Media Playout Techniques for Video Intra-Stream Synchronization: Review and Analysis

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Abstract
Video streaming over best-effort networks is a challenging problem due to network delay and packet loss. To deal with video playout disruptions, client-side data buffering is a public and known technique. Increasing of data buffering inevitably prevents playout interruption. As a result, more memory requirement is needed and playout delay is increased. Thus, Adaptive media playout (AMP) is a client-side technique which reduces the data buffering and avoids buffer outage. The playout adaptation technique takes on the responsibility of temporal reconstruction of the stream, that is, the restoration of its intra-stream synchronization quality. This article surveys the Adaptive Media Playout techniques, aiming to succinctly merge established concepts and schemes have been proposed for the adaptation playout rate, and classify AMP techniques based on playout rate control. In addition, the pros and cons of each technique are briefly discussed. This study also deliberates challenges and issues related to video stream quality, playout rate control, adapt playout rate based on network conditions handling of buffer outage and quality evaluation metrics.
Keywords
About; Adaptive Media Playout; Multimedia Synchronization; Playout Delay; Video streaming

1. INTRODUCTION

Due to advancement in technology, the use of multimedia streaming applications such as mobile TV, video on demand, video conference, and Internet Protocol Television (IPTV) [1, 2] has grown swiftly. Users receive video streams as a continuous flow of data packets. This is why the video stream can be watched instantly without necessarily downloading it. Nevertheless, streaming high-quality video over packet switch network is still uptight with a lot of challenges [3]. This is due to the real-time property of video traffic and as well the intrinsic dynamic property of the packet networks.

As video applications have a stringent deadline of presentation, delay variations in the network may cause significant jitters to network packets reaching the user. This causes the inability of packets meeting deadline, subsequently leading to jerkiness or frozen playback. Furthermore, the dynamic topology change, time-varying wireless channel, and multi-hop relay can together cause major discrepancies in the end-to-end throughput and delay due to the widespread adoption of various dynamic networks such as notably Peer-to-Peer (P2P) networks and mobile ad hoc networks [4]. Consequently, the need to resolve these challenges and provide users with stable high-quality video is essential for future multimedia demands.

One seamless method to handle the network bandwidth degradation is to adapt video stream based on network channel fluctuation [5-7]. Adaptive streaming rate, Server-side technique, is a technique that enables the optimum streaming video viewing experience of a diverse range of network bandwidth. Adaptive streaming is a widely-used solution to avoid playback interruptions and compensate for insufficient bandwidth or momentary congestion. A temporary reduction in bit rate results in less or no video break-up and re-buffering. The rate control mechanism of a codec implementation allows the selection of a target bitrate and tries to meet

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this target within certain constraints, e.g. number of frames, with a certain percentage overshoot.

Another kind of control to handle the network dynamics in packet video streaming is to buffer video packets in the cache at the client side [8, 9]. Basically, client-side data buffering is used to handle playout interruption produced by the network jitter. In order to eradicate the effects of delay jitter, a part of video frames are buffered and video playback is delayed. On the other hand, when the receiver buffer has never been emptied, the playout delay of the video stream is reduced by rejecting new arrived packets. The number of discarded packets indicates the effectiveness of techniques to decrease latency. Most playout schedulers have charted a fitting approach in the directive of the buffering delay, they increase or decrease it in constant amounts that equal the duration of a Media Units (MUs) [4, 8, 10-15]. Neglecting late frames lead to a sharp delay reduction that is equal to the duration of a video frame. These techniques seem very crude, particularly in cases of low-frame-rate streams where video frames have significant duration. These techniques have been mentioned in detail in [16].

While this aforementioned method is successful in eliminating the effect of playout interruptions and network jitter, it has the drawback of increasing memory requirements [17] as well as playout delay [18, 19]. Therefore, in the presence of intensive network jitters, the choice of playback threshold should seek a balance between both the start-up delay and the fluency of the playback. This calls for the need for extensive research to be carried out on issues about buffering fewer data [20, 21] or by proposing ways to achieve a few playout delays [21, 22] at the same time preserving the Quality of Service (QoS).

The Adaptive Media Playout (AMP) method, which is based on regulating the playout frame rate dynamically, allows the client to buffer fewer data, thereby introducing fewer delay and at the same time preventing buffer outage [23, 24]. It gives the client the opportunity to regulate its data use by controlling the playout interval as regards to the condition of the network. The playout interval is increased when the level of the buffer is reduced, and the playout interval is reduced when the level of the buffer is raised. The concept of AMP is that the playout interval variation is less annoying and better tolerated than long delays and playout interruptions [24][1], [2]. AMP practice must be smooth enough to lower the deterioration of the visual quality that is generated by the playout interval variation.

In this paper, an effort is made to present a structured presentation of the proposed AMP techniques by observing different methods of initiating the playout rate control. The proposed AMP techniques are classified into two types of categories. The main objective is to explain the algorithms in detail and figure out the advantage and disadvantage of each technique. The functional assessment of several schemes is accomplished to show their appropriateness for various real world applications.

The rest of the paper is structured as follows: The video streaming background is presented in section (2). Section (3) presents AMP-Based challenges and issues. Adaptive Media Payout (AMP) Technique is presented in (4). The Buffer Threshold-Based AMP Control is exhibited in (4.1). Section (5.2) displays the Buffer Variation-Based Control. The Buffer Hybrid-Based Control is displayed in section (4.3). Evaluation Metrics for AMP Mechanism are presented in (5). Conclusions and future research directions are presented in (6).

2. VIDEO STREAMING BACKGROUND

Video streaming denotes to real-time transmission of stored video to the client. There are two modes to transmit a stored video over IP network, the known download mode and streaming mode [25]. In the download mode, the entire video file is downloaded before it starts to play the video file back. In contrast, the streaming mode concentrates on playing the video content while parts of the content are being received and decoded. Video streaming addresses the problem of transferring video data as a continuous stream. To achieve this, the efficiency and flexibility of a network channel are very important as well as many challenges [26]. Actually, unknown and time-varying bandwidth, a variation of packet delay (Delay Jitter), and packet loss are the fundamental challenges in video streaming. In response to such challenges, a variety of video coding [27-33] and streaming techniques [34-36] have been proposed to provide video streaming services.

There are multiple transmission modes for video content delivery: download, progressive download, streaming and adaptive streaming [37]. Streaming mode automatically starts playout almost immediately with minimal buffering in real-time. In contrast, an adaptive streaming mode is specially designed to adapt dynamic conditions of unmanaged networks [38].

Most of the modern streaming services have been adopted to use HTTP for streaming purposes [39]. HTTP adaptive streaming (HAS) is based on classical HTTP video which adjusts the video quality during the playout according to network conditions [36]. The HAS is improved by combining it with the MPEG [40].

2.1. General Format, Page Layout and Margins

Figure 1 shows a typical architecture of video streaming over an IP network. Raw video data are compressed and saved in storage devices. Upon the client's request, a streaming server retrieves the compressed video data from storage devices. Then, the transmitter control adapts and sends the video bit-streams according to the network bandwidth. In contrast, the receiver first stores the video frames in the receiver buffer for decoding and playing them back. To achieve synchronization between video frames, an intra-stream synchronization mechanism is required.

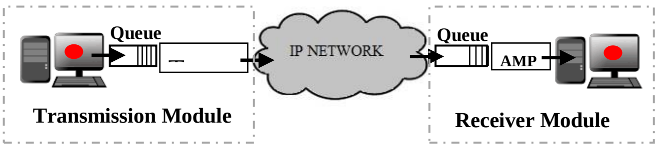

Figure 1: Typical Architecture of Video Streaming Over IP Network

3. AMP-BASED CHALLENGES AND ISSUES

As mentioned earlier the AMP has the ability to reduce the playout delay and playout interruptions. This is done by adjusting the playout rate in the imperceptible speed variation range. A lot of methods have been proposed to evaluate the quality of intra-stream synchronization in order to control the playout rate of AMP process:

- To reduce the chance of buffer underflow, the playout rate is decreased. But this will raise the chance of buffer being overflowed. On the other hand, an increase in the playout rate will decrease the chance of buffer overflow and raise the chance of buffer underflow. These are frustrating issues that need to be systematically dealt with [41].
- The playout delay which is used to alleviate the buffer outage occurrence is the time period between receiving the first frame in the client buffer and starting of playback this frame. The playout delay cannot adjust to the condition of the network. If the playout delay is very short, the underflow cannot be controlled successfully in situations where jitter is high. On the other hand, if the playout delay is very long, additional playout delay will become needless in situations where jitter is small. Therefore, situations, whereby playback starts during the pre-roll period, is another issue in the AMP process [41].
- The buffer fullness threshold is another issue. The playout rate adjustment is triggered according to the buffer fullness threshold. When the buffer fullness threshold is adjusted too high, it will result in needless playout rate alteration even if the buffer fullness is very far from outage leading to diminished visual quality and buffer flux. On the contrary, if the threshold is adjusted very low, playout control will have a short reaction time, therefore, there will be a high probability of buffer outrage occurring before playout control will be able to counterbalance the estimated error of receiving rate [42, 43].
- Another issue is regarding the video playout smoothness. To improve the smoothness of video streaming playout, the playout adjustment rate should be regulated gradually which reduces the perceptible effect of playout speed variation. However, since there is a trade-off between

the quality of video playout and the risk of buffer outage, the playout adjustment speed should be considered carefully [44].

4. ADAPTIVE MEDIA PAYOUT (AMP) TECHNIQUE

AMP being a receiver-based buffer control method is proposed to handle the bit-rate fluctuations of networks that might lead to playout interruption during video streaming [19]. AMP can minimize both the playout delay and chance of occurrence of interruptions by permitting users to easily switch the playout rate in a certain slightly adjusted speed range. This can be done on the client side without involving the server [45].

The main principle of AMP technique is that playout interval differences are bearable and less frustrating to users than the playout interruptions as well as prolonged delays [24, 46][3], [4]. The examination has shown that usually, up to 25% playout speed variations are no obvious but it depends on the content as well. Likewise, usually up to 50% playout speed variations are tolerable in some circumstances.

Based on the literature, the conventional AMP algorithms generally invoke two steps, playout frame rate control and target playout rate estimating. The control of playout frame rate is a control for adjusting the frame rate based on the channel condition. On the other hand, the aimed frame rate estimation is used to adapt the new playout frame rate with regards to buffer fullness and current arrival frame rate. However, AMP technique can be classified based on playout frame rate control into the following three categories.

In summary, the playout frame rate control method for refining the quality of the video stream is categorized into three, shown in Figure 2. Throughout this paper, various playout control techniques from the previous studies will be presented. One main point of differentiation will be whether the techniques address the problems of QoS (buffer underflow, buffer overflow, and playout delay). The AMP techniques implicitly adapt the playout rate to absorb the network jitter; see Table 1 for an overview of AMP techniques.

Table 1. An overview of AMP techniques quantities in terms of buffer underflow, buffer overflow, playout delay and visual quality degradation

Technique	**Algorithm**	**Buffer Underflow**	**Buffer Overflow**	**Playout Delay**	**Visual Quality**
Threshold-Based Technique	Buffer Threshold Control	✓	✓	✓	
	Dynamic Buffer Threshold Control	✓	✓		
	Arrival Process Tracking Algorithm	✓	✓	✓	✓
	Scene-aware AMP	✓			
Variation-Based Technique	Smooth Control of AMP	✓	✓		✓
	Safety Guaranteed Smooth Buffer	✓	✓		✓
	Smooth Playout Control	✓	✓		
Hybrid-Based Technique	Online Buffer Fullness Estimation Based	✓	✓		

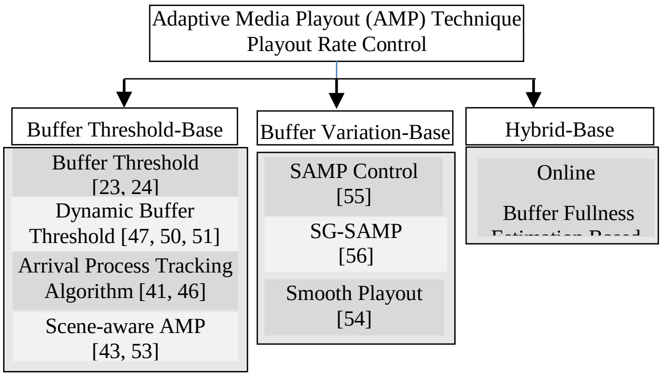

Figure 2: The Classification of Adaptive Media Playout (AMP) Techniques Based on Playout Rate Control

4.1. The Buffer Threshold-Based AMP Control

The Buffer Threshold-Based AMP controller is designed to improve the quality of video in streaming applications [23, 24, 41, 47-53]. The main idea of these techniques is to use buffer level as an indicator of channel quality. In general, two threshold values are set on the

receiver buffer for controlling the playout rate. If the buffer fullness level has reached a certain threshold value, the playout rate controller dynamically regulates the playout rate in accordance with the situation while considering the present buffer frames amount. The receiving buffers are monitored and the playout controller is triggered when the buffer fullness reaches the threshold. Slowing down the frame rate when the buffer fullness is lower than the minimum threshold or conversely speeding up playout frame rate when the buffer reaches the full level or exceeds the high threshold.

These techniques involve two stages, first is to define the activation control threshold by choosing an appropriate buffer fullness threshold value in order to eliminate buffer outrage. Next is the computation of the playout rate while putting into consideration present buffer fullness and predicted threshold.

4.1.1. Buffer Threshold Control

In [23, 24], the AMP technique is proposed to study the ability to reduce the chance of a playout interruption and playout latency. In general, AMP technique is capable of reducing the playout delay and likewise the playout interruption by varying the playout rate smoothly depending on channel conditions. This method adaptively adjusts the frame duration if present buffer fullness is reduced lower than the underflow threshold or surpasses overflow threshold.

However, the buffer threshold based is used for triggering playout rate adjustment. Two buffer fullness thresholds (BH and BL) are predefined. If the buffer fullness level is lower than the predefined threshold (BL), the playout rate ($\mu(n)$) slows down with a constant amount ($s = 1.25$) to reduce the data consumption. On the other hand, if buffer fullness level surpasses the predefined threshold (BH), the playout rate ($\mu(n)$) is increased by a constant amount ($f = 0.75$) to avoid packet loss, as seen in the equation fellow.

$$\mu(n) = \begin{cases} \frac{1}{s.t_F} & n \le B_L \\ \frac{1}{t_F} & B_L < n < B_H \\ \frac{1}{f.t_F} & n \ge B_H \end{cases} \quad (1)$$

where: t_F = normal playout rate
s = factor of slowdown playout rate
f = factor of speed up playout rate

Furthermore, reducing playout delay, which is introduced by pre-roll data, is studied. Essentially, the client starts playout video stream before the buffer fullness reaches the target buffer level, and lets the video data accumulated till it reaches the target level over time by slowing down playout frame rate.

Hence, the disadvantage of this technique is that it is very difficult to select a suitable threshold. This is because the selection of lower threshold leads to higher probability of buffer underflow, likewise, the selection of higher threshold will lead to unwanted frame-separation modifications.

4.1.2. Buffer Threshold Control

Based on the buffer fullness threshold values influence on the visual quality of the playback video stream, the dynamic buffer threshold technique is proposed. According to literature, high threshold value causes a more fluctuation on the playout frame rate due to more unnecessary control activation. On the other side, the low threshold value increases the probability of buffer outage due to insufficient time to react. As a result, an appropriate threshold value should be determined to eliminate the quality degradation. As a result, the dynamic buffer threshold technique [48, 51, 52] is proposed to obtain a good performance of video playback.

A novel scalable video technology is then proposed to address issues of control buffer which can be used to absorb the fluctuations of channel conditions like jitters, delays, and bit rate variations. These papers [48, 52] addressed the problem of the buffer in video streaming, between base-station queue and mobile-station buffer.

The association between buffer fullness and the chances of underflow is discussed. Intuitively, when the data level in the buffer is high, the probability of buffer outage is low and vice versa. According to the relationship between buffer fullness and the probability of buffer outage, an algorithm for AMP control is proposed. It mainly uses the parameter estimation to update the buffer fullness threshold dynamically.

To perform the buffer control, the active buffer fullness threshold is determined based on estimating the quality factor that is chosen in advance. Once the buffer fullness threshold is computed, the playout frame rate is linearly adapted as shown in equation (2).

$$t_D = \frac{B_{SIZE}}{B_L} \times t_F \quad (2)$$

where: B_{SIZE} = buffer size
B_L = low threshold
t_F = normal playout frame rate

4.1.3. Buffer Threshold Control

An enhanced video quality for multimedia streaming services based on an AMP is proposed [41, 47]. The AMP technique is based on buffer fullness. Two threshold values are set for the playout controller to control the playout rate based on a number of buffered frames. Otherwise, the playout frame rate is controlled according to instant packet arrival rate which has been assessed by the Arrival Process Tracking Algorithm (APTA) proposed. The current packet arrival rate (λ_i) is computed based on Moving Average concept, as shown in next equation:

$$\mu(n) = \begin{cases} \dfrac{A_i - A_1}{t_i - t_1} & i < w \\ \dfrac{A_i - A_{i-w+1}}{t_i - t_{i-w+1}} & i \geq w \end{cases} \qquad (3)$$

where w = windows size of the Moving Average

A_i = total number of packet arrivals up to time t_i

The estimated arrival rate is smoothed by Exponential Average approach.

$$\lambda_i'(n) = (1-a).\lambda_i'(n-1) + a.\lambda_i(n) \qquad (4)$$

In this algorithm, the quadratic function is used to adjust the playout rates. In actuality, the playout buffer is divided into three zones, a Safety zone, and two Warning zones. When the buffer fullness level is in the safety zone, playout rate is adjusted to the estimated frame arrival rate. In contrast, when the buffer fullness is in the warning zones (overflow zone, underflow zone), the playout rate is determined by the number of buffered frames. When the buffer level is in the overflow zone, the playout rate $\mu(n)$ is given by next equation:

$$\mu(n) = (1+r_2)t_F - \left(\frac{B_{SIZE} - B_{LVL}}{B_{SIZE} - B_H}\right)^2 .(r_2 - r).t_F \qquad (5)$$

On the other hand, the buffer level in the underflow zone, the playout rate $\mu(n)$ is given by the next equation:

$$\mu(n) = (1+r_1)t_F - \left(\frac{B_{LVL}}{B_L}\right)^2 .(r_1 - r).t_F \qquad (6)$$

where: t_F = normal playout rate

B_{LVL} = current level of buffer fullness

B_H = high threshold

B_H = low threshold

B_{SIZE}= buffer size

r_1, r_2 *and* r = the restricted deviation ratios for playout rates

4.1.5. Scene Aware Smooth Playout Control Algorithm

The central concept of scene aware frame rate regulation technique with dual threshold [43, 54] is to estimate the playout rate of the frames with reduced motion strength more than that with higher motion strength in order to lower the effect on user perception. This process is composed of two phases. The first phase involves controlling the threshold to trigger the process while the second phase is when the playout rate is adjusted. The idea of adjusting the threshold is for increasing the low threshold whenever there is a reduction in the amount of data in the buffer.

In regards to the playout rate adjustment, underflow time is estimated and compared with a threshold. When the estimated underflow time is smaller than a low threshold, the playout rate needs to be reduced. In contrast, when estimated underflow time is between low and high threshold, the buffer will be in a fluctuating state and playout rate should be tuned based on the scene variation.

$$U_t \leq \frac{(B_{LVL} + R_D(t))}{t_F} \tag{7}$$

where: B_{LVL} = current level of buffer fullness
$R_D(t)$= received data during time period (t)
t_F = normal playout rate

According to the Underflow Time Estimation (U_t), the playout frame rate is adjusted. The first step is to dynamically adjust the threshold to activate the algorithm in a timely manner. When there is reduction in the number of data in the buffer, the value of buffer low threshold (B_L) needs to be set higher in order that the process would be triggered faster, and vice versa. The second step is the adjustment of playout rate to reduce the probability of buffer underflow. According to the Underflow Time Estimation (U_t), the playtime frame is adjusted. Figure 4 shows the flowchart of the algorithm.

In [43], the extension of the previous work, the problems of buffer underflow below a VBR channel in multimedia applications are discussed. It considers the choice of estimate for the underflow time in case of changing to a worst case scenario and when the available bandwidth decreases abruptly in the next cycle. Based on the association concerning the data consumption rate and the data arrival rate, the buffer fullness level is estimated. First, if the payout rate is kept as before, the expected size of the buffer fullness $(\dot{B}_{LVL+1})$ is equal as follows:

$$B'_{LVL+1} = B_{LVL} + (\lambda_m S_m \times T_{m-1} - Y_m) \tag{8}$$

where: Y_m = size of Group of Pictures
λ_m= arrival rate of data packet
S_m = average packet size
B_{LVL} = current buffer fullness

Second, if the playout rate is slowed down with factor (1.25), the expected size of the buffer fullness (B''_{LVL+1}) is equal as follows:

$$B''_{LVL+1} = B_{LVL} + (\lambda_m S_m \times 1.25 \times t_F \times G - Y_m) \tag{9}$$

where: G = Length of GoP.

Third, if the playout rate is adjusted to normal rate, the expected size of the buffer fullness (B'''_{LVL+1}) is equal as follows:

$$B'''_{LVL+1} = B_{LVL} + (\lambda_m S_m \times t_F \times G - Y_m) \tag{10}$$

Based on the buffer fullness estimation, the estimated buffer underflow time $(U'_{m+1}, U''_{m+1}, U'''_{m+1})$ is calculated and compared with buffer threshold as shown in next equation:

$$U'_{m+1} = \frac{B'_{LVL+1}}{\mu'_m - A'} \tag{11}$$

where: $\acute{\mu}_m$ = Estimated playout rate
$\acute{A}$ = data arrival rate

$$U''_{m+1} = \frac{B''_{LVL+1}}{\mu'_m - A'} \qquad (12)$$

$$U'''_{m+1} = \frac{B'''_{LVL+1}}{\mu'_m - A'} \qquad (13)$$

4.1.5. Advantages and Disadvantages of the Buffer Threshold-Based AMP Control

Buffer threshold-based methods are sample techniques. Two thresholds (BL and BH) are set according to the buffer fullness. The playout rate slows down when the buffer fullness drops down below the low threshold. On the other hand, it speeds up when the buffer fullness exceeds the high threshold. The main drawbacks are that most previous buffer threshold-based AMP techniques trigger the playout rate adjustment based on the buffer fullness but do not consider the visual quality degradation caused by the playout rate variation. Moreover, it is difficult to select an appropriate threshold. When the buffer threshold is set too high, unnecessary playout rate adjustment even the buffer fullness is far from buffer outage. On the contrary, when the threshold is set too low, the likelihood of buffer outage is high due to a short time for adjustment. Even though some techniques dynamically adjust buffer threshold based on channel quality, the channel quality is difficult to be predicted especially for wireless networks. Furthermore, the motion of video scene is exploited to reduce the buffer outage; this technique cannot be used in high motion streaming such as football games, action movie and so on.

4.2. The Buffer Variation-Based Control Technique

The Buffer Variation-Based AMP control is aimed to keep video playout smooth while adjusting to the channel situation [44, 55, 56]. The main idea in these techniques is to be able to control playout rate according to buffer variation rather than buffer fullness. In general, the technique monitors the client buffer and triggers the playout control when the buffer variation is higher than the predefined variation threshold. According to variance tendency, the system adjusts the playout frame rate; it is slowed down when the variance tendency is negative. On the other hand, the playout frame rate is speeded up when the variance tendency is positive.

4.2.1. Smooth Control of AMP (SAMP) Algorithm

SAMP technique was proposed to tackle the difficulty of the degrading of the visual quality instigated by the difference in playout interlude [44]. The buffer variation is considered instead of the fullness of the buffer level as a condition to activate the playout controller; this is because it reflects the deviancy in the playout amount from the rate that should be normally received.

As Figure 6 shows, the scheme keeps monitoring the fullness and variation of the receiving buffer. When the buffer variation exceeds the predefined variation threshold (π), the scheme triggers for a playout adjustment (PA) in order to estimate a new playout rate. Before adjusting the playout interval, two values need to be determined the target playout rate and the playout adjustment rate. Target payout rate is computed by estimating the receiving rate component of the system. However, the playout tuning rate is computed by determining the expected change for the buffer component.

- The Receiving Rate Re-estimation (RRR)

  As mentioned in the buffer variation, the definition mirrors the deviancy of playout rate from the estimated receiving frequency. When the buffer variation appears, it means that the receiving rate has been changed based on the network channel condition. The system must change its playout rate according to the current receiving rate (target playout rate).

$$E[I] = \frac{T}{N_F} \tag{14}$$

where T = time interval between the last playout adjustment and the current playout adjustment
N_F = number of frames receivered during this period of time.

- The Expected Change Determination (ECD)

It is essential that playout is smoothly adjustable. Yet, reducing the tuning rate will increase the period it will take to complete a playout tuning and also increases the buffer outage probability. Hence, an accurate consideration of both the playout quality and risk of buffer outage is necessary. This study uses a predicted control parameter to monitor the adjustment rate. It depicts the amount of buffer that is to be adjusted, which forms the change between the buffer fullness level that is expected at the transition period end and the present level of the buffer fullness.

$$C = \begin{cases} (B_{MID} - \pi) - B_{LVL} & \mathrm{B_{LVL}} \geq \mathrm{B_{MID}} + \pi \\ (B_{LVL} - \pi) - B_{LVL} & \mathrm{B_{LVL}} \leq \mathrm{B_{MID}} + \pi \\ (B_{MID} - 2\pi) - B_{LVL} & \text{otherwise} \end{cases} \tag{15}$$

where B_{MID} = middle of the buffer size,
B_{LVL} = buffer fullness level
π = threshold of buffer fullness variation

Then the target playout interval ($E[I]$) and the expected change (C) are used to determine the transition period (T_s) of the playout adjustment.

$$T_s = C\left(\frac{1}{I} - \frac{1}{I - I_0} ln\left(\frac{I}{I_0}\right)\right)^{-1} \tag{16}$$

The playout interval is adjusted from the value I_0 to I according to the transition function.

$$I_t = I_0 + \frac{I - I_0}{T_s} \times (t - t_0) \tag{17}$$

where t = time variable
t_0 = current time
I_t = playout interval at time t
I_0 = current playout interval
T_s = duration of the transition period

4.2.2. Smooth Playout Control Algorithm

Smooth Playout Control was proposed to adjust the playout interval according to the estimated channel quality [56]. Its methodology was in two phases; the channel situation estimation and the playout interval tuning. Since channel condition reflects on the average receiving intervals, by computing the receiving interval and tuning the playout interval appropriately, the current channel condition could be tightly incorporated in computing the playout interval.

The receiving interval is estimated based on buffer fullness and the period to elapse time. When the video stream starts to playout, the present time is recorded. If there is no packet loss, smooth playout is experienced and also a nearly stable buffer fullness. On the other hand, if there is packet loss, buffer fullness will begin to reduce. If this reduction gets to a certain threshold (π), the elapsed time (TP) is noted. The relationship between the elapsed time and the number of the played frame during the period of decrement is:

$$T_P = t_F \times N_F \tag{18}$$

where N_F = number of played frames during the decrement period.

The relation between the playout interval t_F and the receiving interval t_R is governed by

$$t_F \times N_F - t_R \times (N_F - x) < t_R \tag{19}$$

Since t_F and N_F are known, t_R could be estimated by choosing the minimum value that satisfies the above relationship. The estimated receiving interval (t_R) is used to adjust the playout interval.

4.2.3. Safety Guaranteed Smooth Playout Algorithm (SG-SAMP)

Extension of SAMP technique has been proposed in [55]. Two buffer level threshold values are added to the receiver buffer to eliminate the limitation of the previous works. The AG-SAMP algorithm is proposed to correct the quality of playout and decrease the buffer outage chance. A safe area, an unsafe area and variation threshold are predefined. By controlling the arriving buffer level, playout control is activated when buffer level moved from the defined safe area. When the buffer level is around the safe area, the buffer outage chance is lowered. On the other hand, the current buffer level moves from middle level, and the buffer outage chance rises according to the deviancy. This technique is aimed to keep buffer level in the safe area by predicting receiving rate and computing playout interval rate. Furthermore, the safe buffer range is defined to reduce the playout delay and at the same time keeping the frame loss rate.

4.2.4. Advantages and Disadvantages of the Buffer Variation-Based AMP Control

The buffer variation-based AMP techniques are proposed to consider buffer outage and playout rate smoothness. In general, smooth control schemes were designed for adaptive playout rate, which depended on adapting the playout rate and the adaptation speed. The main drawback is a potential risk of buffer outage exactly when the current buffer level is not enough for reacting. The adaptation speed cannot be fast enough to meet the variation of the receiving speed.

4.3. The Buffer Hybrid-Based Control

Hybrid AMP control techniques use buffer fullness and its variation to trigger playout rate adjustments. The probability of buffer underflow is estimated based on buffer fullness variation and achievable buffer fullness. When the average queue length reduction is greater than the achievable average reduction, the probability of buffer underflow is high after N time slots. On the contrary, when the average queue length reduction is less than the achievable average reduction, the probability of buffer underflow is low. Even though the buffer outage and playout smoothness are considered in these AMP techniques, the probability of buffer outage increases equally when the buffer fullness is not safe enough.

4.3.1. Online Buffer Fullness Estimation Based AMP Algorithm

The buffer fullness and its variation have been used to estimate the probability of buffer underflow [42]. The queue length reduction is characterized by the difference between the arrival rate and departure rate. When the queue length reduction is positive, it means the playout rate is too fast during the time slot. On the other hand, when it is negative, it means the playout rate is too slow. Mean queue length decreases during the time slot window is calculated and compared with the possible mean decrease during the forthcoming time slot. The average queue length reduction (m_u) and the achievable average reduction (a_u) are calculated as follows.

$$m_u = E\left[\frac{\sum_{i=1}^{N} I_{n+i}}{N}\right] \tag{20}$$

$$a_u = \frac{B_{LVL} - B_L}{N} \tag{21}$$

where I_i = queue length reduction

$$I_k = \mu_k - \lambda_k \tag{22}$$

where λ_k = number of arrival frames during slot k
μ_k = number of departure frames during slot k.

The proposed online estimation based adaptive playout control technique assesses the playout rate control based on the queue length (m_u) and the average queue length reduction (a_u) per slot. When the present buffer fullness is below or above the given threshold (low/high), a buffer underflow or overflow has occurred and the playout rate should be decreased or increase accordingly to recover from the buffer outage. On the other hand, if the current buffer fullness exceeds the given low threshold and the actual average queue reduction is greater than the achievable averaged reduction per slot of time, the playout rate must be reduced in order to decrease the buffer underflow probability. Also, if the current buffer fullness is below the given high threshold and as well the actual average queue reduction is greater than the achievable averaged reduction per slot of time, the playout rate must be reduced in order to decrease the buffer overflow chances.

4.3.2. Advantages and Disadvantages of the Buffer Hybrid-Based Control

The buffer Hybrid-based AMP techniques are proposed to overcome the limitations of the previous techniques (buffer threshold-based and buffer variation-based). It estimates the buffer outage based on both buffer fullness variation and achievable buffer fullness. Even the buffer level is considered, it still has a drawback which is unpredictable channel quality. When the buffer level is low and channel quality was changed to bad, the system cannot react and adjust the playout rate to the buffer outage.

5. EVALUATION METRICS FOR AMP MECHANISM

The AMP technique is affected by the degrading quality resulting from the playback speed variations, buffer underflow, buffer overflow and playout delay. To estimate the efficiency of the playout system, some effective metrics were defined. The key performance metrics defined for evaluating AMP playout control systems are as follows:

5.1. Frequency of Playout Interruptions

The buffer underflow case accrues when there is no frame existing in the playout receiver buffer [42, 43, 50]. The incidence of buffer underflow will interrupt the video playout and destroy the quality of the video streaming. Hence, the average number of buffer underflow seems to be an essential metric to evaluate an AMP technique.

$$Avg_F = \frac{N_U}{T} \tag{23}$$

where N_U = number of playout interruption
T = video playing duration

5.2. Playout Delay

The playout delay is the time duration for pre-buffering, the time duration from the first frame received until its playout time [41, 43, 47]. Larger pre-buffering frames result in few underflow circumstances. Likewise, since there is the existence of a trade-off between the underflow occurrence frequency and the playout delay, an AMP scheme which reduces both playout delay and the probability of underflow to occur at the same time is a challenging issue.

$$P_{Del} = \frac{\sum_{i=1}^{n} P_{Del}^{i}}{T} \tag{24}$$

Where P_{Del}^{i} = playout delay for each i^{th} times

5.3. Average Buffer Overflow

Whenever the received buffer reaches fullness, newly arriving frames will be cast-off [41][5]. The video quality is seriously destroyed since serious frames are lost. Therefore, it is essential that the chance of buffer overflow happening must be reduced in order to produce high-quality video. Hence, the average buffer overflow is an essential metric for computing an AMP algorithm which is calculated as shown in next equation.

$$Buf_o = \frac{N_o}{T} \tag{25}$$

where N_o = number of discarded frames

5.4. Short-Term Standard Deviation of Frame Playout Interval

The variation of the frame playout interval has been used to evaluate the playout process smoothness [42, 43, 48]. A large amount of variation usually causes inferior quality. While this metric indicates the long term deviation of the playback it also represents one of the criterions of QoS. The metric is calculated as follows:

$$Std_{Dev} = \sqrt{\frac{\sum_{i=0}^{N} (t_i - T)^2}{N - 1}} \tag{26}$$

where N = number of video frames
t_i = playout interval of i^{th} frame
T = mean of playout interval.

5.5. Variance of Distortion of Playout

Certainly, the disruption of playout destroys the quality of experience (QoE) extremely. To reduce the chances of playout disruption occurring, the playout rate is adjusted dynamically. However, the user-perceived quality might be also affected by the variance of playout rate. The distortion of playout (DoP) has been used in [41, 47] to handle the effect of playout rate deviations. The effects of playout rate deviation, underflow, and frame losses are considered simultaneously to calculate the DoP. The DoP of the nth frame was defined as the following formula:

$$DoP(n) = \begin{cases} |S - s_n - T| & 1th\ frame\ after\ preroll \\ T & lost\ frames \\ |s_n - T| & otherwise \end{cases} \tag{27}$$

where s_n = actual playout duration of the n^{th} frame
S = time interval from a certain underflow to the next playback instant.

Accordingly, the mean of DoP was computed as follows:

$$\overline{DoP} = \frac{\sum_{n=1}^{k+y} DoP(n)}{k+y} \tag{28}$$

where k = number of loss frame
y = total number of frames being played.

The variance of DoP (VDoP) was computed by the following formula:

$$VDoP = \frac{\sum_{n=1}^{k+y} DoP(n)}{k+y} - \overline{DoP}^2 \tag{29}$$

6. CONCLUSIONS

This article presents a review of the literature on the AMP techniques. AMP technique is very important in video streaming over best effort networks as it ensures that the intra-stream synchronization quality of the video stream is protected.

A substantial number of studies have been carried out to improve AMP technique, little variations such as the playout rate control, buffer outage estimation, playout delay reduction and the evaluation metrics. This article presents an overview of these issues and the classification of AMP techniques based on playout control.

Sustaining a good trade-off among the reducing playout delay, the risk of buffer outage and the quality of video playout is a challenge. Thus, an appropriate AMP technique should be designed according to network conditions, variations of video bit rate to derive the estimation of the buffer outage time and reducing the amount of data buffering as well.

Hence, from the literature, the existing AMP techniques still face difficulties in terms of choosing an appropriate buffer threshold, playout speed adjustment, and buffer outage estimation. Therefore, our future work will be to optimize the AMP technique to cope the aforementioned problems.

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