Estimation of Network Jitter Using Delivery Instant Parameter of MPEG MMT Protocol

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***Abstract*—Recently, ISO MPEG has developed a new media transport technology called MPEG Media Transport (MMT) protocol for the next generation hybrid video delivery services over IP networks. In this paper, we propose an efficient estimation of network jitter using delivery instant parameter of MPEG MMT protocol.**

Keywords-MPEG MMT; jitter estimation; video delivery over IP networks.

I. INTRODUCTION

ISO MPEG developed the MPEG-H standard suite (ISO/IEC 23008) for the delivery of audio-visual information compressed with high efficiency over a heterogeneous environment. The MPEG-H suite consists of three functional areas: High-Efficiency Video Coding (HEVC), 3D audio, and MPEG Media Transport (MMT) [1].

Maintaining timing relationships among packets in a single media stream or between packets from different media streams is an essential role of MMT. It is the function of the synchronization and de-jittering algorithms to re-adjust timing relationship between the MMT packets to assure synchronized playback [2]. Thus, delivery of time constrained MPEG media on time, according to their temporal requirements, is an important goal of MMT. For this purpose, MMT provides delivery instant parameter in its packet header to specify syntax and semantics of a timing model to be used by the delivery functions [3]. By using the delivery instant parameter, it is possible to estimate end-to-end transmission delay or network jitter. The total end-to-end delay should be kept constant for continuous decoding and playback at the receiver.

In this paper, we propose an efficient method to estimate the network jitter for predicting constant end-to-end delay and estimating appropriate buffering time using delivery instant parameter of MPEG MMT protocol.

The organization of this paper is as follows. In Section II, we describe conventional network jitter estimation method. Section III describes the proposed network jitter estimation method using MMT protocol. Concluding remarks and future works are given in Section IV.

II. CONVENTIONAL NETWORK JITTER ESTIMATION METHOD

Internet Engineering Task Force (IETF) provides timestamps, such as Real-time Transport Protocol (RTP) timestamp [4], and Network Time Protocol (NTP) timestamp. Each RTP packet carries a 32 bits timestamp which reflects the sampling instant of the first byte in the data packet. For MPEG payloads, the sampling instant is derived from a 90 KHz clock. Besides RTP timestamps, an NTP timestamp is also transported in the RTCP Ssender Report (SR) packet.

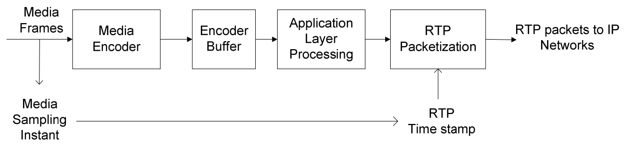

Figure 1. Media sampling instant used as an RTP timestamp in RTP packetization.

Figure 1 shows a media sampling instant used as an RTP timestamp in RTP packetization. As shown in Figure 1, if RTP packets are generated periodically, the nominal sampling instant as determined from the sampling clock is used as an RTP timestamp, not a reading of the system clock. In RTP, the inter-arrival jitter is defined to be the mean deviation of the difference in the packet spacing at the receiver compared to the sender for a pair of packets. In IETF RFC 3550 standard, the following method is used to estimate network transmission jitter. If Si is the RTP timestamp from packet i, and Ri is the time of arrival in RTP timestamp units for packet i, then for two packets i and j, the difference in packet spacing at the receiver is expressed as

$$D(i, j) = (Rj - Ri) - (Sj - Si) = (Rj - Sj) - (Ri - Si)\,. \quad (1)$$

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The inter-arrival jitter is calculated continuously as each data packet *i* is received according to the following formula

$$J(i) = J(i-1) + (|D(i-1,i)| - J(i-1))/16 \,. \quad (2)$$

Because the jitter calculation shown in (2) is based on the RTP timestamp, any variation in the delay between the sampling instant and the time the packet is actually transmitted will affect the resulting jitter that is calculated. Such a variation in delay would occur for video encodings because the timestamp is the same for all the packets of one video frame but those packets are not all transmitted at the same time. Thus, the variation in delay until transmission could reduce the accuracy of the jitter calculation as a measure of the behavior of the network by itself.

III. Proposed Network Jitter Estimation Method Using MMT Protocol

Preservation of timing relationships among packets in a single MMTP packet flow or between packets from different MMTP packet flows is an important feature of MMT. In the conventional method, if the time gap between the sampling instant and the transmission instant could be always kept as a constant, there would be no problem in the accuracy of the calculated jitter. However, the problem could occur in transmitting a video frame with huge resolution, such as 4K/8K UHD video, as an example.

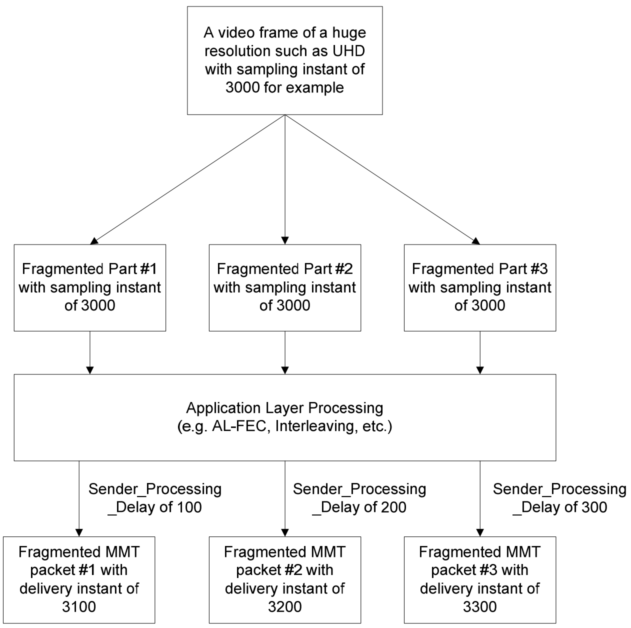

Figure 2. A video frame of huge resolution is fragmented into three different delivery packets, each with different delivery time.

Due to its big size of 4K/8K UHD video frame, the compressed bitstream may need to be fragmented into several delivery packets before transmission as shown in Figure 2.

Although the three fragmented packets are originated from the same video frame with the same sampling instant, they are transmitted at different delivery time due to additional application layer processing such as Applicatoin Layer Forward Error Correction (AL-FEC) and Interleaving. Thus, by assigning the delivery instants to MMT packets as shown in Figure 3, we would be able to achieve accurate estimation of inter-arrival jitter in MMT service. Instead of using (1), we can employ the following equation to obtain the difference in packet spacing for two MMT packets *i* and *j* at the receiver:

$$D_{MMT}(i,j) = (T_{A,j} - T_{A,i}) - (T_{D,j} - T_{D,i}) = (T_{A,j} - T_{D,j}) - (T_{A,i} - T_{D,i}) \,. (3)$$

In (3), $T_{D,i}$ and $T_{D,j}$ denote the time instant of delivering two MMT packets *i* and *j*, respectively. $T_{A,i}$ is defined as the time of MMT packet arrival in D-layer timestamp units (may be in NTP timestamp format) for MMT packet *i*. The inter-arrival jitter is calculated continuously as each MMT packet *i* is received according to the following formula:

$$J_{MMT}(i) = J_{MMT}(i-1) + (|D_{MMT}(i-1,i)| - J_{MMT}(i-1))/16 \,. \ (4)$$

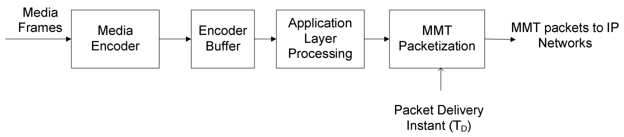

Figure 3. Delivery instant information to be included in the MMT packet header in D-layer packetization.

The proposed algorithm described in Section III has been included in the International Standard document of MMT [5]. We will focus our attention on the test and implementation of the proposed algoirhtm using MMT protocol to verify the performance of the proposed network jitter estimation method in a real video delivery environments over IP networks.

IV. Conclusion and Future Work

In this paper, we proposed an efficient method to achieve accurate estimation of inter-arrival jitter using delivery instant parameter of MPEG MMT at the receiver side. To achieve the accurate estimation of network jitter, the sender side needs to transmit delivery instant information based on the MMT timing model to the receiver side. As our future work, we are going to verify the

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performance of the proposed network jitter estimation method by implementing the algorithm in a real video delivery service over IP networks.

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