even intermittent loss of the connection [3][4]. However, for improving low bit rate video coding efficiency, the compressed video services are extremely vulnerable against packet losses because of error propagation. The previous encoded and reconstructed video frames are used to predict the next frame. Therefore, the loss of packets in one frame would produce considerable impact on the quality of the following frames [5][6].

MPEG-4 is an ISO/IEC standard developed by MPEG (Moving Picture Expert Group) [7][8]. It can integrate and synchronize the data associated with multiple objects so that they can be transported over network channels. MPEG-4 provides powerful error resilience capabilities, such as resynchronization, reversible variable length coding, header extension code, and data partitioning, to deal with the highly variable network environment. In addition, MPEG-4 provides scalable coding and object-based coding representation of audio-visual information that are suitable for the transmission of limited available bandwidth of networks. Therefore, MPEG-4 technology is suitable for related multimedia applications under high available bandwidth variation and limited QoS guarantee environments.

The rest of this paper is organized as follows. Section II reviews the related work as the necessary background. Section III outlines the architecture of proposed adaptive MPEG-4 rate control mechanism and the detail implementation issues. The simulation results are described in Section IV with conclusions in Section V.

II. Related Work

In this session we briefly discuss related research work for the MPEG-4 video applications over IP networks.

Wu et al. [9] proposed an end-to-end architecture for the MPEG-4 video applications over the Internet environment. The proposed architecture presented a rate control algorithm that was implemented by reducing target encoding bits directly. In addition, the packetization mechanism was based on the IP

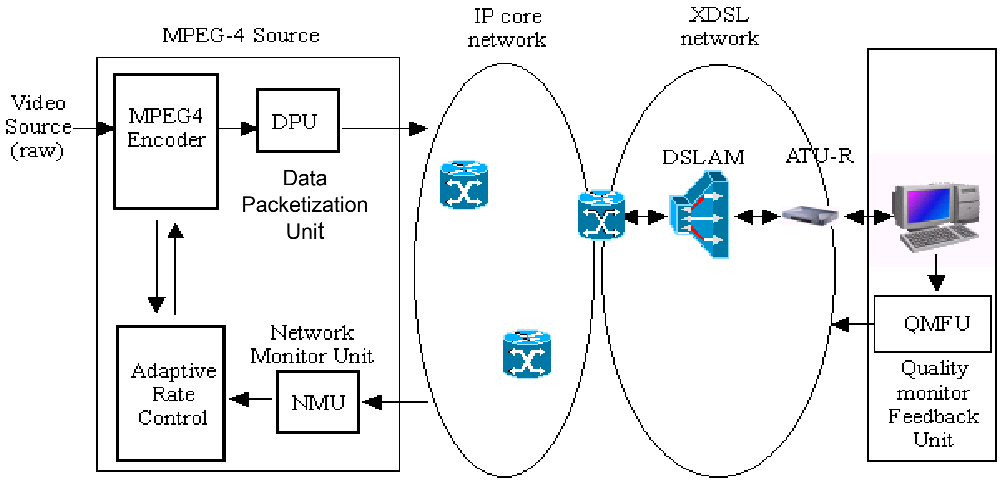

Fig. 1. Adaptive MPEG-4 rate control mechanism over Internet for ADSL subscribers.

layer. The influence of the SAR (Segmentation and Reassembly sublayer) operations in ATM Adaptation Layer (AAL) for IP over ATM, was not discussed. Ronda et al. [10] focused on the rate control and bit allocation in the MPEG-4 source. They formalized the modelization of the source and the optimization of a cost criterion based on signal quality parameters. However, the research was mainly focused on the MPEG-4 source. In [9], [11], [12], the feedback control packets were assumed loss-free. However, under the best effort IP environment, how to promise loss-free for feedback control packets was not discussed.

In this paper, we introduce an IP/MPLS/POS network architecture to resolve the issues of packetization and feedback control packet loss [13][14]. We also propose an adaptive MPEG-4 rate control mechanism over Internet for ADSL subscribers. The mechanism integrates the adaptive MPEG-4 source rate control algorithm and bi-directional negotiation scheme over proposed network to maintain the MPEG-4 video quality. Our mechanism would adjust the target quantization value according to the network environment instead of reducing target encoding bits.

III. Adaptive MPEG-4 Rate Control Mechanism

The adaptive MPEG-4 rate control mechanism is shown in Fig. 1. We organize this section as follows. In sub-section III-A we introduce the network architecture for IP over ADSL which can guarantee the reliability and delay of feedback control packets and improve the packetization performance. In sub-section III-B, we present the details of Quality Monitor and Feedback Unit (QMFU) that is located at the ADSL end hosts. In sub-section III-C we outline the operations of Network Monitor Unit (NMU). In sub-section III-D we describe the detail of Adaptive MPEG-4 Rate Control Unit (ARC). Finally, the functions of Data Packetization Unit (DPU) are presented in sub-section III-E. NMU, ARC, and DPU are all located at the MPEG-4 source encoder.

A. Architecture of IP Core Network

For IP over ADSL applications, ADSL end hosts connect to DSLAM (Digital Subscriber Line Access Multiplexer) by means of classical copper transmission lines. DSLAM then transmits these packets into general IP packet switching networks. ADSL standard organizations have proposed ATM technology as one of the transport networks [15].

However, Because of the SAR operations in AAL layer of ATM, the maximum throughput per port of current ATM switch is bounded, in general, to STM-4. The bound will limit the future development and provision for broadband multimedia services. Moreover, the additional processing overhead will decrease the efficiency of MPEG-4 packetization. Another limitation of IP over ATM networks is its scalability, especially in the Metro environment. One fully mesh topology of ATM PVC (Permanent Virtual Circuit) connections is needed in IP over ATM network if we request low propagation and processing delay within the network. However, especially in the Metro network, when the number of access switches or routers, N, increases, the total number of ATM PVC connections will increase by the order of O(N^2). The exact amount is equal to [N × (N-1)] / 2. Furthermore, the network

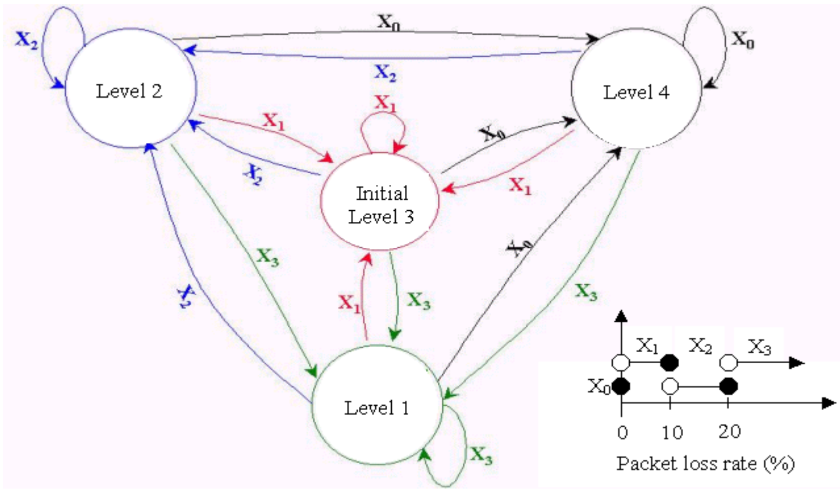

Fig. 2. The translation algorithm of NMU

management will become complex and the scalability will severely be limited because of the mass PVC connections.

Therefore, we introduces IP/MPLS/POS network for IP core network to overcome these issues mentioned above. After using RSVP-TE (RSVP-Traffic Engineering) mechanisms and suitable scheduling policies provided by MPLS [16], POS network can reserve bandwidth and promise the maximum delay jitter for special IP traffics, such as control packets. Thus, the feedback control packets could be protected well and transmitted rapidly within the IP core network in terms of RSVP-TE. Using IP/MPLS/ POS core network to replace IP/ATM core network also has the flexibility of packetization for encoded MPEG-4 bit-stream. After omitting the overhead of SAR and packetization limitation in ATM, the processing capability of network equipment for real time video applications can speed up dramatically.

B. Quality Monitor and Feedback Unit (QMFU)

QMFU is the mechanism that monitors the video packet loss status over networks and feedbacks the results to MPEG-4 source in order to adjust the encoding bits for meeting the network situation. QMFU is located at the ADSL end hosts where the packet loss status over network is monitored periodically. A monitoring period of 1-sec is used after considering the trade off between video quality variation and delay of feedback. Packet loss may occur if the network is congestible or the transmission delay is too long to decode properly. When the MPEG-4 video applications is enabled by the ADSL end host, the feedback control packets are transmitted periodically.

C. Network Monitor Unit (NMU)

After passing through the highest CoS (Class of Service) connection of IP/MPLS/POS core network, the feedback control packets are terminated by NMU. NMU translates the contents of feedback control packet to one of the quality levels according to the current quality level used. Then, the resulted quality level is sent to the ARC for more operations.

The translation algorithm is plotted in Fig. 2. The packet loss rate is monitored by QMFU and sent by means of feedback control packets. Four possible scopes of packet loss rate are presented by X_0, X_1, X_2, and X_3, respectively. When the resulted packet loss rate falls in the range of X_1, i.e. between 0 and 10%, the new quality level is 3 no matter what the current quality level is. If the resulted packet loss rate falls in the range of X_2, the new quality level is 2 no matter what the current quality level is. Similarly, when the resulted packet loss rate falls in the range of X_3, the new quality level is 1 no matter what the current quality level is. Finally, if the resulted packet loss rate falls in the range of X_0, the new quality level is 4 no matter what the current quality level is. For preventing any feedback control packet loss because of any possible network accident such as physical link fails, NMU periodically enables a delay counter to monitor the receiving situation of feedback control packets. When the counter times out and the event that expected feedback control packet is not received occurs twice, NMU will reduce the quality level from n to n-1 automatically.

D. Adaptive MPEG-4 Rate Control Unit (ARC)

When ARC receives the suggested quality

level from NMU, ARC enables the Target Bit Rate (TBR) computing algorithm. The algorithm is described in Table I.

According to the quantization value given by ARC, MPEG-4 encoder executes remaining encoding processes. After complementing the post-encoding operations, the encoded bit-stream is exported to DPU.

Table I. TBR computing algorithms of ARC

IF *Level = 4*
    THEN
    // Increase available encoding bits //
    available encoding bits ＋= *5Kbps*
    $T\_Rb = T\_Rb + 5Kbps \times (R\_P \div frame\ rate)$
    //*T_Rb*: Total available remained bits //
    // *R_P*: remained P frame number //
    Calculate *Target bits*
    Calculate *Target Q*

ELSE IF *Level = 3*
// Hold current available encoding bits //
    THEN
    Calculate *Target bits*
    Calculate *Target Q*
    // *Q* : quantification value //

ELSE IF *Level = 2*
// Reduce encoding bits by increasing Q //
    THEN
    Calculate *Target bits*
    Calculate *Target Q*
    $Q = Q + 1$

ELSE IF *Level = 1*
// Skip the frame directly //
    THEN
    Skip frame

E. Data Packetization Unit (DPU)

Before transport over networks, MPEG-4 encoded bit-stream has to be packaged into packets by DPU. DPU provides three packetization methods, fixed packet size based scheme, GOB (Group of Blocks) based scheme, and VOP based scheme [9][17][18]. Finally, these packets are transmitted into networks.

IV. Simulation Results

In this section, we quantify the performance of the proposed integrated adaptive MPEG-4 quality control mechanism through system level simulations. We first describe the simulation model, followed by the simulation results and discussions.

A. Simulation Model

At the video source side, we utilize three standard MPEG-4 raw video sequences, “ Akiyo”, “News”, and “Foreman”, in QCIF format for the MPEG-4 video encoder. The video complexity of these sequences is “Low”, “Medium”, and “ High”, respectively.

The default target rate is 56Kbps, the frame rate is 15 frames/sec, and the default quality level is 3. We use frame based encoding scheme for our simulations. The encoded bit-stream has to be packaged before being sent to the network. We use two different packetization schemes to compare the efficiency. One is the fixed packet size scheme, and the other is the VOP based scheme.

In this paper, IP core networks use IP/MPLS/POS technology. The highest priority queue and scheduling policy in each component of the networks are dedicated to the feedback control packets. RSVP-TE mechanism is enabled. The propagation delay of feedback control packets is assumed as 0.5sec in the worst case. In QMFU, the monitoring period is one second. With the proposed network architecture, the feedback control packets are assumed loss-free under network congestion situation. As shown in Fig.3, the traffic condition of networks is described by four different cases: heavy, variable, light, and loss free.

B. Results and Discussion

In this sub-section, we present the simulation results for demonstrating some of the important features of our work. There are cataloged to three simulation cases as follows.

First of all, in case one, we focus on the performance comparison of different packetization schemes under IP/MPLS /POS core network and ADSL network. Encapsulated MPEG-4 video packets are transmitted into networks without any rate control mechanism. Two types of packet length, 300 bits and 1200 bits, are used in the fixed packet size scheme. With these three kinds of MPEG-4 raw video sequences and four types of network condition, there are twelve simulation combinations. These results are shown in Table II. Our results reveal that, when the traffic condition is fixed to loss free and various MPEG-4 raw video sequences are used, VOP based scheme provides better video quality than fixed packet size scheme. Moreover, when we change the simulation combination from one to the other, VOP based scheme still works well under all of

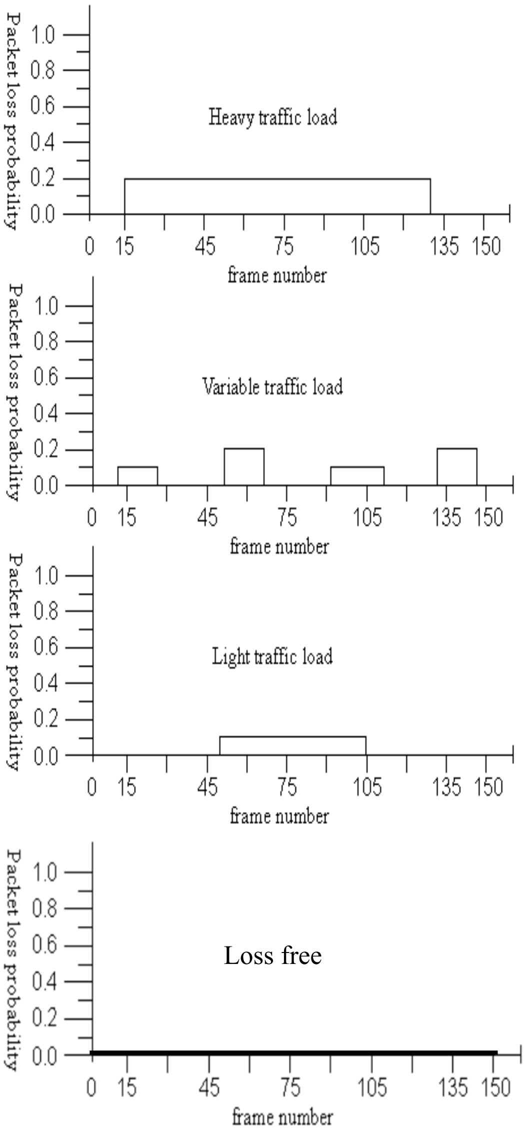

Fig.3 Four types of traffic condition in Internet

the twelve simulation combinations. In summary, no matter what kind of the three video sequences and what type of the four traffic conditions, VOP based scheme always provides better quality performance

One of the main merits of proposed IP/MPLS/POS network architecture for packetization efficiency is that the variable packet size is allowed. After omitting the overhead of SAR and packetization limitation in ATM, VOP based scheme is suitable for MPEG-4 video applications.

In simulation case two, the performance comparison between the two rate control mechanisms, one is the method in terms of adjusting the target quantization value and the other is the method with adjusting the target bits directly, is executed for MPEG-4 applications over networks. Based on the discussion mentioned in simulation case one, we use the VOP based scheme for packetization operations. The network is described by four different traffic conditions respectively, as shown in Fig. 3.

Table II Comparison of packetization schemes without any rate control mechanism

		1 VOP	1200 bits	300 bits
		Average PSNR		
Akiyo	Loss-Free	***40.425***	40.29	39.976
	Heavy	***32.522***	32.100	31.947
	Variable	***36.389***	35.890	35.744
	Light	***38.010***	37.981	37.145
		1 VOP	1200 bits	300 bits
		Average PSNR		
News	Loss-Free	***33.853***	33.513	33.364
	Heavy	***25.892***	25.499	25.463
	Variable	***30.005***	29.127	29.039
	Light	***31.118***	30.364	30.274
		1 VOP	1200 bits	300 bits
		Average PSNR		
Fore man	Loss-Free	***29.876***	29.638	29.300
	Heavy	***20.371***	20.238	20.090
	Variable	***27.372***	26.516	26.113
	Light	***27.848***	27.211	26.858

The results are shown in Table III. With all possible condition combinations of three kinds of rate control method and four types of network condition, Akiyo always has the best PSNR and Foreman get the worse PSNR quality. This is because the video complexity of Akiyo is less than News and Foreman. As expected, video applications without any rate control always get the worst PSNR under all possible combinations of four traffic conditions and three MPEG-4 video sequences. Moreover, when two rate control mechanisms are simulated, the MPEG-4 rate control method with adjusting target quantization value instead of target bits get the best received video quality. Especially, the higher the traffic load, the more PSNR improvement the MPEG-4 rate control with adjusting target quantization value. More importance, the higher the video complexity of MPEG-4 sequence, the more PSNR improvement the MPEG-4 rate control with adjusting target quantization value.

Table III Comparison of various rate control schemes with VOP based packetization scheme

		Without Rate control	With rate control R'=0.8R	With rate control Q↑
		Average PSNR		
Akiyo	Loss-Free	40.425	40.877	40.877
	Heavy	32.522	33.490	***33.564***
	Variable	36.389	36.469	***36.521***
	Light	38.010	38.058	***38.258***
		Without Rate control	With rate control R'=0.8R	With rate control Q↑
		Average PSNR		
News	Loss-Free	33.853	34.953	34.953
	Heavy	25.892	27.036	***27.383***
	Variable	30.005	29.851	***30.744***
	Light	31.118	31.468	***31.800***
		Without Rate control	With rate control R'=0.8R	With rate control Q↑
		Average PSNR		
Fore man	Loss-Free	29.876	31.230	31.230
	Heavy	20.371	21.271	***21.780***
	Variable	27.372	27.038	***27.553***
	Light	27.848	27.959	***28.782***

Finally, in simulation case three, for showing the influence of fixed packet size based scheme to rate control mechanisms, the simulations are executed using various rate control methods and fixed packet size scheme. These results are presented in table IV and V. Under fixed packet size and the condition combinations used in simulation case two, rate control method with adjusting target quantization value still get the better received video quality than with adjusting target encoding bits directly. Moreover, when the rate control method is used and fixed, VOP based scheme always provides better quality.

V. Conclusions

In this paper, we have proposed the IP/MPLS/POS core network architecture to meet the requirements of ADSL subscribers, characteristics of MPEG-4 bit-stream packetization and delay tolerance of feedback control packets. We show through simulations that, after omitting the overhead of SAR and packetization limitation in ATM, VOP based scheme could always provide better quality performance than fixed packet size scheme under proposed network architecture and proposed rate control mechanism in this paper. We also have proposed the integrated adaptive MPEG-4 rate control mechanism over IP/MPLS/POS networks and ADSL networks. Our results reveal that, the higher the traffic load, the more PSNR improvement the MPEG-4 rate control with adjusting target quantization value. Moreover, we note that, with the proposed MPEG-4 rate control by adjusting target quantization value, the higher the video complexity of MPEG-4 sequence, the more PSNR improvement.

Table IV Comparison of various rate control schemes with fixed packet size of 1200 bits

		Without Rate control	With rate control R'=0.8R	With rate control Q↑
		Average PSNR		
Akiyo	Loss-Free	40.290	40.866	40.866
	Heavy	32.100	33.154	***33.250***
	Variable	35.890	36.039	***36.473***
	Light	37.981	38.034	***38.070***
		Without Rate control	With rate control R'=0.8R	With rate control Q↑
		Average PSNR		
News	Loss-Free	33.513	34.712	34.712
	Heavy	25.499	26.721	***26.935***
	Variable	29.127	29.752	***29.770***
	Light	30.364	30.402	***30.433***
		Without Rate control	With rate control R'=0.8R	With rate control Q↑
		Average PSNR		
Fore man	Loss-Free	29.638	31.129	31.129
	Heavy	20.238	21.250	***21.655***
	Variable	26.516	26.677	***26.750***
	Light	27.211	27.628	***28.120***

Table V Comparison of various rate control schemes with fixed packet size of 300 bits

		Without Rate control	With rate control R'=0.8R	With rate control Q↑
		Average PSNR		
Akiyo	Loss-Free	39.976	40.551	40.551
	Heavy	31.947	32.794	***33.144***
	Variable	35.744	35.765	***35.771***
	Light	37.145	37.683	***37.811***

		Without Rate control	With rate control R'=0.8R	With rate control Q↑
		Average PSNR		
News	Loss-Free	33.364	34.563	34.563
	Heavy	25.463	25.909	***26.710***
	Variable	29.039	29.067	***29.631***
	Light	30.274	30.369	***30.420***

		Without Rate control	With rate control R'=0.8R	With rate control Q↑
		Average PSNR		
Fore man	Loss-Free	29.300	30.795	30.795
	Heavy	20.090	21.206	***21.353***
	Variable	26.113	26.134	***26.422***
	Light	26.858	27.345	***27.430***

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