Delay Constrained ARQ Mechanism for MPEG Media Transport Protocol Based Video Streaming over Internet

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***Abstract*—MPEG Media Transport (MMT) is a new international standard aiming at addressing the emerging multimedia services over heterogeneous packet-switched networks including Internet and broadcasting networks. Due to the heterogeneous characteristics of the broadcast and broadband networks, MMT provides an efficient delivery timing model to provide inter-network synchronization, measure various kinds of transmission delays and jitter caused by the transmission delay, and re-adjust timing relationship between the MMT packets to assure synchronized playback. Exploiting the delivery timing model, it is possible to accurately estimate round-trip time (RTT) experienced during MMT packet transmission. Based on the measured RTT, we propose an efficient delay constrained ARQ (Automatic Repeat reQuest) scheme which is applicable to MMT protocol based real-time video communication over IP networks.**

Keywords-video communication over Internet; ARQ scheme; error control processing; MPEG Media Transport (MMT).

I. INTRODUCTION

In the recent years, digital broadcasting services and IP-based multimedia services over the Internet including mobile Internet have started being integrated and converged [1]. With this trend, there have been so many changes in the multimedia service environments such as media content delivery networks, diverse video signals, 4K/8K video transport systems, and various client terminals displaying multi-format signals. It has become clear that the MPEG standard has been facing several technical challenges due to the emerging changes in those multimedia service environments [2]. So, to address these technical challenges to existing and emerging MPEG standards, ISO MPEG has developed an MPEG-H standard suite (ISO/IEC 23008) for the delivery of audio-visual information compressed with high efficiency over heterogeneous environment. MPEG-H suite consists of three functional areas: High Efficiency Video Coding (HEVC) [3], 3D audio, and MPEG Media Transport (MMT) [4].

In order to deploy efficient solutions for the transport of HEVC video in an interoperable fashion, especially given the recent increased demand of multimedia delivery in the heterogeneous network environment, MPEG has launched a new standardization work item, called MMT since the middle of 2010. MMT has been working on addressing technical challenges of existing standards due to recent changes of multimedia delivery and consumption environments and new requirements from emerging use cases and application scenarios in the area of multimedia services [5]. MMT aims to overcome current limitations of available standards for media streaming by addressing streaming format that is transport and file format friendly, cross-layer optimized between video and transport layer, error resilient for MPEG streams, convertible between transport mechanisms and content adaptation to different networks [6]. The challenge to error control schemes for the real-time video communication is focused on how to endeavor to recover the packet loss and then to reduce impairment to the playback quality [7]. Many different error control techniques have been proposed to solve these issues and Automatic Repeat request (ARQ) has been known as one of the promising solutions [8].

MMT provides delivery timing model as a means to calculate jitter and the amount of delay introduced by the underlying delivery network, so that constant delay for data stream can be achieved [4]. Using the delivery timing model, an MMT receiving entity provides information in the feedback to allow the MMT sending entity to calculate the RTT. To circumvent the packet loss problem, MMT employs ARQ function as a basic error control technique in which the MMT receiving entity asks for retransmission when packet loss is detected at the receiver side [6][7]. When the RTT is so high that the retransmitted packet will not arrive in time, the retransmission will not improve the quality. The current ARQ function of MMT, however, could result in many late packets that arrive after play-out deadline because it does not consider round-trip delay caused for the retransmission. To circumvent this problem, we propose an efficient delay constrained ARQ scheme for MMT packet based real-time video communication.

The remainder of this paper is organized as follows. In Section II, we overview MMT technology and summarize the basic ARQ process supported in MMT. In Section III, we describe the proposed delay-constrained ARQ scheme. The experimental results and performance evaluation are presented in Section IV. Finally, concluding remarks are provided in Section V.

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II. OVERVIEW OF MMT

In order to support efficient delivery and effective consumption of coded media data for multimedia services over packet-switched networks including IP networks and digital broadcasting networks, MMT defines three functional areas: encapsulation functional area, delivery functional area, and signaling functional area as illustrated in Figure 1 [4].

The encapsulation functional area defines the logical structure of media content, the Package, and the format of the data units to be processed by an MMT entity and their instantiation with ISO base media file format (ISOBMFF) as specified in ISO/IEC 14496-12. It produces Media Processing Unit (MPU) as an output.

The delivery functional area defines an application layer transport protocol including the payload format required for transferring encapsulated media data from one network entity to another. The payload format is defined to enable the carriage of encoded media data which is agnostic to media types and encoding methods.

The signaling functional area defines formats of signaling messages to manage delivery and consumption of media data.

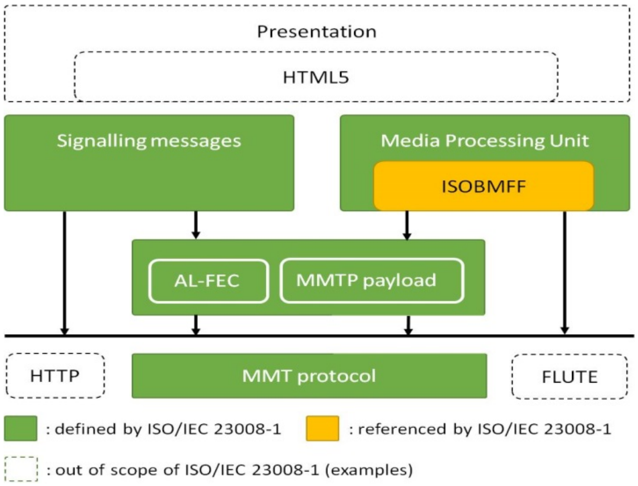


Figure 1. MMT functional areas and interface

The MMT technology adopts ARQ technique as an error control method for data transmission over error-prone networks. Like the general behavior of the ARQ scheme using a Negative Acknowledgment (NACK) [8], in case of packet loss, a NACK is sent back from the receiving entity to the sending entity and the sending entity retransmits the lost packet [6].

Figure 2 illustrates the basic operation of the MMT ARQ process [4]. The basic operation of ARQ process is as follows. The first step is the generation of ARQ configuration (AC) message by MMT sending entity, then it is delivered to the receiving entity. MMT receiving entity stores the ARQ configuration information. MMT receiving entity continues with receiving MMT media packets and checks whether there is a lost packet or not.

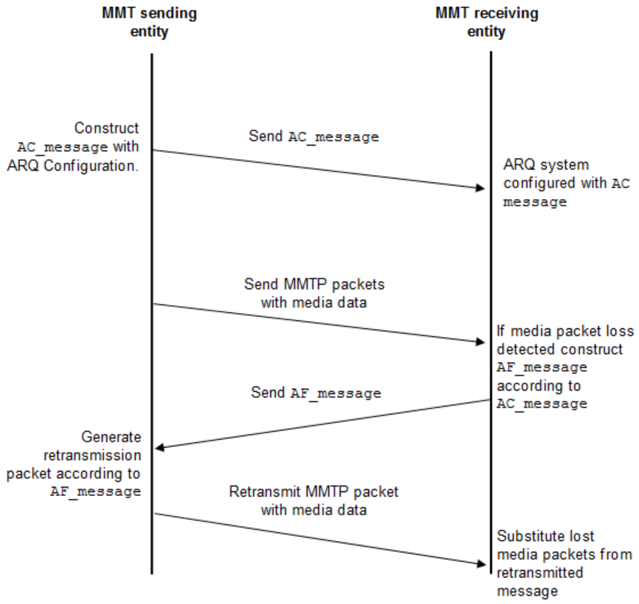


Figure 2. Operation of MMT ARQ process

Once MMT receiving entity determines that a packet has been lost, an ARQ feedback (AF) message is generated according to the configuration information defined in the AC message. The AF message is sent to MMT sending entity which will generate the retransmission packet to be sent to MMT receiving entity. MMT receiving entity is able to substitute the lost MMT packet with the retransmission packet.

III. PROPOSED DELAY CONSTRAINED ARQ MECHANISM FOR MMT PROTOCOL

One of the well-known problems in ARQ happens if networks experience severe congestion. In this situation, the retransmission packet leads more congestion and causes the further network degradation. Furthermore, in case of real-time video communication, the ARQ will be successful only if the retransmitted packet is received before its *arrival deadline*. When the RTT is so high that the retransmitted packet will not arrive in time, the retransmission will not improve the quality [7].

The objective of the proposed delay constrained ARQ is to suppress retransmitting requested lost packets that will not arrive in time for playback. By using the delay constrained retransmission, the MMT sending entity can avoid unnecessary retransmission of the out-of-date packets and therefore minimize the probability of wastefully retransmitted packets. This results in a reduced amount of data traffic wastefully injected into the network. The proposed delay constrained ARQ uses arrival deadline to decide whether MMT sending entity retransmits the requested lost packet or not.

Figure 3 shows an exemplary timing diagram of the proposed delay constrained ARQ. In Figure 3, arrival deadline denotes the maximum tolerable latency for the requested retransmission packet to arrive at the receiver. If the retransmitted packet arrives later than arrival deadline, even if the packet arrives intact, it is regarded as useless and discarded so that decoding may not occur. Thus, the recovery latency for the lost packet should be taken into account for the real time video delivery wherein late repair becomes useless because of the real time nature of the data. RTT information which is crucial for the delay constrained retransmission is already available to the sender by the reception quality feedback (RQF) message supported in the MMT standard. This could be achieved by using the timestamp field existing in the MMT packet header. This timestamp field specifies the time instance of MMT packet delivery based on Coordinated Universal Time (UTC), which corresponds to the sending time at the first byte of MMT packet.

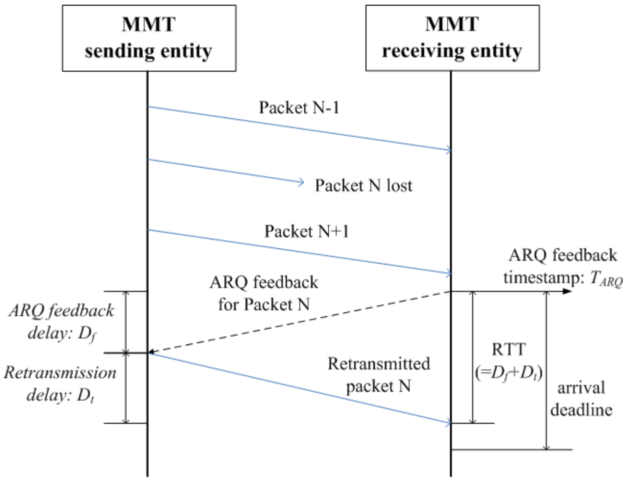


Figure 3. Timing diagram of the proposed delay-constrained ARQ

Based on the above analysis, when the MMT receiving entity detects the loss of packet, the receiver sends AF message to the MMT sending entity, and then the sender takes the decision procedure for delay constrained retransmission as shown in Figure 4 to decide whether to transmit the requested lost packet or not. The estimated RTT can be obtained *a priori* at the sending entity by RQF message provided in MMT [6]. For more accurate estimation of the RTT, up-to-date ARQ feedback delay (D_f) value can be informed to the sending entity by enclosing ARQ feedback timestamp (T_{ARQ}) into the AF message. ARQ feedback timestamp corresponds to the time instant of sending the AF message to the MMT sending entity. Using the ARQ feedback timestamp, T_{ARQ}, the sending entity can obtain up-to-date ARQ feedback delay (D_f). And this updated D_f value can be used to compute up-to-date RTT. Finally, if the RTT is estimated to be less than the arrival deadline, sender retransmits the requested lost packet to the receiver. Otherwise, the sender decides not to transmit the requested lost packet to the receiver. In order to implement the proposed delay constrained ARQ scheme, the ARQ feedback timstamp and the arrival deadline information needs to be included in the AF message of MMT. The AF signaling format supporting the proposed delay constrained ARQ scheme could be found in [9] and [10], and has been approved to be included in the standard document of MMT AMD1 (Amendment 1) in the MPEG meeting.

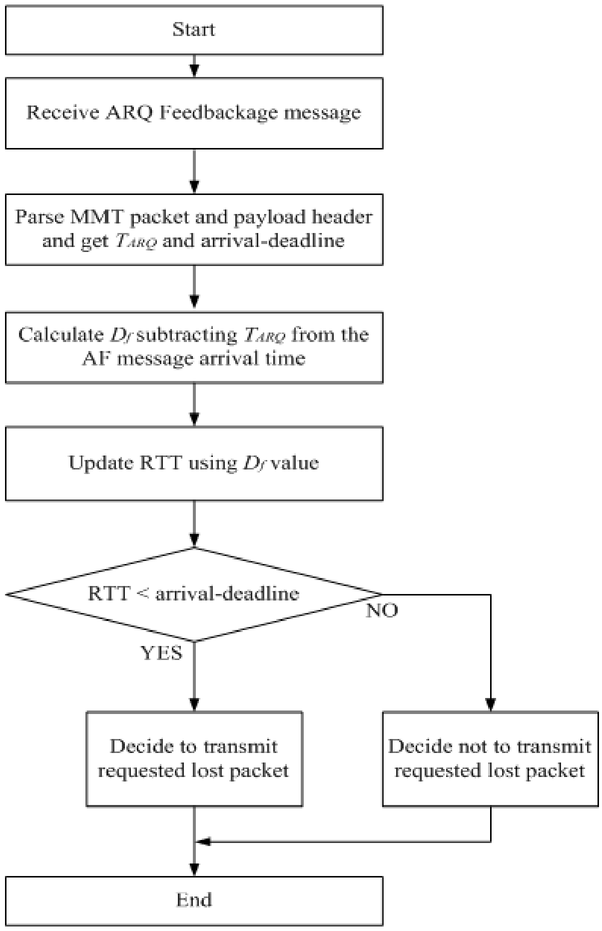


Figure 4. Procedure to decide whether to retransmit the lost packet at the MMT sending entity

The arrival deadline can be obtained at the receiver side by considering the remaining amount of safely arrived packets in the receiver buffer when the ARQ feedback message is prepared for sending. Therefore, we exploit the remaining amount of safely arrived packets in the receiver buffer and average bit rate of the MMT packet stream when the ARQ feedback message is prepared for sending.

IV. EXPERIMENTAL RESULTS

To evaluate the efficacy of the proposed delay constrained ARQ scheme, we performed extensive experiments over the MMT protocol-based HEVC video streaming system. The video streaming system consisted of an MMT sending entity and receiving entity. Before initiating video streaming, the AC message which includes the retransmission policy to be adopted by the MMT sending and receiving entities in the event of packet loss is sent to the receiving entity by the ARQ policy manager. The MMT sending entity retains an MMT packet in the buffer until the timeout, and it is thus available for retransmission. If MMT sending entity receives an AF message requesting retransmission of the lost packet, it checks whether the lost packets is available in the retransmit buffer and it retransmits the packet only if the arrival deadline value is greater than RTT. To verify the performance of the proposed ARQ scheme in an error-prone video transmission environment, the NIST-Net [11] Linux-based network emulation tool was used to emulate the packet-loss network environment. We used the *Stockholm* video sequence with HD 1080P resolution. The *Stockholm* video sequence was coded by an HEVC encoder, which is being increasingly used for broadcasting and mobile multimedia applications. The run-time of the generated video streaming was 4 min 39 s, and the number of generated MMT packets to be delivered was 193,042 in total. The size of each MMT packet was 1500 bytes, and the receiving buffer was set to store 300 MMT packets.

In the first experiment, the network delay was set to 50 ms while the packet loss rate has varied from 3% to 10% by the NIST-Net network emulator. Figure 5 shows the ratio of the number of renounced retransmission to the total number of retransmission request for various packet loss rates. We can observe that the ratio of the renounced retransmission increases as the packet loss rate increases. This is due to the fact that as the packet loss rate increases, the arrival deadline for the lost packet is generally decreased because of the less safely arrived MMT packets in the receiver buffer.

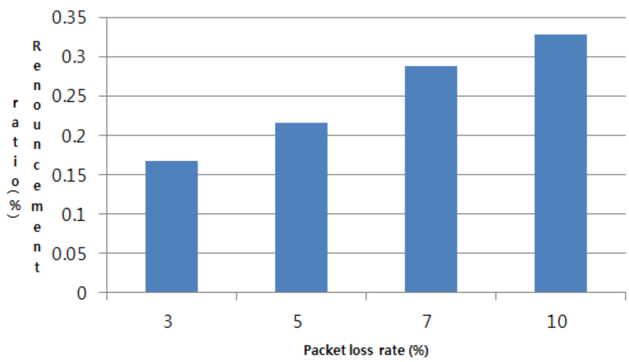


Figure 5. Ratio of renounced retransmission for various packet loss rates under fixed network delay of 50 ms

Figure 6 shows the saved network bandwidth resulted by the proposed delay constrained ARQ scheme. We can observe that the saved network bandwidth increases as the packet loss rate increases. As the packet loss rate increases, the ratio of the renounced retransmission also increases. This results in avoiding unnecessary retransmission of the out-of-date packets and therefore reduces amount of data traffic wastefully injected into the network.

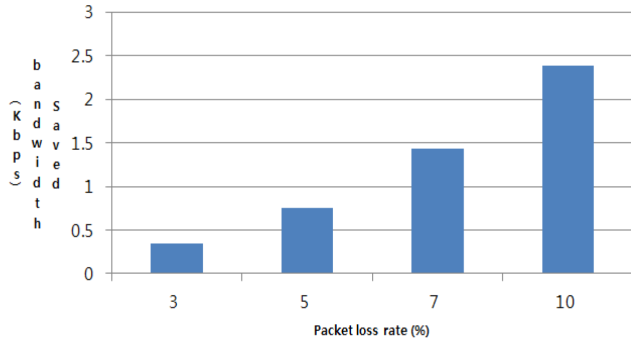


Figure 6. Saved network bandwidth for various packet loss rates under fixed network delay of 50 ms

In the second experiment, the packet loss rate was set to 10% and the network delay has varied from 100 ms to 400 ms by the NIST-Net network emulator. When compared to the simulation conditions for the first experiment, the channel condition significantly deteriorated, which showed a higher packet loss rate and much longer network delays experienced during the video streaming.

Figure 7 shows the ratio of the number of renounced retransmissions to the total number of retransmission requests for various network delays under the same test condition. It is evident that the ratio of the renounced retransmissions significantly increased as the network delays increased. This was due to the fact that, as the network delay increased, the arrival deadline for the lost packet generally decreased on account of fewer safely arrived MMT packets in the receiver buffer.

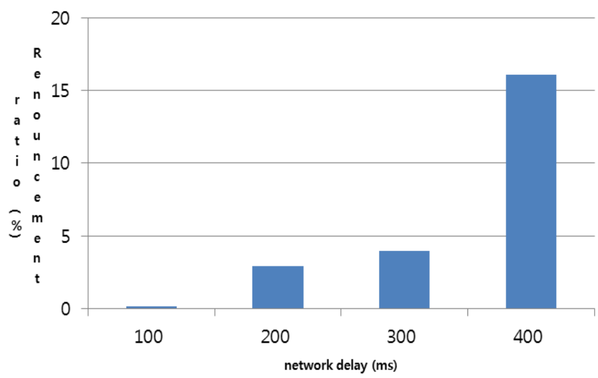


Figure 7. Ratio of renounced retransmission for various network delays under the fixed packet loss rate of 10%

Figure 8 shows the saved network bandwidth resulting from the proposed scheme. We can observe that the saved network bandwidth increased as the network delay increased. As the network delay increased, the ratio of the renounced retransmission also increased. This resulted in the avoidance of unnecessary retransmissions of out-of-date packets and therefore reduced a significant amount of data traffic wastefully injected into the network. As shown in Figure 8, approximately 190 Kbps were saved for the network delay case of 400 ms. This saved bit-rate amount corresponded to 13% of the total data traffic generated by the transmitted MMT packets.

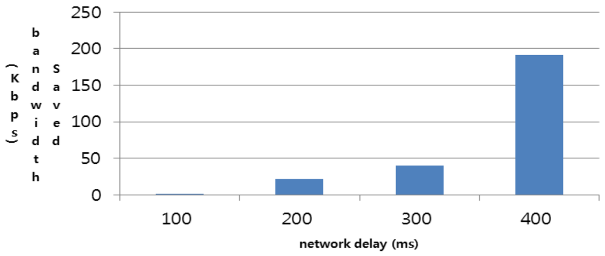
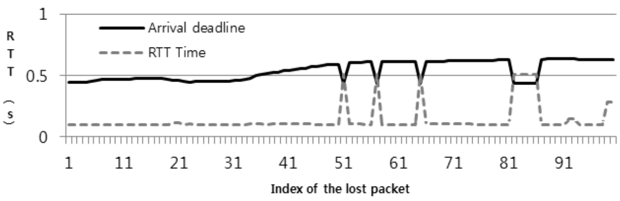
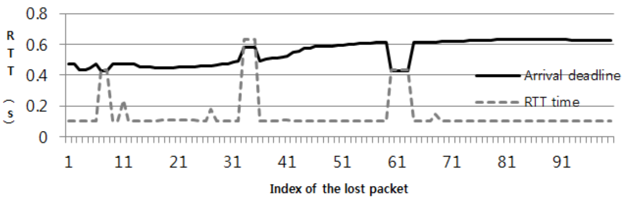


Figure 8. Saved network bandwidth for various network delays under the fixed packet loss rate of 10%



(a) packet loss rate of 5%



(b) packet loss rate of 7%

Figure 9. Comparison of RTT and arrival deadline for 100 randomly chosen lost packets during video streaming over various packet loss rates

In Figure 9, we compare RTT and arrival deadline values for one hundred randomly chosen lost packets during video streaming over packet loss rates of 5% and 7%. If the RTT is greater than the arrival deadline, the sending entity decides not to retransmit the requested lost packet to the receiving entity. Otherwise, the sending entity decides to retransmit the lost packet to the receiving entity. As shown in Figure 7, as the packet loss rate increases, the portion of the lost packet which has greater RTT value than the arrival deadline also increases.

V. CONCLUSIONS

In this paper, we proposed a delay constrained ARQ scheme to enhance the effectiveness of the basic ARQ process of MPEG MMT standard for video streaming over Internet. Using the proposed ARQ scheme, it is possible to avoid unnecessary retransmission of the out-of-date packets and therefore minimize the probability of wastefully retransmitted packets.

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REFERENCES

[1] S. Aoki, K. Aoki, H. Hamada, Y. Kanatsugu, M. Yamamoto, and K. Aizawa, “A new transport scheme for hybrid delivery of content over broadcast and broadband,” Proc. Broadband Multimedia Systems and Broadcasting (BMSB), 2011, pp. 1-6.

[2] Y. Lim, K. Park, J. Lee, S. Aoki, and G. Fernando, “MMT: An emerging MPEG standard for multimedia delivery over the Internet,” IEEE Multimedia, vol. 20, no. 1, Jan. 2013, pp. 80-85.

[3] ISO/IEC 23008-2, High efficiency coding and media delivery in heterogeneous environments – MPEG-H Part 2: High Efficiency Video Coding (HEVC), 2013.

[4] ISO/IEC 23008-1, High efficiency coding and media delivery in heterogeneous environments – MPEG-H Part 1: MPEG Media Transport (MMT), 2014.

[5] Y. Lim, S. Aoki, I. Bouazizi, and J. Song, “New MPEG transport standard for next generation hybrid broadcasting system with IP,” IEEE Trans. Broadcasting, vol. 60, no. 2, Feb. 2014, pp. 160-169.

[6] ISO/IEC 23008-1 AMD 1, Additional technologies for MPEG Media Transport (MMT), 2014.

[7] Y. Wang, and Q. Zhu, “Error control and concealment for video communication: A review,” Proc. of the IEEE, vol. 86, no. 5, May 1998, pp. 974-997.

[8] M. Podolsky, S. McCanne, and M. Vetterli, “Soft ARQ for layered streaming media,” Journal of VLSI Signal Processing, vol. 27, 2001, pp. 81-97.

[9] ISO/IEC JTC1/SC29/WG11 m32363, Update of ARQ feedback message for delay-constrained retransmission in MMT, 2014.

[10] ISO/IEC JTC1/SC29/WG11 m35228, Update of Implementation guideline for delay constrained ARQ in MMT, 2014.

[11] NIST-Net, Software provided by National Institute of Standards and Technology (NIST), available on-line at http://www-x.antd.nist.gov/nistnet, [retrieved: December, 2014].