

Dynamic Rate Control for Continuous Media Transmission

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On continuous media transmission using compression techniques, a variable bit rate transmission method is required to maintain the video frame rate at a constant. At the same time, user's Quality of Service must be guaranteed even if packet loss and delay occur by CPU load deviation on the client or server, and network traffic increase. In this paper, two variable bit rate transmission methods are described when MPEG compressed video are used. We also introduce both a packet rate control method to reduce the packet loss, and frame rate control method to maintain the frame rate constant under dynamic load conditions. We implemented a prototyped Packet Audio/Video System(PAVS) to evaluate performance of these rate control functions. Through the performance evaluation our PAVS performance has demonstrated the usefulness of our suggested control methods.

連続メディア転送における動的レート制御について

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圧縮を考慮したメディア転送において、一定レートで転送表示を行なうためには、可変ビットレート転送を行なう必要がある。一方、計算機やネットワークの負荷変動が発生した場合において、利用者の要求するサービス品質(QoS)を保証する必要がある。本稿では動画像圧縮法としてMPEGを使用した場合の2つの可変ビットレート転送方式を導入する。さらに、動的な負荷変動に対して、パケットロス率を許容以下に抑制するパケット間隔制御と、連続メディアの持つ時間的制約と安定したフレームレートを維持するフレームレート制御を提案する。そして、これらの制御を導入したパケットオーディオ・ビデオシステム(PAVS)のプロトタイプを構築し、性能評価を行った結果、その有効性が確認できたので報告する。

1 Introduction

In order to provide continuous media services, such as Video-on-Demand over high speed networks, a suitable QoS(Quality of Service) as requested by the user has to be guaranteed by taking account of the characteristics of offered media data, the processing capabilities and load deviations of both client stations and video servers, and the available network bandwidth and its traffic load. Therefore, the system has to include mechanisms which guarantee End-to-End QoS from the application through network layers.

On the other hand, video compression techniques such as MPEG[1], are efficient to store and transmit over network to reduce amount of video data. However, since the amount of data of each video frame varies in time depending on the motion of the video contents when the video compression methods used, a variable bit rate transmission is required to maintain the video frame rate constant.

At the same time, light and simple protocol such as UDP, is expected to provide higher throughput rather than reliable protocol such as TCP.

In this paper, we described two variable bit rate transmission method when MPEG video data are used: 1) GoP(Group of Picture) transmission method in which that transmits GoP of MPEG video as an undivided whole. 2) Frame transmis-

sion method that discriminates each MPEG video frames from a GoP respectively.

Also, due to packet loss and delay by increase of CPU load at client station or video server, and network traffic load changes, the video frame rate is seriously influenced and dropped. So we introduced packet rate control method to reduce the packet loss by adjusting the packet interval time and frame rate control method to maintain the frame rate at a constant and keep the time constraint of continuous media data by controlling transmitted frame rate on the server based on the feed back signal sent by the client station.

We implemented a prototyped Packet Audio/Video System(PAVS) on FDDI and Ethernet to evaluate performance of these rate control functions. And we compared temporal variation with and without these suggested control methods through the network traffic loads are dynamically changed. As a result, when these control functions were introduced, we could reduce packet loss rate under the admissible rate, maintained video frame at a constant and keep the time constraint of the continuous media data.

In the following section, we introduce a PAVS architecture in section 2, In section 3, we introduce two different variable bit rate transmission methods for MPEG video. In section 4, a dynamic

rate control which contains packet rate control and frame rate control is described. In section 5, the prototype system for PAVS and its performance evaluation are discussed.

2 System Architecture

In order to realize dynamic rate control mechanism on continuous media transmission service, we introduce client/server model architecture based on our PAVS three layers, synchronization, data transform and media flow control layers between the application and transport layers in the OSI reference model (Figure 1). This system architecture contains the functions required to smoothly provide continuous media data to users and guarantee QoS from the application through the network layer. We have integrated these three layers into one layer as the media coordinate system layer[4].

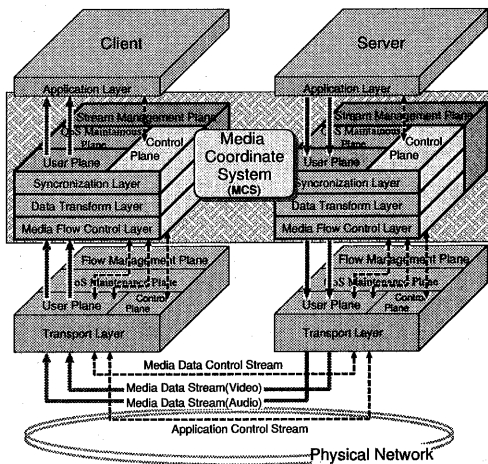


Figure 1: System Architecture

The media coordinate system is furthermore vertically divided into four planes: the user plane, the control plane, the QoS maintenance plane and the stream management plane, considering high speed network, ATM and adopted QoS-A model of Lancaster university[3]. In the user plane, synchronization function between different media, such as audio and video, data transform function between different media attributes, media flow control for both constant and variable bit rate transmission for continuous media are performed[5]. In the control plane, connection establishment/release of the media stream and QoS renegotiation and maintained. In the QoS maintenance plane, each entity is responsible for the fine-grained monitoring and maintenance of their associated protocol entities. In the stream management plane, the most suitable QoS parameter values on each protocol layer are determined according to the user's QoS requirements, characteristics of the source media data, output device attributes, and available computing and network resources. We define these factors as QoS decision factors, such as frame rate, synchro-

nization interval, packet interval and packet loss rate.

Figure 2 shows the functional modules necessary to realize the system architecture and our suggested dynamic rate control functions.

Packet rate management module: the numbers of packet to be transmitted on server or to be received on client are managed.

Frame rate management module: actual frame rate to be read from hard disk on server or to be displayed on client are managed.

Stream management control module: CPU loads on client and server, and traffic condition of network are periodically monitored and feed back message between client and server, is managed.

The detail of the dynamic rate control functions is explained in the section 4.

3 Variable Bit Rate Control

In order to transmit and display both uncompressed and compressed video frames, we introduce two different variable bit rate methods including Group of Picture (GoP) transmission method and frame transmission method[6]. In both methods, the packet length is assumed to be fixed although the number of the packets in a time unit and the time interval between the packets are controlled according to GoP or frame size.

In the case of GoP transmission method for the MPEG video depicted in Figure 3 a), where one GoP is consisted of 8 frames (1 I-frame, 2 P-frame and 6 B-frames), the multiple packets equivalent to one GoP are generated and transmitted. When we define the time interval for a GoP as $T_{GoP}[sec]$, the number of the packet for a GoP as N_p is the number of the packet and data size of a GoP as $L_{GoP}[bytes]$, and transmitted packet size as L_p , then $N_p = \lceil L_{GoP}/L_p \rceil$ packets are generated at video server and transmitted within $T_{GoP}[sec]$ to the client. Thus, constant frame rate with variable data sizes can be attained by generating packets according to the data size of each GoP within T_{GoP} . We define this as GoP transmission method. In this method, since each GoP consisted of multiple frames is regarded as a basic unit for transmission, not only packet overhead can be reduced, but also the processing load of intra-frame and inter-frame synchronizations can be simplified. However, when one packet is lost, then the multiple frame are influenced.

On the other hand, in the case of the frame transmission method as depicted in Figure 3 b), each frame including I-frame, P-frame and B-frames on each GoP is identified and the packets equivalent to each frame are generated on the server and transmitted to the client.

Those two methods individually have the following characteristics:

The GoP transmission method: since MPEG data can be packetized without considering the type of frames in a GoP, the number of

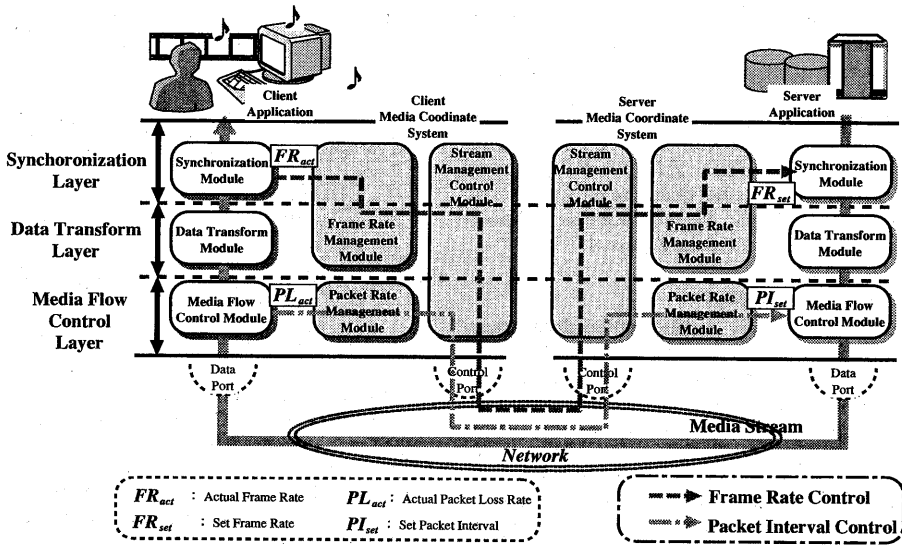


Figure 2: Functional Module Configuration and Flow of Rate Control

the packet generated and their packet processing can be reduced. On other hand, when the packet is lost by increasing the CPU and network traffic loads, a number of frames belong to one GoP is lost.

The frame transmission method: since each frame in each GoP is identified and transmitted individually, the number of the packets generated increases, and the complicated packet process on each control module must be implemented. On the other hand, the influence of the lost packet can be restricted within one frame.

By controlling the generation rate of the fixed length packets depending on the amount of a GoP or a frame data, variable bit rate transmission can be attained.

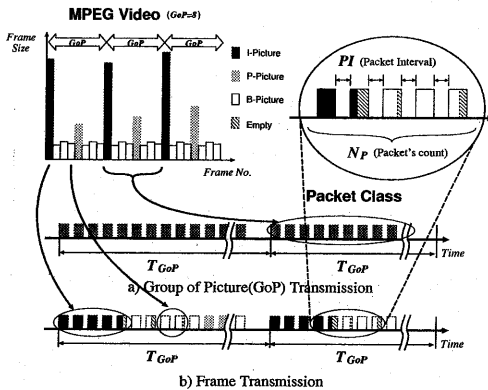


Figure 3: Variable Bit Rate Transmission

4 Dynamic Rate Control

Too much CPU load of the client stations or video servers, and network traffic changes cause

packet loss, delay and jitter by packet buffer overflow of client/server or router, and eventually reduce the transmitted End-to-End Quality of Service. In this research, we propose two dynamic rate control methods adapted to the environment corresponding to the conditions of the loads[7].

In packet rate control, inter-packet time interval is controlled on the server to reduce the packet loss rate under the admissible loss rate. In frame rate control, transmitted frame rate is controlled on the server to maintain the frame rate at a constant and keep the time constraint of continuous media data. By combining the two rate control methods, audio/video quality can be guaranteed and the video frame rate displayed on the client can be maintained at a constant under various traffic load conditions.

4.1 Packet Rate Control

In order to reduce packet loss rate to within the admissible loss rate when the CPU loads in both client and server or network traffic load has been dynamically changed, packet interval, $PI[msec]$, is adjusted.

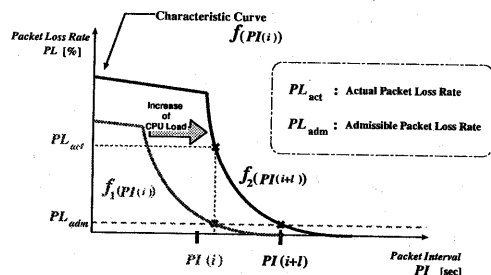


Figure 4: Characteristic Curve of Packet Interval and Packet Loss Rate

In this system, when the packet loss rate, $PL_{act}[\%]$, for a measurement interval, $T[sec]$, is exceeded over the admissible packet loss rate, $PL_{adm}[\%]$, then the packet interval, PI , is immediately increased based on the characteristic curve between the packet interval and packet loss rate and dynamically controlled to keep the condition $PL_{act} < PL_{adm}$. Here, the characteristic curve between the packet interval and packet loss rate can be measured as a characteristics curve as depicted in Figure 4.

4.2 Frame Rate Control

In order to normally display video data transmitted from the server on the client station, the frame rate by which the video source was initially stored at the video server must be maintained during video session and displayed on the client. However, in the case where the frame rate of the source video cannot be attained at the client, then sub-sampling of the frame has to be introduced to reduce the source frame rate before transmitting the video data from the server.

The client frame rate controller in the QoS maintenance plane periodically monitors the real frame rate FR_{act} to be displayed and compares it with the set frame rate FR_{set} . If the FR_{act} is smaller than FR_{set} for the interval $T[sec]$, then the counter value for rate difference status, $Miss_{cnt}$, is incremented. When $Miss_{cnt}$ exceeded a threshold, then a rate control message to reduce the current frame rate by ΔFR is sent from the client station to the video server's rate controller. The rate controller at the video server then dynamically updates the current frame rate based on the control message by sub-sampling the source frame rate FR_{src} . The rate controller at the client station also updates the set frame rate FR_{set} to the new value which has been sent to it from the video server's frame rate controller. By periodically adjusting the set frame rate, a constant rate is maintained.

5 Prototype and System Evaluation

In order to evaluate the usefulness of our suggested variable bit rate transmission method and dynamic rate control methods, a Packet Audio/Video System(PAVS) was prototyped based on the combination of FDDI with $100[Mbps]$ and Ethernet with $10[Mbps]$ using UDP/IP protocol as a transport protocol (Figure 5).

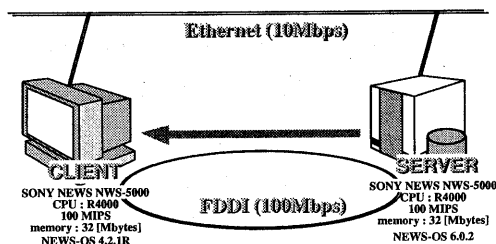


Figure 5: Prototype

The characteristics of the source video and other system parameters in PAVS are shown in Table 1.

Physical Networks	FDDI	Ethernet
Complession Format	MPEG-1	
Avarage Bit Rate[Mbps]	5.0	1.5
Pixel Resolution[pixels]	640 × 480	
Color Depth[bits]	24	
Frame Rate[fps]	30	
Number of Frames in GoP	N = 15	
I,P Frames Cycle	M = 5	
Packet Length[byte]	4K	1K

Table 1: System Parameter of Evaluation

As extra loads on the user client, the number of the process of MPEG-1 software decoder(mpeg_play[2]) was increased from 0 to 2 on the user client. As network traffic load, continuous remort file transfer equivalent to about $7.5[Mbps]$ between the other workstations was intentionally executed.

Since the evaluation of two different variable bit rate transmission[6] and the dynamic rate control against at client extra load[7] has been already reported, we discuss the evaluation of dynamic rate control against at network traffic load.

5.1 Packet Interval and Packet Loss Rate

In order to control the packet interval when the actual packet loss rate PL_{act} exceeded the admissible packet loss rate PL_{adm} , the relation between inter packet interval and the packet loss rate was investigated using the prototype system. The packet loss rate was observed by actually sending the packets with the fixed length from the video server to one of the client station and changing the packet interval for different network traffic load conditions as shown in Figure 6.

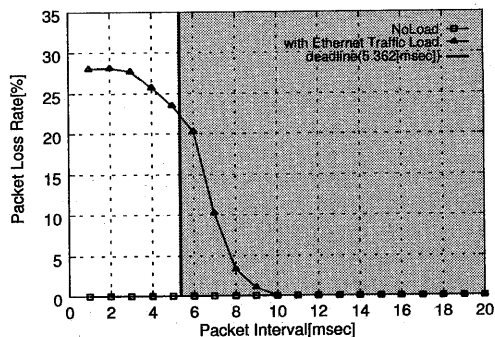


Figure 6: Relationship between Packet Interval and Packet Loss Rate

As understood from this Figure 6, the packet loss rate gradually dereases from 28[%] to 0[%] as the packet interval increases. However, when packet interval is larger than $5.4[msec]$, the time constraint of the source video fram rate with $30[fps]$ cannot be maintained. We define this time interval as "deadline" of the packet interval for this video

source. In the case of the packet interval larger than the deadline, the original frame rate cannot be maintained. Because the required average packet throughput for an video frame cannot be attained, the original frame rate cannot be maintained. Therefore, the source video must be sub-sampled and the packet interval must be controlled to maintain the admissible packet loss rate while maintaining the frame rate higher as much as possible. Thus, this characteristic curve with the packet loss rate, PL is expressed by the following equation:

$$PL = \begin{cases} -1.58 \times PI + 31.35 & (PI < \text{deadline}) \\ -6.62 \times PI + 58.4 & (PI \geq \text{deadline}) \end{cases}$$

5.2 Evaluation of Dynamic Rate Control

Figure 7 a) and b) show the results of the packet rate control when the extra network traffic load (7.5[Mbps]) was given over the network during video data transmission. In the case where the packet rate control function was not introduced, the packet loss rate randomly increased from 0[%] to around 50[%] when the extra network traffic load was given between the 30[sec] to 120[sec] elapsed time while the packet interval was constant at 1[msec]. On the other hand, in the case where the packet control function was introduced, the packet interval immediately was increased to 12[msec] to reduce the packet loss rate, after that the packet loss rate could be maintained under the admissible loss rate. However, the packet loss rate could not completely maintained at constant because only the current value of the packet loss rate was feeded back from the client to the video server. It is required to improve the feed back control function to always keep the packet interval within the admissible loss rate by combining the change values of the CPU load on both client and server or network traffic loads with the packet interval.

Figure 8 a) and b) show the result of the frame rate control when the extra network traffic load was given during video data transmission. In the case where the frame rate control function was not introduced, the actual frame rate decreased and randomly and widely varied during the extra traffic load period although the set frame rate was constant at 30[fps]. On the other hand, in the case where the frame rate control function was introduced, the set frame rate on the video server was immediately updated according to the feed back signal from the client. After that the actual frame rate decreased to 15[fps] and maintained this frame rate at a constant. After the CPU load was released, the frame rate was gradually increased and approached to the original frame rate 30[fps].

Figure 9 and 10 show the evaluation of combining packet and frame rate control. In the case where only a packet rate control was introduced. The packet loss rate could be reduced, but frame rate was fluenced when the extra load was given (Figure 9 a)), and since the packet interval exceed the deadline, the time constraint of the video frame

could not maintained (Figure 10 a)). However, in the case of both the packet and frame rate controls are introduced at the same time, the frame rate could be maintained at a constant (Figure 9 b)) while keeping time constraint of the original video data (Figure 10 b)).

Thus, we could verify the usefulness of the suggested these rate control function.

6 Conclusions

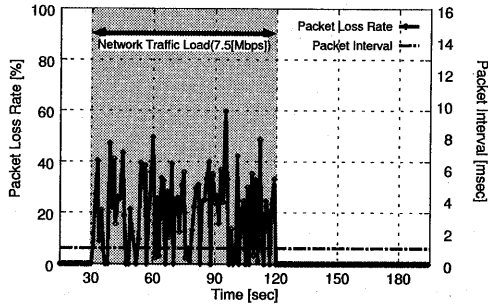
In this paper, we described two variable bit rate transmission methods, the GoP transmission and the frame transmission when compressed video data were transmitted over networks. The GoP transmission method can expect efficiency. On the other hand, the frame transmission can expect flexibility. These transmission method could be used selectively by QoS requested to media services.

At the same time, we suggested packet rate control and frame rate control methods as dynamic rate control against extend load on client or server, and network. We implemented these rate control to evaluate performance of them. As a result, our PAVS performance has demonstrated the usefulness of our suggested control methods even through the network traffic loads are dynamically changed by comparing the performance between with and without these two rate control methods.

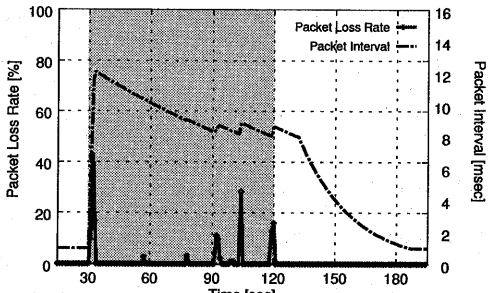
We will evaluate of these rate control considering audio/video synchronization and implemented and evaluate packet recover control.

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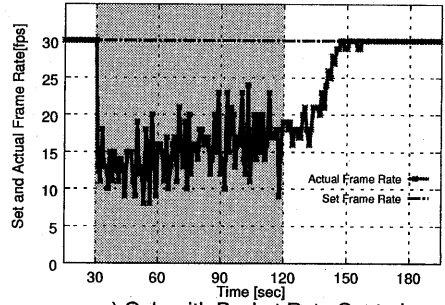


a) without Packet Rate Control

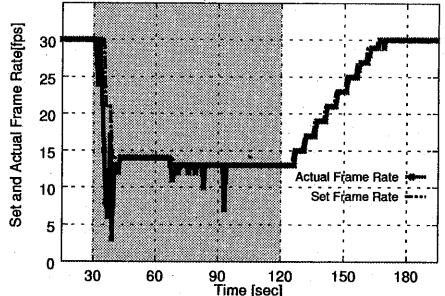


b) with Packet Rate Control

Figure 7: Evaluation of Packet Rate Control with Network Traffic Load(7.5Mbps)

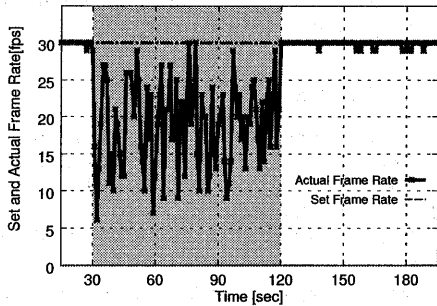


a) Only with Packet Rate Control

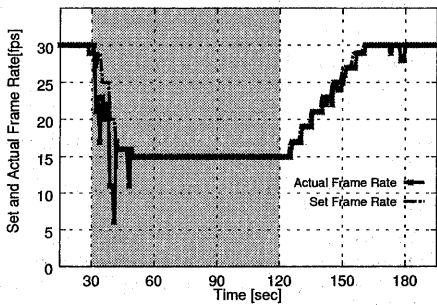


b) with Both Packet/Frame Rate Control

Figure 9: Evaluation of Combining Packet Rate and Frame Rate control with Network Traffic Load(7.5Mbps) Part 1

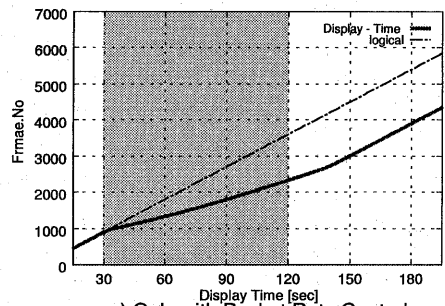


a) without Frame Rate Control

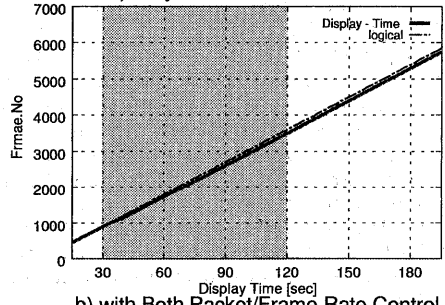


b) with Frame Rate Control

Figure 8: Evaluation of Frame Rate Control with Network Traffic Load(7.5Mbps)



a) Only with Packet Rate Control



b) with Both Packet/Frame Rate Control

Figure 10: Evaluation of Combining Packet Rate and Frame Rate Control with Network Traffic Load(7.5Mbps) Part 2