Playing Rate Adaptation for Increased QoE over Streaming Services

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Abstract: - Rate adaptation is a general problem in streaming services where sending and receiving sides should compromise on a common rate, e.g. to minimize the rebuffering failures, also called start-stop failures here. Currently, start-stop failure has the dominant effect on QoE (quality of experience) since the other types of streaming errors, such as blurring and color distortion of media frames, can be significantly avoided with the protection of the streaming channels against packet losses, e.g. using Forward Error Correction or TCP/IP protocol. In this paper, the term Playing Rate Adaptation is introduced to further increase the rate adaptation capability of the streaming services by introducing a synthetic failure type, controlled fluctuations in playing rate, which in turns can reduce or eliminate the start-stop failures. Playing some video segments in negligible reduced rate is argued to provide better QoE than the start-stop failures that otherwise would be possible. As a case study, the streaming service is considered to be progressive downloading type over the 3GPP's MBMS (multimedia broadcast multicast service) as the underlying network and as worst case scenario the playing rate adaptation is applied to all frames of 2 minutes video. In order to prove the accuracy of the proposed model, objective and subjective study using DSIS (double stimulus impairment scale) and DSCS (double stimulus comparison scale) are provided. The results show that the playing rate adaptation even in worst case scenario increases the user satisfaction from the service.

Key-Words: QoE, Streaming, Playing Rate Adaptation, MBMS, Progressive Download

1 Introduction

Rate adaptation is a challenging problem in streaming services where the delivery of time critic media data over limited networks and the differences in sending and receiving device profiles, such as the processing and memory capacities, all should be considered. Particularly, for the broadcast networks or unidirectional delivery platforms the problem gets harder to be solved due to having no feedback channel to the sender side. Thus, many unwanted conditions, such as the packet losses, delays, and bandwidth limitations, may occur in these service platforms. Quality of Service (QoS) is a way of classification that manages how the unwanted conditions are controlled and mapped to the service quality. QoS could be considered as a compromise of the both sending and receiving sides on a common service quality where the rate adaptation plays an important role for regulating the fluctuations in quality. However, QoS could not reflect how the end-user experience is. At this point, the QoE describes the achieved QoS and the end-user satisfaction with the service [1]. The interaction between the QoS and QoE is shown as layers in Fig. 1.

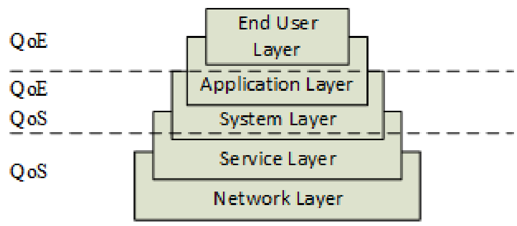

Fig. 1. Layered approach for QoS and QoE.

At the network layer the QoS parameters are the communication requirements, such as bandwidth, delay, jitter, loss, and reliability. System level QoS is related to operating system and processing/buffering capability of the end-user equipment. Application layer QoS are media related parameters such as media player, frame size, frame rate, media encodings as well as the player buffering method. End-user layer reflects the whole interactions between the service and end-users. At this level, QoE describes purely the degree of user satisfaction from the service as a whole. Currently, the QoE is getting an overlay over the other layers due to its increasing importance and popularity in new generation service platforms. QoE aware traffic management [2], QoE aware service cost, QoE aware operating system [3], the QoE aware error controls [4], and QoE aware players[1] are examples of the convergence between QoS and QoE.

There are many solutions for rate adaptation problem depending on the target platform considered. One major classification could be based on whether the approaches are QoE aware [5,6,7], QoS aware [12-17] or Non-Quality aware[19,20]. Majority of the related works in literature could be considered as Non-Quality aware in that they just aim to manage the rate-control without considering any QoS issue. Usually these works describes best-effort services that attempt to maximize the average quality. The QoS aware approaches take one or more of the QoS parameters into account in decision process of the rate-control algorithms. Thus, they attempt to preserve the QoS level despite of the fluctuations in network conditions. Recently, QoE aware rate adaption approaches are considered for various service platforms[8,9,10]. These approaches attempt to preserve the QoE level rather than the QoS level. Thus, they usually need a QoE measurement method by which the rate adaption might be triggered. The QoE aware rate adaption was first introduced for a traffic optimization, in which a utility function capturing the user satisfaction as a function of data rate is applied [11]. Later approaches further improved the utility function to measure the QoE for various service platforms [5,6]. For example, [5] provides the impact of different rate adaptation techniques on the user perceived video quality where the selection of the rate adaptation scheme is based on the QoE based utility function.

Another classification of the rate adaptation approaches in literature could be based on whether the adaptation is triggered by sender-side [17,18], receiver-side[19,20] or network/content-centric approaches[21,22]. Sender-driven approaches control the transmission rate depending on the feedback from receiver or feedback from network nodes that keep track of the available bandwidth in receiver-side. Receiver-driven approaches usually no need a reverse channel to the sender instead they drop/increase the rate in some way with the cost of losing/gaining some kind of quality. For example, in [23] the video streaming is deployed over a few channels where the primary channel provides an average quality and the others provide incremental improvement in video quality. Depending on the network conditions the receivers can choose to join or leave the channels. So the common property of the rate adaptation algorithms is their adaptation into some quality where the quality is any matter of quality degrading failures, simply called quality failures, in network, service, system, application, or end-user layers (see Fig. 1). At this point, what makes the work different from those in literature is the introduction of a new quality metric that is degraded by the synthetic fluctuations in playing rate. Synthetic means the fluctuation is created by the algorithm itself, e.g. according to the stream-time media content, in order to reduce the future the start-stop failures. Actually, the playing rate adaptation requires finding suitable video segments in streaming where applying the playing rate adaptation has minimal effect on the perceived quality. For example, some segments may contain exciting scenes with fast actions while some others may contain stable scenes with slow actions and the user perceived quality depend on the stream-time content. Finding the suitable segments for playing rate adaptation is out of scope of the study. However, as a worst-case study, playing rate adaptation is considered to span all frames of the 2 minutes video in that worst-case test allow us to make some generalization.

In this paper, we provide an application-layer overlay to the existing rate adaptation approaches by introducing playing rate adaptation that is not considered yet in the literature. The proposed adaptation provides a trade-off between the start-stop failures and the disturbance of the negligible drops in playing rate where the resulting fluctuations in playing rate can be better hided from users than that of the start-stop failures. Thus, in this paper, it is claimed and proved that reducing the playing rates of some consecutive video frames (segments) in a way that the user perception is minimally affected decreases the future start-stop failures in advance and hence increases the QoE of users from the service. The proposed rate adaptation is not an alternative of the existing ones. It simply attempts to further increase the subjective quality when the start-stop failures are possible. Thus, we studied two types of quality failures, the start-stop failures versus the fluctuations in playing rate, with their effects on QoE. In the case study, the progressive download service[24,25,26], which is a streaming technology using “play while download” approach, over MBMS network is considered where the channels are assumed to be packet-lossy and protected using FEC. The proposed rate adaptation is equally applicable to all types of streaming services regardless of the underlying networks. In order to prove the accuracy of the proposed model, we provide some subjective study using DSIS (double stimulus impairment scale) and DSCS (double stimulus comparison scale) as well as some objective study computing the rebuffering lengths. The results show that the playing rate adaptation even in worst case scenario increases the user satisfaction from the service.

The paper is organized as follows; second section provides the system model and formulization of the problem. Third section demonstrates the experimental results over various MBMS link conditions. Finally, conclusion and future directions are given.

2 System Model

The progressive download client is based on the model introduced in [1], which is shown in Fig. 2.

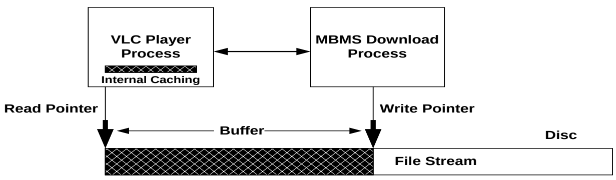

Fig. 2. MBMS progressive download client

In the model, MBMS download client and MBMS download server introduced in [25] are used. The progressive download client contains both the downloading process and media player process. Thus, to emulate progressive download clients, the prototype in [25] is integrated with the VLC player's open source codes [27]. The download services are fully protected against loss errors using FEC overheads, which are given in the experimental results section.

2.1 Buffering Model

A QoE aware player introduced in [1] is considered in the model where the maximum initial delay (MID) that the end-user can tolerate and a minimal blocking length (MBL) that the users prefer are subjective parameters of the buffering model. They in some sense personalize the buffering behaviors and create a value for the user expectation. The MID provides the user tolerance against the initial delay. This parameter also shows the user's tradeoff between the initial disturbance and the intermediate disturbances. For example, users who prefer higher initial delay know in advance or expect that they will see smaller number of intermediate rebuffering events.

A receiver doing a reliable progressive download is shown in Fig. 3(*a*) and 3(*b*) for buffering and no-buffering cases respectively. In the figures, the receiver is assumed to have a constant media play rate, shown in thick lines, just for the visualization purpose, and the receiving rate, shown in thin lines, is assumed to be less than the media play rate in order to formulize the problem. The player starts playing media only after the time T_d, which is the initial delay for the progressive download. Formally, buffering occurs at time T_i when the downloaded media size F_i is consumed by the player, or similarly the read pointer is reached to the write pointer in Fig. 2. The bufferings cause the end-user experience to be divided into phases, shown in Fig. 4.

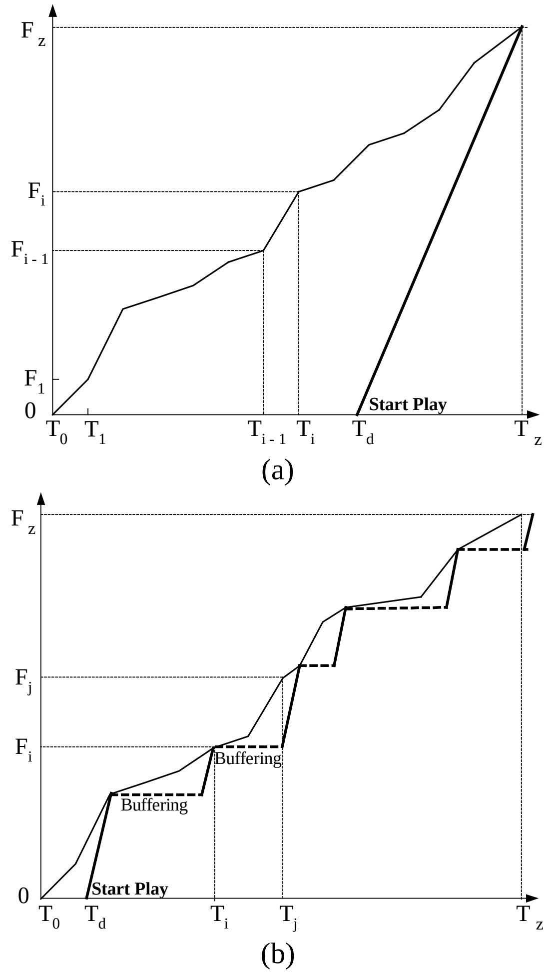

Fig. 3. Examples of progressive download with a) no-buffering case, b) buffering cases.

The expected initial delay predicted at Equation (1) is the required waiting time to have no intermediate disturbance. However, in any case the initial delay will not take longer than the MID value. Initially the player starts with the buffering state for the initial delay computed at Equation (2).

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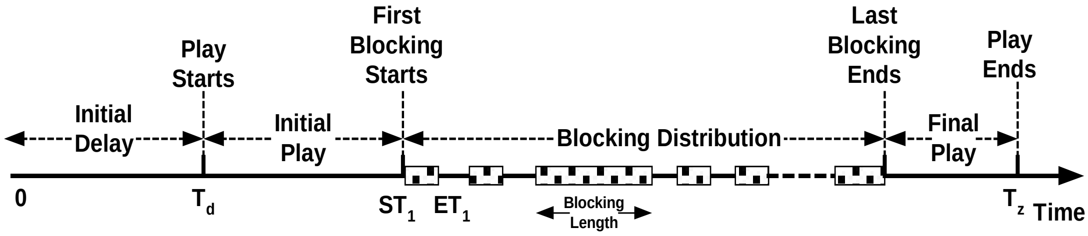

Fig. 4. Possible phases during streaming over the progressive download

The player decides its state later on using state algorithm below. Once the player switches to the playing state, it will not compute the initial delay any more. The player switches to the buffering state when it reaches to the eof of available data where the difference in downloaded data and played data becomes zero, meaning *Diffk=0* (see Equation (3)). The further details of the player can be found in [1].

$$ExpectedInitialDelay_k \approx MediaSize * \left(\frac{1}{ExpectedDownloadRate_k} - \frac{1}{ExpectedPlayingRate_k} \right) \quad (1)$$

$$InitialDelay_k = \min(MID, ExpectedInitialDelay_k) \quad (2)$$

$$Diff_k = DownloadSize_k - (PlayedSize_k + CacheSize) \quad (3)$$

State Algorithm:

If $T_k < InitialDelay_k$
  then $State_k = BufferingTill(InitialDelay_k)$
ElseIf Player is at EOF
  then $State_k = BufferingFor(MBL)$
ElseIf $(State_{k-1} = Buffering)$ *and* $(Diff_k > MBL * MediaSize \,/\, MediaLength)$
  then $State_k = Playing$
ElseIf $(State_{k-1} = \text{Buffering})$
  then $State_k = BufferingFor(MBL)$

2.2 Problem Formulation for Objective Study

The aim of the objective test is to compute the reduction in the total delay due to the playing rate adaptation. The total delay involves the initial or later delays (critical delay) in the critical region. The critical region is defined as the time interval between the first and last blocking shown as blocking distribution in Fig. 4. Let's consider two identical streaming services, meaning the same network, link and buffering conditions, identified by the same configuration parameters in Table 1. One of the services has the playing rate adaptation enabled and the other is not. Let $CriticalDelay^{PRA}$ and $InitialDelay^{PRA}$ indicate the critical delay and the initial delay for the playing rate adaptation enabled service respectively. Similarly, *CriticalDelay* and *InitialDelay* is for the playing rate adaptation disabled service. One objective way to find how much blocking-time is prevented by the playing rate adaptation enabled service with regards to the adaptation disabled service for a particular configuration is given at Equation (4) and Equation (5).

$$TimeGain(\%) = \frac{InitialDelay + CriticalDelay - (InitialDelay^{PRA} + CriticalDelay^{PRA})}{InitialDelay + CriticalDelay} \quad (4)$$

$$TimeGain(Sec) = InitialDelay + CriticalDelay - (InitialDelay^{PRA} + CriticalDelay^{PRA}) \quad (5)$$

2.3 Problem Formulation for Subjective Study

The aim of the subjective test is to experience the quality difference from the user perspective between the two identical streaming services, identified by the same configuration parameters in Table 1, where one stream is the playing rate adaptation enabled and the other is not. Two subjective tests, namely DSCS and DSIS, are considered over 20 subjects. With DSCS, subjects are presented with a pair of video streams. The order within the pair is randomized. Subjects directly rate the quality difference of the second stream from the first one on a seven point scale, *-3(much worse), -2(worse), -1 (slightly worse), 0 (same), 1 (slightly better), 2 (better), 3 (much better).* With DSIS, subjects are presented with a pair of video streams where the first one is the reference, and the subjects are informed about it, second one is impaired. After their playback, subjects are asked to give their opinion using five impairment scales, *5* *(imperceptible), 4 (perceptible, but not annoying), 3 (slightly annoying), 2 (annoying), 1 (very annoying).*

Let $ComparisonScore_i$ is the score of the *i.* subject among *S=20* subjects for the pair of streaming service over the same network, link and buffering conditions, identified by the same configuration parameters in Table 1. The comparison scores of the subjects for a particular configuration parameter is averaged at Eq. 6.

$$ComparisonScore = \frac{\sum_{i=1}^{S} ComparisonScore_i}{S} \qquad (6)$$

3 Experimental Results

The parameters used in the experiments are given in Table 1. Experiments are emulated on the single computer where MBMS download server and MBMS progressive download clients running together constitute the system. Each configuration involves buffering configuration, such as MBL and MID, link and bandwidth configuration considered for MBMS. The optimum values of the FEC parameters to overcome the packet losses are given at the first row of each table for the link layer losses indicated. For playing rate adaptation, the drop in playing rate is considered to be *5%*, which is empirically found. For objective tests, each streaming test for a particular configuration is repeated 10 times and the results are averaged.

Table 1. Configuration parameters of the experiments.

Parameter	Experiment Set
Media Source	138 sec. of the Ice Age 3 trailer (9.46 MB)
Frame Resolution	480 x 254
Media Encoding	AAC+ Stereo 44100 Hz / H.264 AVC codec, 25 fps
FEC Codec	Reed S. Encoding-ID="129" , Instance-ID="0"
SDU Block Size	{800,1000} Byte
Symbol Length	{SDU –48}Byte
SB Size	{200, 214}Symbol
IP Packet Size	{SDU} Byte
PDU Block Size (RLC Block Size)	1280 Byte
PDU (RLC Link Layer) Loss Rate	{0,1,5} %
Transmission Rate	612 Kbps
VLC Player Caching Time	0.750 sec.
Minimum Blocking Length	{2,4,6} sec.
Minimum Initial Delay	{5, 10} sec.
Drop in Playing Rate for playing rate adaptation	5 %
Critical Delay, Blocking Frequency, Time Gain, Comparison Score	Target

The results of the objective tests for *1%* and *5%* link layer losses are given in Table 2, 3 and 4. For subjective test, 20 users are considered. Each user is allowed to experience the same streaming service for a particular configuration with the playing rate adaption disabled versus enabled. The comparison scores of the subjects are then averaged. The results of the subjective tests for 1% and 5% link layer losses are given in Fig. 5 and Fig. 6 where DSCS method is used to assess the users' QoE for comparison of the two services. The subjective test also includes DSIS study for no-loss case where no blockings occurs at all and users only assess the impairment imposed by the playing rate adaptation itself. The results for DSIS study is given in Fig. 7.

Table 2. Objective results with the 1% pdu losses.

SDU Size 1000 B, SB Size 200 Symbols, FEC 8%					
Playing Rate Adaptation Enabled /Disabled	MID (sec)	MBL (sec)	Average Download Rate (Kbps)	Critical Delay (sec)	Blocking Freq.
Disabled	5	2	516	**10**	**4**
Disabled	5	4	516	**8**	**2**
Disabled	5	6	516	**12**	**2**
Disabled	10	2	516	**6**	**2**
Disabled	10	4	516	**4**	**2**
Disabled	10	6	516	**6**	**1**
Enabled	for all	for all	516	**0**	**0**

Table 3. Objective results with the 5% pdu losses

SDU Size 800 B, SB Size 214 Symbols, FEC 19%					
Playing Rate Adaptation Enabled /Disabled	MID (sec)	MBL (sec)	Average Download Rate (Kbps)	Critical Delay (sec)	Blocking Freq.
Disabled	5	2	448	**28**	**10**
Disabled	5	4	448	**32**	**8**
Disabled	5	6	448	**30**	**5**
Disabled	10	2	448	**24**	**8**
Disabled	10	4	448	**24**	**6**
Disabled	10	6	448	**24**	**4**
Enabled	5	2	448	**22**	**8**
Enabled	5	4	448	**24**	**6**
Enabled	5	6	448	**24**	**4**
Enabled	10	2	448	**18**	**7**
Enabled	10	4	448	**20**	**5**
Enabled	10	6	448	**18**	**3**

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The playing rate adaptation for low loss rate (*1%*) produce better objective results where 10 sec. blocking-time is prevented on the average for the initial delay of 5 sec, than that of the higher loss rate (*5%*). The main reason is the users are able to distinguish the two services having small number of start-stop failures. When the number of blockings or start-stop failures increases the ability of the users to differentiate the service quality decreases. That is, the higher packet losses put additional fluctuations and dominate the effect of the playing rate fluctuations. For small initial delays (*MID =5*), a better time gain is achieved than that of the higher initial delay (*MID=10*).

Table 4. Time gain from playing rate adaptation

MID	MBL	%1 PDU Loss		%5 PDU Loss	
		Time Gain (Sec)	Time Gain (%)	Time Gain (Sec)	Time Gain (%)
5	2	**10**	**67**	**6**	**18**
5	4	**8**	**62**	**8**	**22**
5	6	**12**	**71**	**6**	**17**
10	2	**6**	**38**	**6**	**18**
10	4	**4**	**29**	**4**	**12**
10	6	**6**	**38**	**6**	**18**

The objective results show that the start-stop failures can be avoided significantly with the cost of negligible drop rate in playing rate where the fluctuations in the playing rate is almost hided from the users' perception(see Fig. 5 and 6).

The playing rate adaptation can be considered as a time-saving method, with the cost of negligible drop in temporal quality of the streaming, and the saved time can be used in later time, e.g. to reduce the start-stop failures. So small initial delay means that the playing rate adaptation, or the time-savings, is started early during streaming, which gains more time than that of the higher initial delay. *MID* and *MBL* pair is just a parameter of the QoE aware player. As seen in tables, their various combinations create different characteristics in fluctuations of the start-stop failures. From the experiments, one can easily discover that higher *MBL* reduces the number of the start-stop failures with the cost of increased blocking lengths. However, its effect on users' perception can vary since the users' experienced quality depends on highly their internal states, such as expectations and psyche.

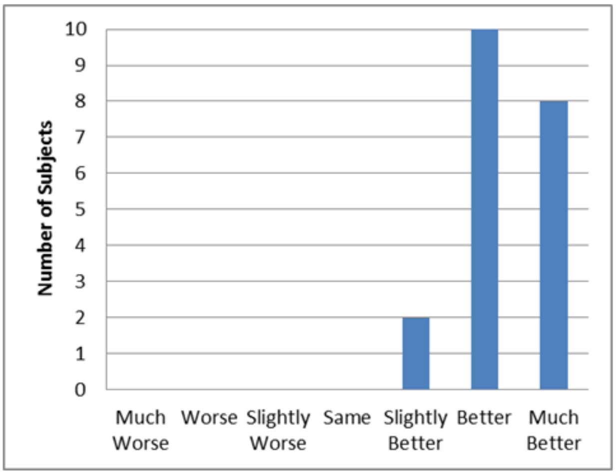

Fig. 5. Comparison scores according to DSCS for *1%* pdu losses

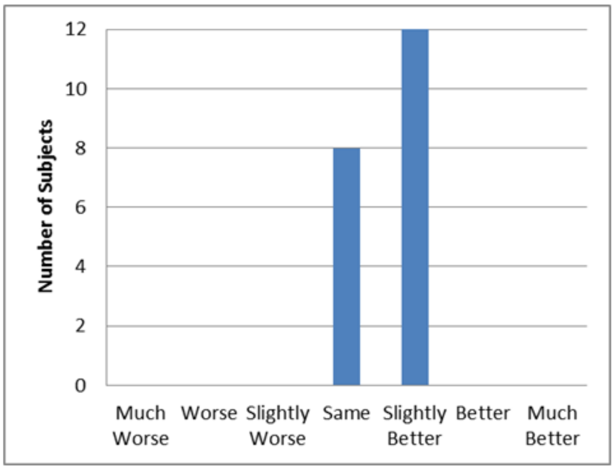

Fig. 6. Comparison scores according to DSCS for *5%* pdu losses

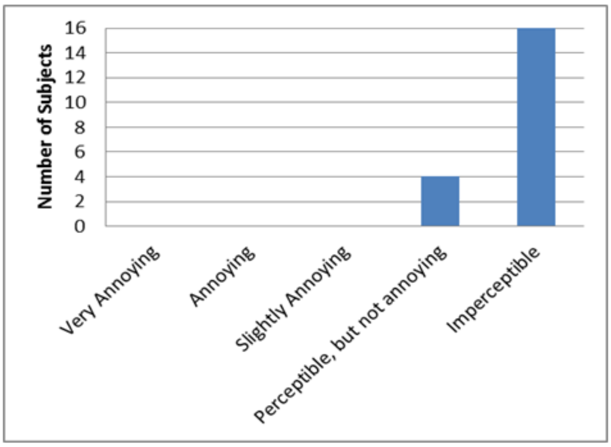

Fig. 7. Comparison scores according to DSIS for *0%* pdu losses

This study does not aim to discover the overall effect of the *MBL* and *MID* pair on subjective quality. So a single *MBL* and *MID* pair (*MBL=2, MID=5*) is chosen for the subjective tests given in

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Figure 5 and 6 in order to show the effect of the playing rate adaptation on subjective quality. The results are important in that we can make some generalization about the effect of the playing rate adaptation on perceived quality.

Generally, the playing rate adaptation can be completely hided from the users' perception when suitable adaptation rate is given. Also, the objective results conform to the subjective results where a better QoS is achieved with the low loss rate (*1%*). Fig. 5 shows that all subjects perceived a better quality, (*Comparison Score = Better*), with the service having the playing rate adaptation. For higher loss rates, the higher number of blockings dominates and reduces the overall quality perceived by the users. However, as seen in Fig. 6, even in higher loss rate (*5%*), users are still able to perceive slightly better QoS, (*Comparison Score = Slightly Better*), with the service having the playing rate adaptation. In order to prove that the playing rate adaptation is completely hided from the users' perception, an additional comparison test according to the DSIS method is done. The aim of the test is to remove the dominant factor caused by the losses and discover the standalone effect of the adaptation.

The results in Fig. 7 shows that majority of the subjects are unaware of the playing rate adaptation with the *Comparison Scores = Imperceptible*. With the results, the streaming service can be better perceived by the users by enabling the playing rate where suitable in the stream-time content. Although this study only considers a worst-case test, meaning adaptation applied to all video frames of 2 minutes video, the results provide us to make following generalization; applying the playing rate adaptation to some selected segments will surely provide better achievement.

4 Conclusion

Playing rate adaption is proposed to increase the QoE of the streaming services. The proposed adaptation is an application layer overlay of the existing rate adaptation methods. So, it can be applied to any type of streaming services regardless of the underlying network. The playing rate adaption introduces new quality failure, namely synthetic fluctuations in playing rate, which degrades the temporal characteristics of the streaming video. Thus, the proposed method aims to convert one failure type to another one where the effect on QoE is better with a tradeoff between the two failure types. The objective and subjective results show that playing rate adaptation provides better QoE achievements even in worst-case tests. The work also provides the case study that shows the objective/subjective analysis of the model for various MBMS link conditions. The work opens new research door in the area of QoE over streaming. Further study is needed to discover the overall aspect of the playing rate adaptation, such as developing methods to find the suitable video segments for the playing rate adaption.

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