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A Robust Watermarking Algorithm for Adaptive Synchronization for OTT Services

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ABSTRACT :*Time delay during the multimedia data transmission through internet causes non-synchronized phenomena between video and sound when they play back. To prevent this quality lowering, media synchronizations are necessary to secure seamless and harmonized playback. However, cell loss occurs in the process of multimedia synchronization, and inserted watermarks are also lost in the process of cell loss. In this paper, a synchronization algorithm is proposed to be able to deliver watermark to the final display terminal, minimizing watermark deletion during the synchronization process for video streaming.*

KEYWORDS *digital watermarking, MPEG synchronization, VOD service, adaptive synchronization.*

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I. INTRODUCTION

Multimedia services integrated with information communications and broadcasting are provided thanks to the advancements of internet technology and various mobile devices. Unlike traditional broadcasting services that transmit the contents by fixed exclusive broadcasting network, OTT(Over The Top) transmits them by general internet that multiple users are easily accessible. Hence, desired programs can be viewed in a variety of devices such as smart phones, tablet PC, and so on, without limitation of service hours. However, there are many cases of video playback delay during the streaming course in the OTT services due to heavy network traffic. Time delay during the multimedia data transmission through internet causes non-synchronized phenomena between video and sound when they play back. To prevent this quality lowering, intra and inter-media synchronizations are necessary to secure seamless and harmonized playback. Generally, the process to maintain the time of one or multiple media streaming is called as multimedia synchronization. During the synchronization process, loss of cell is occurred[1]. Since VBR(Variable Bit Rate) MPEG video generates much larger traffic at I-picture which is the start of GOP(group of pictures) than other pictures, it shows the cyclic pattern of traffic generation depending on the structure of GOP. Hence, in case of multiple sources of VBR MPEG videos, starting time arrays of I-pictures affect the cell loss in multiplexer, significantly.

Park et al.[2] proposed a method to array I-picture starting time by each video source to minimize the rate of cell loss in multiplexer when VBR MPEG video sources were multiplied to be transmitted into one line. In the proposed method, the probability that arrival rate of cells from the multiplied sources exceeded the capacity of the transmitting line was used to identity accurate I-picture starting time, effectively.

Kong et al.[3, 5] suggested the method to insert watermark into LSB(Least Significant Bit) of DCT quantization parameter after performing 8x8 block DCT on every frame of the original video. This method was durable for the change of frame rate and general compression technique, but not for size modification or change of video quality.

Wang et al.[4] inserted watermark into AC parameter near DC parameter of I-picture referring to DCT parameters before and after frames, after performing DCT by group during the coding process of MPEG-2. This method is durable for MPEG-2 compression or general geometric attacks, but not for missing reference frame or frame change by modification. There are many cases to delete the delayed video for synchronization if the video transmission is delayed due to network traffic. In this case, watermark in the video is also deleted[5].

In this paper, a synchronization algorithm is proposed to be able to deliver watermark to the final display terminal, minimizing watermark deletion during the synchronization process for video streaming.

II. RELATED WORKS

Multimedia synchronization is essential to offer the multimedia services such as OTT services, and so on. Synchronization between transmitter and receiver is maintained if loss and delayed jitter are not occurred in the transmitter system that generates and transmits media stream, in the receiving system that receives and plays back the media stream, and in the transmitting network that transmits the media stream. However, in the transmitter and the receiver, packet loss or delay can be occurred in the transmitting network due to the difference of system performances. Because of this, media stream arrived in the receiver will lost the original time relation, especially causing serious problems in the time-dependent continuous media such as audio or video. Therefore, the synchronization technique is required to maintain the time relation among multimedia streams before the arrived messages play back in the receiver[6]. Existing studies on multimedia synchronization can be classified into static synchronization, adaptive synchronization by sender and adaptive synchronization by receiver.

Fig.1 shows the processing of MPEG encoder/decoder[7-10]. In Fig.1, SIF means Source Input Format, ME means Motion Estimation, MC means Motion Compensation, VLC means Variable Length Coding, VLD means Variable Length Decoding.

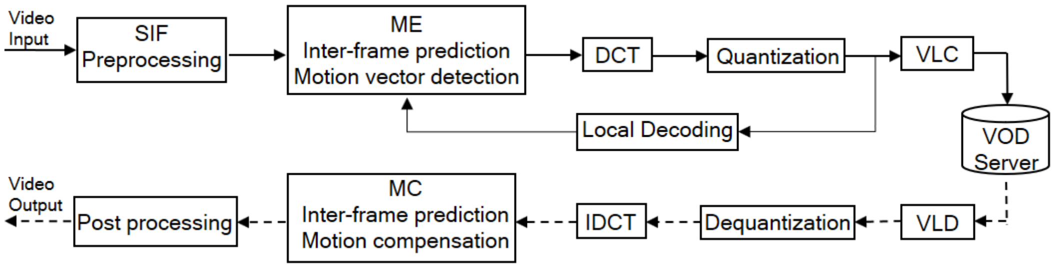

Fig.1. Processing of MPEG encoder/decoder

III. ADAPTIVE SYNCHRONIZATION ALGORITHM CONSIDERING WATERMARK SURVIVAL RATE

3.1 Synchronization model of OTT system

Fig. 2 shows the synchronization process between OTT server and client[11]. The server has the time information of SCR(System Clock Reference), and the client also performs the synchronization process with 90kHz STC(System Time Clock). In the pack transmitter of the server, pack-based transmission is performed due to the nature of MPEG system streaming, scheduling transmission time by SCR value of the pack header. That means when SCR value of the current pack corresponds with that of the server with 90kHz SCR clock, the pack is transmitted to be able to maintain synchronization with 90kHz STC which is referred during the decoding process of the client. The client consists of pack receiver and video/audio decoder. In the pack receiver of the client, STC clock is increased from the initial delayed time after receiving the first pack, maintaining STC time with 90kHz as a reference time. Clock drift in SCR of the server and STC of the client is neglected while only packet delay and loss in the network are considered[6].

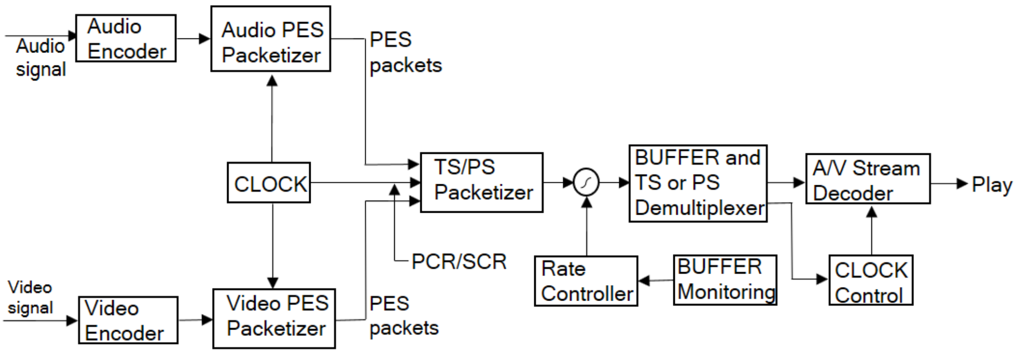

Fig. 2.Synchronization process between OTT server and client

In Fig. 2, TS and PS mean transport stream and program stream, respectively. PS or TS stream is developed from audio/video stream through packetizer. Encoded streams by audio/video encoder are packetized by PES(Packetized Elementary Stream) packetizer.When the pack is transmitted from the server, RTP(Realtime Transport Protocol) header by each unit is attached as the additional information and transmitted to the client.

Since RTP header has the number of order, loss factor in the network can be found from this. If RTP packet is received in the client, loss factor is found using RTP header information and controlling operation by the loss factor is performed upon feedback of the information. If loss factor is increased, number of transmitting pictures should be lowered to solve the network traffic problem. If loss factor is declined again, number of transmitting pictures should be increased[6].

3.2 A robust watermarking algorithm for adaptive synchronization

In MPEG system stream, audio/video are multiplied, however, they can be considered as one media in terms of pack level. Hence, intra-media synchronization can be achieved using SCR time information in the pack level and inter-media one using PTS information in the packet level. During intra-media synchronization process, delay and loss experienced in the network are processed[12]. Also, network resources are efficiently utilized considering both picture loss due to lack of playback processing capacity in the decoder and loss occurred in the network. First, transmission control is performed on the playback processing capacity in the decoder and loss occurred in the network, then, playback time control is performed regulating the buffer effectively, considering the delay experienced in the network and playback delay of the client. Picture loss can be occurred in the network due to network traffic and in the decoder due to the difference between transmission rate of the server and playback processing rate of the client[6].

This algorithm considers both cases. MPEG pictures contain three types including I-picture, P-picture, and B-picture, and first two are more important than the last during decoding process. If loss factor is increased due to network traffic, B-picture is selectively discarded. This is the way to lower the transmission rate of the server to lower the loss of I- and P-pictures. On the other hand, if loss factor of I- and P-pictures is declined, their receiving rates will be increased, providing better video quality than the case without the control. MPEG video in this paper has the format of I, P and B-pictures, and I and P-pictures should not be discarded to decode B-pictures. The video information in the MPEG system has the unit of GOP(Group Of Picture), hence, decoding can be started based on this. Therefore, it is desirable to discard B-pictures by GOP unit.

In this paper, watermark is inserted to I-pictures as seen in Fig. 3, considering this situation. Since I-pictures are not discarded when the video playback is performed in the MPEG system, the watermark is highly likely to be remained. In Fig. 3, M represents the cycle to show I-pictures or P-pictures. If playback of pictures is not performed to be discarded due to the difference between the transmission rate in the server and playback processing rate in the client, loss of the watermark is very likely. Video decoder in the client checks the number of available pictures for playback periodically to deliver its information to the server, and the server controls the transmission not exceeding the number of available pictures for playback, using the network resources, efficiently.

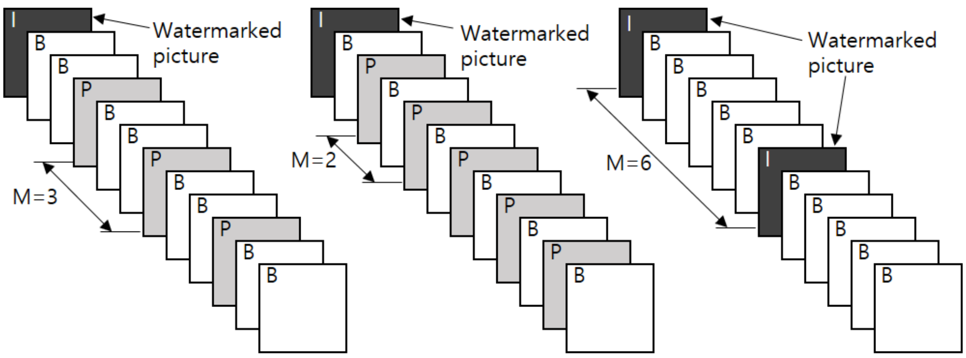

Fig. 3. Insert a watermark into an I-picture

In this paper, the algorithm for minimizing the loss of watermarked I-pictures in the adaptive synchronization method is as follows.

Step 1: Calculate the delay time of the pack by the network traffic.
Pack's delayed time = SCR of the received pack - Client STC upon arrival
Step 2: Packs that arrive earlier than the scheduled playback time are immediately played if there is no currently playing pack, and if the client is playing, it is inserted into the buffer and waited.
Step 3: For packs waiting in the buffer, each scheduled play time is calculated.

Step 4: If a new pack arrives when the buffer is full, the latest pack among the packs in the buffer is discarded and a new pack is inserted into the buffer again. At this time, the I-picture is not discarded.

Step 5: This is a case in which packs continue to arrive while the buffer is full. In this case, the packs that arrived later among the packs in the buffer are intensively discarded, resulting in a phenomenon in which the screen is not connected. To prevent this, the number of discarded packs is counted. If more than a certain number of packs are discarded consecutively, the odd-numbered packs are discarded from the beginning of the buffer. Of course, even at this time, the less damaged B-picture is discarded first, and the I-picture is not discarded.

3.3 Experimental results

In this section, some experimental results are given to illustrate: (1) the invisibility of embedded watermark; (2) watermark robustness. Standard video sequence “Susie”, “Garden”, and “Football” are taken as test sequences. Frames in these sequences are 256 gray-level images. A binary image is taken as watermark. PSNR is used in the experimental results, it is a well-known measure for video or image quality after being processed, and it is computed as:

$$\mathrm{PSNR} = 10 \times \log\left(\frac{255^2}{\mathrm{MSE}}\right), \quad \mathrm{MSE} = \frac{\sum_{n=1}^{\mathrm{FrameSize}} \left(I_n - I_n'\right)^2}{\mathrm{FrameSize}},$$

Where I_n is the nth pixel of the original frame, I'_n is the nth pixel of the processed frame.The difference image between the original image and the image received by the terminal is extracted, and the I-picture loss rate is calculated from the difference image. The best way to reduce the loss of the watermark is to reduce the loss rate of the I-picture. This is also the purpose of this paper.

The left column of Table 1 shows the PSNR of an image for transmission in which a watermark is embedded. From Table 1, PSNR is bigger than 30 dB, so the watermarked I-Picture has good quality.The right column of table 1 shows the loss rate of I-pictures. Compared with the general VOD service, when the algorithm presented in this paper is used, the I-picture loss rate is small.

Table 1. PSNR of watermarked I-Picture and loss ratio of watermark

Video sequence	PSNR of watermarked I-Picture	Loss ratio of watermark(%)
Susie	31.12	0.62
Garden	31.78	0.93
Football	32.24	0.79

IV. CONCLUSION

Time delay during the multimedia data transmission through internet causes non-synchronized phenomena between video and sound when they play back. To prevent this quality lowering, intra and inter-media synchronizations are necessary to secure seamless and harmonized playback. Generally, the process to maintain the time of one or multiple media streaming is called as multimedia synchronization. During the synchronization process, loss of cell is occurred.If playback of pictures is not performed to be discarded due to the difference between the transmission rate in the server and playback processing rate in the client, loss of the watermark is very likely. Video decoder in the client checks the number of available pictures for playback periodically to deliver its information to the server, and the server controls the transmission not exceeding the number of available pictures for playback, using the network resources, efficiently. However, in this case, the watermark inserted in the video is also deleted.

In this paper, a synchronization algorithm is proposed to be able to deliver watermark to the final display terminal, minimizing watermark deletion during the synchronization process for video streaming.In the future research project, we plan to study the algorithm that can restore the watermark deleted during the video transmission process.

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REFERENCES

[1]. J. Takahashi, H. Tode, K. Murakami: QoS enhancement methods for MPEG video transmission on the Internet. In: Proc. ISCC 2002, 106-113(2002).
[2]. S. H. Park: A Novel I-picture Arrangement Method for Multiple MPEG Video Transmission. In: The journal of the Korea Institute of Maritime Information & Communication Sciences, vol.9, no.2, 277-282 (2005).

[3]. X. Kong, Y. Liu, and H. liu: Adaptive video watermarking scheme. In: IEEE Pacific-Rim Conference On Multimedia (PCM), LNCS 2195, pp. 933-939, Oct. Springer-Verlag(2001).
[4]. Y. Wang, Al.Pearmain: Blind MPEG-2 video watermarking robust against geometric attacks- A set of approaches in DCT domain. In: IEEE Trans, on Image Processing, Vol. 15, No.6, 1536-1543 (2006).
[5]. J. S. Yoon, S. H. Lee, Y. C. Song, B. J. Jang, K. R. Kwon, M. H. Kim: Robust Blind Video Watermarking against MPEG-4 Scalable Video Coding and Multimedia Transcoding. In: The journal of the Korea Multimedia Society, Vol.11, No.10, 1347-1358 (2008).
[6]. D. J. Cho, K. Y. Yoo: The Study on Development of a Multimedia Synchronization Algorithm for Internet Based VOD Services. In: The journal of the Korea Information Processing Society, vol.8-B, no.1, 74-80 (2001).
[7]. ISO/IEC JTC1/SC29/WG11/N5231, Applications and requirements of scalable video coding N6830, Palma de Mallorca, Spain, Oct. (2004).
[8]. ITU-T document, Joint draft 10 of SVC amendment, Joint video team JVT-W201, JVT 23rd meeting, San Jose, USA, Apr. (2007).
[9]. ISO/IEC JTC1/SC29/WG11, Joint scalable video model (JSVM) 7.0 reference encoding algorithm description, N7556, Klagenfurt, Austria, July (2006).
[10]. J. C. Jung, Hiroshi Fujiwara: Applied MPEG by Figure, Japan Multimedia Communications Research Society, published by ASCII Corporation(1997).
[11]. R. S. Ryu et al.: MPEG system, Daeyung press(1997).
[12]. Yu Xiaoyan, Wang Chengyou, Zhou Xiao: A Survey on Robust Video Watermarking Algorithms for Copyright Protection. In: Applied Sciences, Vol. 8, Iss. 10 (2018).

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