QoE Aware Player using Playing Rate Adaptation for Progressive Download Services over Broadcast Networks

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Abstract: In this paper, a QoE (quality of experience) aware player is introduced that changes the temporal characteristics of streaming according to subjective parameters to increase the QoE. With the proposed player, two contributions are provided. First, the capability of rate adaptation methods is further enhanced by the playing rate adaptation where playing some video segments in negligible reduced rate is argued to provide better QoE than the ones degraded by start-stop failures that otherwise would be possible. Second, the temporal distribution of the start-stop failures can be modelled according to the user preference so that user‘s expectancy is better satisfied. Currently, start-stop failure has the dominant effect on QoE since the other types of streaming errors, such as color distortions, can be significantly avoided with the protection of the streaming channels against packet losses, e.g. using Forward Error Correction. With the channel protection, the overall network errors are projected onto behaviors of the buffering model. The buffering should be described by well-defined states and expected behaviors in that expected behaviors, from the user expectation point of view, are better than the random ones. As a case study, the streaming service is considered to be progressive downloading over 3GPP’s MBMS (multimedia broadcast multicast service) as the underlying network. 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 QoE aware player increases the user satisfaction from the service.

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

1 Introduction

QoE aware streaming is not a new concept and some content providers have already started the QoE aware streaming services, in some degree e.g. using content distribution networks (CDNs) and adaptive bitrate (ABR) streaming where a video player adjusts resolutions based on end-to-end network conditions. Quality of service (QoS) is a way of classification that manages how the network conditions are controlled and mapped to the service quality. QoS could also 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. Currently, rate adaptation is the primary mechanism for the sending and receiving sides to be compromised on a common rate, e.g. to minimize the start-stop failures. Particularly, for the broadcast delivery platforms the rate adaptation is a challenging problem due to having no feedback channel to the sender side. Thus, the fluctuations in quality, due to packet losses and delays, expose a great deal of work in these service platforms. QoS could not reflect how the end-user perceives the fluctuations in quality. At this point, the QoE describes the end-user satisfaction with the service [1]. The interaction between the QoS and QoE is shown in Fig. 1

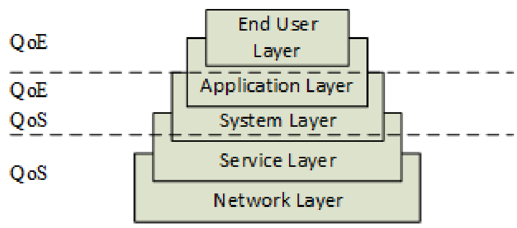


Fig. 1. Layered approach for QoS and QoE.

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 and its buffering method, frame size, frame rate, and media encodings. End-user layer reflects purely the degree of user satisfaction from the service as a whole. At this level, subjective parameters may be used to define the overall interaction between the end-users and the service application. Currently, the QoE is getting an overlay over the other layers due to its increasing importance and popularity in new generation service platforms[2,3,4,5,6]. QoS / QoE aware traffic management [7, 8], QoE aware streaming [9], and quality aware power control[10,11] are examples of the convergence between QoS and QoE.

In literature, quality aware streaming services are studied as closely related to the underlying rate adaptation methods. One major classification of these works could be based on whether the approaches are QoE aware [12-15], QoS aware [20-25] or Non-Quality aware [26]. 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 changes in the network conditions. Recently, QoE aware streaming approaches are considered for various service platforms [16,17,18]. These approaches usually incorporate various QoE metrics that are used to match the end-user's perceived quality. Thus, they usually need a QoE measurement method by which the behavior of the rate adaption can be changed. 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 [19]. Later approaches further improved the utility function to measure the QoE for various service platforms [13, 14]. 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).

All the aforementioned works only considers the spatial tolerance of the human visual systems. Thus, they play with a spatial quality such as adapting to a particular resolution or frame quality with a support from the sender-driven [25], receiver-driven [26], or network / content centric [27] approaches. However, recent studies [28] prove the importance of the temporal tolerance of the human visual systems in streaming applications. Thus, small fluctuations in playing rate can be tolerated, in some degree, by the human-visual system and this property can be used to reduce the start-stop failures by having the player reduce the playing rate wherever possible, e.g. according to the stream-time media content. That is, the playing rate can be changed dynamically by the player itself so as to minimize the start-stop failures without any disturbance on the perceived quality. This approach is called as playing rate adaptation. Ideally, 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.

In this paper, we provide a QoE aware player that considers both the playing rate adaptation to reduce the start-stop failures, and a buffering method to distribute the start-stop failures according to user-defined subjective parameters. The proposed player dynamically switches between two temporal quality levels using the playing rate adaptation where one is temporally impaired and the other one is normal. The switching between the quality levels could be as much complex as considering the stream-time media content and its effect on users' psyche. However, as a worst-case study, all the video frames are impaired by the proposed adaptation in that worst-case test allow us to make some generalization. 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. In the case study, the progressive download service [30,31,32], 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. 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 using the state of the arts techniques. The results show that the proposed player even in worst case scenario increases the user satisfaction from the services.

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 system model is shown in Fig. 2 where the MBMS network is simulated according to MBMS link conditions as adapted in [1].

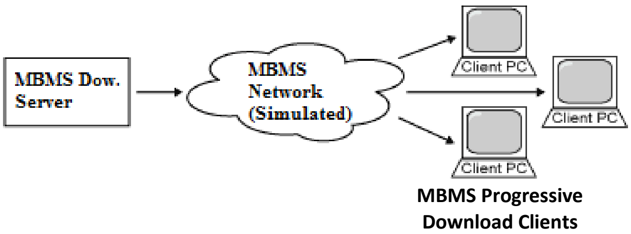


Fig. 2. System Model

In the model, MBMS download client and MBMS download server introduced in [31] are used. The progressive download client is modelled in Fig.3, which contains both the downloading process and media player process. To emulate progressive download clients, the prototype in [31] is integrated with the VLC player's open source codes [33]. The download services are fully protected against loss errors using FEC overheads, which are given in the experimental results section.

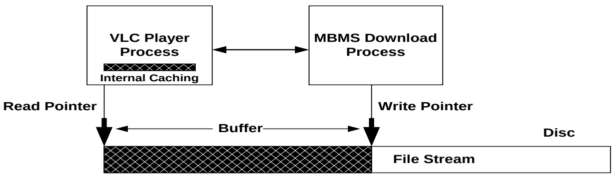
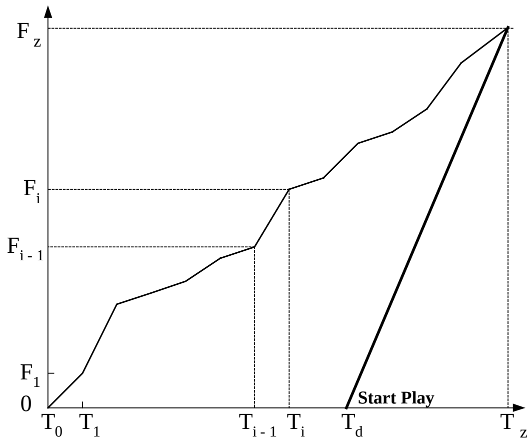


Fig. 3. MBMS progressive download client

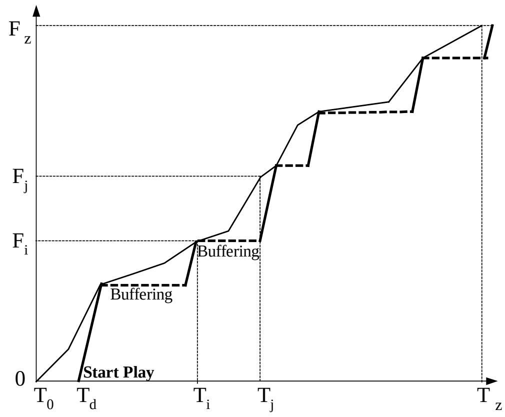
2.1 Buffering Model

The proposed QoE aware player considers the maximum initial delay (MID) that the end-user can tolerate and the minimal blocking length (MBL) that the users prefer as 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 start-stop failures.

A receiver doing a reliable progressive download is shown in Fig. 4(*a*) and 4(*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. 3. Thus, the rebuffering events cause the end-user experience to be divided into phases, shown in Fig. 5. The expected initial delay predicted at Equation (1) is the required waiting time to have no intermediate disturbance where *Expected Playing Rate* and *Expected Downloading Rate* are simply considered as the average playing and downloading rates respectively. However, in any case the initial delay will not take longer than the MID value.



(a)



(b)

Fig. 4. Examples of progressive download with a) no-buffering case, b) buffering cases.

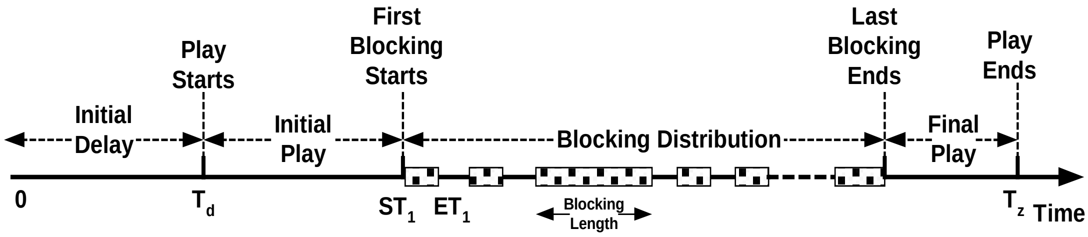


Fig. 5. Possible phases during streaming over the progressive download

Initially the player starts with the buffering state for the initial delay computed at Equation (2). 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)).

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

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

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

State Algorithm:

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

2.2 Playing Rate Adaptation Model

A simple model is used for playing rate adaptation where only two quality-levels are considered: impaired versus un-impaired. The impaired level has temporal quality degradation as a result of the negligible drop in playing rate where the degree of the drop in playing rate is a user defined parameter, and reflects the users' tolerance against slowness of the video motion. The player dynamically switches between the quality levels using a switching algorithm that decides when and which quality level to change to. However, for worst-case study no switching is used, meaning the playing rate adaption is either enabled or disabled all the time during streaming service.

2.3 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 and objectively measure how the end-user experience is using the recently proposed QoE measurement methods [1]. 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. 5. 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). QoE scores are calculated objectively using Equation (6) and Equation (7), which are proposed in [1].

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

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

$$QoE_1 = \left(1 - \frac{InitialDelay + CriticalDelay}{TotalPlay} \right) \quad (6)$$

$$QoE_2 = QoE_1 * \left(1 - \frac{1}{2} * \left(\frac{VPL}{EPL} + \frac{VBL}{EBL}\right)\right) \quad (7)$$

where Expected Blocking Length (EBL) and Expected Playing Length (EPL) are the average blocking and playing segments respectively, and VBL and VPL indicates the variances in blocking lengths and variances in playing lengths respectively. The playing and blocking segments are equally emphasized by taking the average variances of the two. The segments' variances are measured with respect to their means (expected segment lengths) so that the expectancy is used as a reference for the variations in segment durations. Further details can be found in [1].

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. 8.

$$ComparisonScore = \frac{\sum_{i=1}^{S} ComparisonScore_i}{S} \quad (8)$$

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, QoE Scores	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 player. 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. 6 and Fig. 7 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. 8.

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

SDU Size 1000 B, SB Size 200 Symbols, FEC 8%						
PRA Enabled /Disabled	MID (sec)	MBL (sec)	Critical Delay (sec)	Blocking Freq.	QoE_1 (%)	QoE_2 (%)
Disabled	5	2	**10**	**4**	**89**	**82**
Disabled	5	4	**8**	**2**	**91**	**91**
Disabled	5	6	**12**	**2**	**88**	**88**
Disabled	10	2	**6**	**2**	**88**	**81**
Disabled	10	4	**4**	**2**	**90**	**90**
Disabled	10	6	**6**	**1**	**88**	**88**
Enabled	all	all	**0**	**0**	**100**	**100**

Table 3. Objective results with the 5% pdu losses

SDU Size 800 B, SB Size 214 Symbols, FEC 19%						
PRA Enabled /Disabled	MID (sec)	MBL (sec)	Critical Delay (sec)	Blocking Freq.	QoE_1 (%)	QoE_2 (%)
Disabled	5	2	**28**	**10**	**76**	**61**
Disabled	5	4	**32**	**8**	**73**	**64**
Disabled	5	6	**30**	**5**	**75**	**62**
Disabled	10	2	**24**	**8**	**75**	**61**
Disabled	10	4	**24**	**6**	**75**	**67**
Disabled	10	6	**24**	**4**	**75**	**68**
Enabled	5	2	**22**	**8**	**80**	**63**
Enabled	5	4	**24**	**6**	**79**	**70**
Enabled	5	6	**24**	**4**	**79**	**72**
Enabled	10	2	**18**	**7**	**80**	**64**
Enabled	10	4	**20**	**5**	**78**	**70**
Enabled	10	6	**18**	**3**	**80**	**71**

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*).

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. 6 and 7).

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**

QoE scores in Table 2-3 also show that the proposed player achieves a better user satisfaction with the playing rate adaption enabled services. Objective and subjective results well complies with each other. 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.

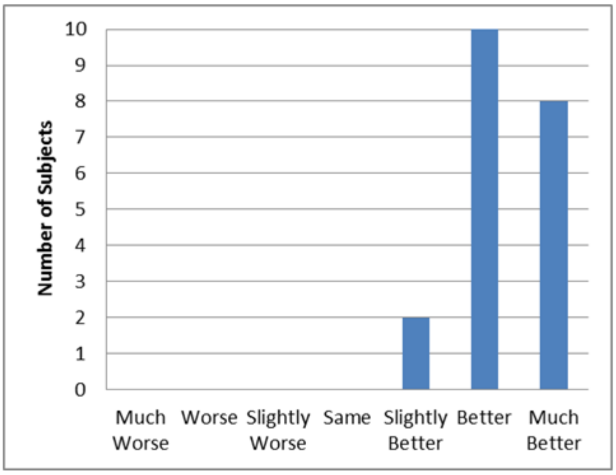


Fig. 6. Comparison scores according to DSCS for *1%* pdu losses

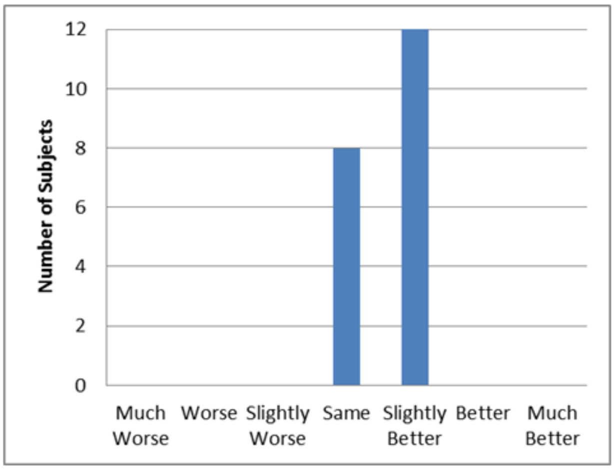


Fig. 7. Comparison scores according to DSCS for *5%* pdu losses

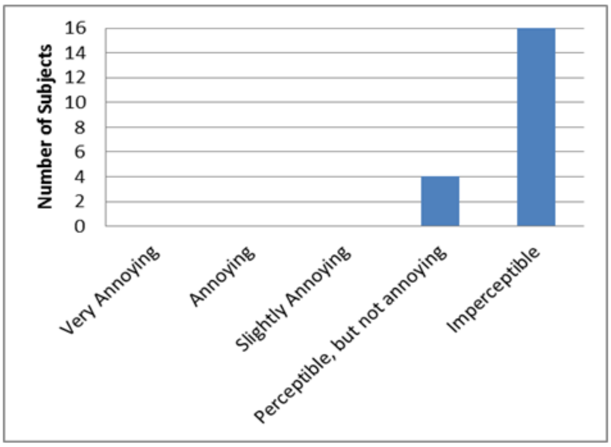


Fig. 8. Comparison scores according to DSIS for *0%* pdu losses

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 Figure 6 and 7 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. 6 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. 7, 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. 8 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

A QoE aware player is proposed for progressive download type of streaming services. The proposed player uses playing rate adaptation to reduce the start-stop failures, and a buffering method to distribute the start-stop failures according to subjective parameters so that a better QoE is achieved. The proposed adaption converts one failure type to another one where the effect on QoE is better with the latter. The proposed adaptation is an application layer overlay of the existing rate adaptation methods and can be applied to any type of streaming services regardless of the underlying network. The objective and subjective results show that the proposed player 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 playing rate adaptation based on the stream-time content.

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