Synchronized Playback in Rapid Acquisition of Multicast Sessions

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Abstract

When watching the same IPTV channel, different TV sets may not render the same picture and the associated audio at the same moment. The variation in end-to-end delay that resulted in such asynchronous playback between users is referred to as inter-user playback delay. Unicast based rapid acquisition of multicast RTP sessions (RAMS) as specified in [I-D.ietf-avt-rapid-acquisition-for-rtp] is an important technique in achieving fast channel switching in IPTV as well as other multicast applications. In addition, RAMS also significantly relaxes the requirement of relatively short random access point period in encoding of video streams in multicast applications, thus allowing significantly improved compression efficiency. However, on the other hand, the use of RAMS increases inter-user playback delay, which makes users receiving the same multicast session playback the same content asynchronously. This document specifies a mechanism to help reduce inter-user playback delay caused by the use of RAMS.

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1. Introduction

The Internet-Draft "Unicast-Based Rapid Acquisition of Multicast RTP Sessions" in [I-D.ietf-avt-rapid-acquisition-for-rtp] presents a method based on unicast burst stream for rapid acquisition of multicast RTP sessions (RAMS), thus to reduce the waiting time for the so-called Reference Information (RI). This method is effective in reducing channel switching and tune-in delay in multicast applications, such as IPTV. The RI typically starts at an access unit that is a random access point. In RAMS, RTP receivers start playback from the random access point from which the RI starts when they switch from one multicast session to another. As the unicast burst stream starts from the RI and is transmitted as fast as possible and faster than the media rate, on average the receiver can start processing the unicast burst stream faster, because it does not need to wait for the next random access point, as it does if it directly joined the multicast group.

Another important benefit brought by RAMS is that significantly improved coding efficiency for video streams is possible. In conventional multicast applications, video streams must be encoded with frequent random access points, e.g. 0.5 to 1 second, to allow new receivers to tune-in or existing users to switching from another multicast session. Random access points typically contain intra-coded pictures, for which the compression efficiency is significantly, e.g., several to ten times, lower than inter-coded pictures. Therefore, the less the random access points, the higher the coding efficiency. When RAMS is in use, random access period length is not the decision factor for the tune-in or channel switching delay. This means that the video random access point frequency can be significantly reduced, leading to significantly improved compression efficiency.
In a video communication system, an end-to-end delay is unavoidable. Typical end-to-end delay components are transmission delay, receiver buffering delay (which also handles transmission jitter), decoding buffering delay, output buffering delay, and other processing delays. Use of RAMS causes additional end-to-end delay, the amount of which is equal to the time difference between the latest RTP timestamp of primary multicast packets buffered in the RS when the unicst burst starts and the RTP timestamp of the starting point of the unicast burst.

In multicast applications, receivers receiving the same multicast session can have different end-to-end delays and thus the playback of the same content among the receivers is not synchronized. In this document, a delay in playback time of the same content between any two users is referred to as inter-user playback delay (IUPD). This document further refers to the typical end-to-end delay (when RAMS is not in use) as Common End-to-end Delay (CED). Among the delay components of CED, jitter buffer delay is the main part. Typical set-top box de-jitter buffers can store 100-500 ms (of SDTV) video, so network jitter must be within these limits and delay variation beyond these limits will manifest itself as loss [TR-126]. Compared to jitter buffer delay under the level of several hundreds of milliseconds, the acquisition of Reference Information by RAMS may result in much longer delay which depends on the period of random access points, which sometimes are, vaguely, referred to as Group of Pictures (GOP) length (distance between key or I-frames). This document refers to this additional end-to-end delay caused by the use of RAMS as Rapid Acquisition Playback Delay (RAPD).

Due to IUPD, different users may watch different pictures from different TV sets when watching the same IPTV content. In some application scenarios, e.g., remote education or a discussion room for an ongoing TV program in a social network, different users may be discussing the same content received through multicast. In this case, an obvious playback synchronization loss due to excessive inter-user playback delay can generate bad user experience.

As mentioned above, a disadvantage of RAMS is that the use of the technique causes RAPD for each user. Receivers are not synchronized in sending RAMS requests. Regardless of when the receiver starts the RAMS request (i.e. joins the program), the playback will start from a previous random access point. Given the starting point, the closer to the next random access point the receiver joins the program, the longer the RAP will be. Thus, the use of RAMS increases IUPD.
When obvious IUPD affects user experiences in above application scenarios, some actions must be taken. One way is to constrain the use of long random access period length, which significantly hurts video compression efficiency. In this document, we describe some mechanisms to reduce IUPD due to the use of RAMS and to allow the use of long random access period length for improved compression efficiency at the same time.

2. Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119.

3. Definitions

This document uses the following acronyms and definitions:

Inter-User Playback Delay (IUPD): The playback delay between different users, for the same content transmitted on the same multicast session, due to variations in end-to-end delays.

Common End-to-end Delay (CED): The end-to-end delay when RAMS is not in use, which consists of transmission delay, receiver buffering delay (which also handles transmission jitter), decoding buffering delay, output buffering delay, and other processing delays.

Rapid Acquisition Playback Delay (RAPD): The additional end-to-end delay introduced by the use of RAMS. The value is equal to the time difference between the latest RTP timestamp of primary multicast packets buffered in the RS when the unicast burst starts and the RTP timestamp of the starting point of the unicast burst.

4. Reducing Inter-User Playback Delay due to RAMS

Inter-User Playback Delay (IUPD) is caused by the variation in end-to-end delay, which includes Common End-to-End Delay (CED) and Rapid Acquisition Playback Delay (RAPD). CED is primarily determined by jitter buffering delay which is in the range from 100 ms to 500 ms [TR-126]. In practice, IUPD caused by CED, and primarily by jitter buffer delay variations, should be under the level of tens of milliseconds. In contrast to this small range of IUPD values caused by CED, the IUPD caused by RAPD can be much
higher, up to a few seconds or even to ten seconds, as long period of random access points can be used for significantly improved video compression efficiency when RAMS is in use.

To lower IUPD the receiver can speed up video playback until it catches up with the primary multicast stream. The procedures are outlined below.

The playback synchronization of receivers uses the Primary Multicast Stream as a reference point to address RAPD due to the use of RAMS. Each receiver keeps the synchronization with the primary multicast stream so that all receivers can keep an approximate synchronization in playback. The advantage is that it does not need every receiver to get playback synchronization information of other receivers, and thus scalability is not an issue as the procedure does not depend on the number of receivers.

The mechanism for speeding up video frames uses a method of skipping video frame. In other words, one frame per an interval of some video frames is skipped until the number of extra video frames has been skipped. This way, the additional end-to-end delay, RAPD, can be compensated, and at the same time a smooth playback is ensured.

The mechanism involves the following changes to the RAMS method:

1) When the RTP Receiver (RR) sends a rapid acquisition request for the new multicast RTP session, the request MAY contain additional information indicating whether RR supports inter-user playback delay reduction.

2) When the Retransmission Server (RS) receives the RAMS-R message and decides to accept it, RS MAY include the following additional information in the RAMS-I message to RR:

   a) N, the playback delay reduction target in number of frame durations; and

   b) V, recommended interval, in frames, between two continuous events for skipping of one frame.

When the RAMS-R message indicates that RR supports inter-user playback delay reduction, RS SHOULD include the above information in the RAMS-I message.
Retransmission server (RS) can precisely determine the value of N by detecting the time of RAMS-R, as the amount of additional end-to-end delay introduced by the use of RAMS, i.e. RAPD, is known to RS. The value is equal to the time difference between the latest RTP timestamp of primary multicast packets buffered in the RS when the unicast burst starts and the RTP timestamp of the starting point of the unicast burst. The value V is a recommended skip frame interval and the value must be chosen such that there is no noticeable audio distortion. For a video frame rate of 30 frames per second, typically when V is greater than 15 there is no noticeable audio distortion.

3) When RR receives an RAMS-I message containing the above information, it SHOULD speed up media rendering during playback taking into account the information as follows. During the speedup playback, after each V frames, one frame is skipped as if it was not present, and the presentation time of each remaining frame is shifted earlier by one frame duration, until totally N frames have been skipped. Receivers will playback the media content with its original speed after totally N frames have been skipped. Note that decoding remains the same as if speedup playback was not in use.

Besides the above mechanism, RS can use selective transmission of packets in the beginning of the unicast burst, by taking advantage of the temporal scalability of video bitstreams.

Video bitstreams are typically temporally scalable, in the sense that a portion of the bitstream can be extracted and decoded with a lower frame rate. Conventionally video coding standards like MPEG-2 can realize temporal scalability using different types of frames, i.e. I frame, P frame, and B frame, wherein I frames only consist of the lowest temporal layer (corresponding to the lowest frame rate), P frames only consist of the middle temporal layer, and B frames only consist of the highest temporal layer. Decoding of one temporal layer requires the presence of the temporal layer itself and all the lower temporal layers, but not the presence of any higher layer. H.264/AVC and its scalable extension SVC (Scalable Video Coding) support advanced temporal scalability thanks to the flexible inter prediction scheme and flexible reference frame management scheme. It should be noted that in H.264/AVC and its extensions, B frames themselves may be reference frames, which may be used by other frames for inter prediction, therefore the discarding of which may lead to problems in decoding of the rest of the bitstream.
The RS MAY transmit the unicast burst as follows. In the beginning of the unicast burst, the RS discards one or more of the highest temporal layers and transmits only the remaining lower temporal layers. After a certain point, all temporal layers are transmitted. This would speed up the acquisition of the multicast session under the same unicast burst rate, at the cost of lower initial frame rate. At the same time, if the playback of the initial stream with lower frame rate is sped up, inter-user playback delay can be reduced.

5. Message Extensions

This section defines the extensions to RAMS-R and RAMS-I messages for inter-user playback delay reduction.

5.1. Extension to RAMS-R

TBD.

5.2. Extension to RAMS-I

TBD.

6. Security Considerations

TBD.

7. IANA Considerations

TBD.

8. Acknowledgements

TBD.

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9. References

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