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Rate Adaptation for Buffer Underflow Avoidance in Multimedia Signal Streaming

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***Abstract*— In media transmission over packet networks, one of the most challenging issues is the avoidance of receiver buffer underflows. Among the approaches proposed to solve this problem, the source rate adaptation is promising, due to the availability of multi-rate encoders.**

In order to succeed, rate-adaptive approaches should take into account not only the network throughput, but also the playout buffer fullness. Using this information, it is possible to determine at which rate the source should encode the stream, in order to avoid underflows.

In this paper, we derive and analyze the expression of the underflow probability; we show that it can be written in closed form as a function of the source rate, the receiver buffer fullness and the channel statistics. In particular, we study the dependency on the media rate, and we show how to achieve infinitesimal underflow probabilities even in presence of a high channel variance. Furthermore, we specialize this formulation for a CBR channel.

In case of channel variations during the playout, the buffered data may not suffice to ensure zero underflow probability; based on the previous formulation, we present an algorithm to recompute the source rate that allows continuous media playout. This algorithm can be safely iterated at every channel throughput change, and proved to be effective in avoiding buffer underflows.

I. INTRODUCTION

In recent years, the interest of consumers in services over packet networks experienced a fast growth, mainly due to the wide diffusion of high-capacity wireless and wired access technologies. Along with traditional data services like ftp and web browsing, the increased network throughput, together with the development of highly efficient media coding techniques, made transmission of multimedia contents possible across today's Internet. Applications involving streaming of media resources impose several new challenges, mainly related to the bursty nature of the network. In particular, the transmission of stored video, audio, speech and synchronized text imposes strict bounds on the delivery time and jitter, and at the same time requires low loss rates, to avoid quality degradation in the decoded signal.

To compensate for the variability of network throughput, a certain amount of data needs to be stored within the receiver before starting the playout; this buffer will supply data and avoid media interruptions if the network is unable to deliver packets for a limited interval of time. The time spent for this pre-buffering operation should be, at the same time, long enough to ensure jitter compensation for the entire duration of the playout, and short enough to avoid a long waiting for the user.

In the case of constant bitrate (CBR) media transmission over a constant throughput channel, the computation of the minimum pre-roll time is straightforward [1]; if instead the channel is variable, the buffering time should be longer with respect to the CBR case, to compensate throughput variability, therefore the user may experience a long waiting before enjoying the requested multimedia content. Furthermore, in many cases the channel statistics may be unknown, and this computation may be impossible.

Several solutions for receiver buffer underflow avoidance have been proposed in recent studies. A solution may be buffering for a given pre-roll time, either pre-determined or computed under the hypothesis of constant rate and constant throughput, and then reacting to channel variations whenever changes are detected. This operation may be performed in different ways. One of the possible techniques is *Dynamic Playback Rate* [2] which makes use of a threshold for data buffering. If data already buffered exceed the threshold, the media is played at its natural speed; if instead the amount of data buffered is below, the playback rate is reduced proportionally. A similar buffer management technique is the *Adaptive Media Playout* (AMP) [3]–[5] in which the client varies the playback rate, increasing and decreasing its speed, according to the network throughput (and consequently to the playout buffer fullness). In this sense, it extends Dynamic Playback Rate. Both approaches may experience problems if the buffer continues decreasing for a long time, since it is not possible to vary the playout speed beyond a certain limit without introducing excessive perceptual distortion. Techniques to forecast the channel capacity and delay, and to adapt the playout speed in advance, have been studied in [6], [7],

Other buffer underflow avoidance methods are based on the determination of the optimal buffer size. In [8], a buffer dimensioning technique to avoid buffer underflows is presented in the case of variable media rate and variable channel capacity. The problem was also studied, by means stochastic processes, for queues in a packet network [9]; if the arrivals and departure processes are known, then it is possible to evaluate the optimal buffer allocation. The management of source and receiver

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buffer dimensioning in case of a VBR media can be found also in [10].

Another family of techniques behaves as follows. The pre-buffering time is set to a given value, and in case the amount of data present within the receiver is insufficient, they force the source to change the coding rate. In [11], rate recomputing is performed considering both channel delay and loss rate; receiver buffer fullness is taken into account using a system of thresholds. In [12], the source rate is controlled according to the buffer fullness and channel throughput, for transmission over TCP.

In this paper, we propose a source media rate adaptation algorithm. It stores data during a pre-roll period which is determined under the hypothesis of a constant channel throughput; in case of channel variation, the media rate is recomputed taking into account both the new channel rate and the playout buffer fullness at the time of the event. This algorithm selects the new source media rate as the one which guarantees no buffer underflows (and consequently no media freezing), supposing that no further channel variation will occur; however, this recomputing can be safely reiterated at every network throughput change. We show that the buffer underflow probability can be extremely high in case of non-reactive approaches; to demonstrate this, we derive the analytical expression of the underflow probability, supposing channel samples as independent and identically distributed (i.i.d.).

The paper is organized as follows. In Section II, we describe the mathematical framework of the problem, and derive the expression of the buffer underflow probability at a generic time during the playout; we study how this probability is affected by modifications of the source rate. We further specialize this formulation to the constant channel throughput case. In Section III, we present an algorithm for rate selection, based on the above theory, according to the channel throughput and buffer fullness. We show the behavior of this algorithm for some throughput patterns in Section IV. Finally, we draw conclusions in Section V.

II. Problem formulation

Under the hypothesis of constant channel throughput C^* and constant media rate R^*, and considering $C^* < R^*$, data buffering at receiver is necessary to ensure a continuous playout; the buffering time t_B can be simply computed (see, e.g., [1]) as:

$$t_B \geq t_D \left(R^*/C^* - 1 \right), \tag{1}$$

where t_D indicates the duration of the stream to be played. If the receiver allows a pre-buffering period of this length, the underflow probability for time points $t \leq t_D$ is zero.

If the channel throughput diminishes and the media rate is not adapted accordingly, then the data buffered may not suffice to ensure continuous playout. The exact time position of the buffer underflow depends on the particular evolution of the channel and it can be located only if the channel throughput value is exactly known at each instant. Instead, if the specific time evolution is unknown but the statistic characterization of the channel is available, it is possible to compute the expression of the buffer underflow probability as a function of time.

In the development of this study, we suppose the time to be divided into slots of duration δ_t; the generic time instant t can be defined as

$$t = n_t \delta_t, \tag{2}$$

where n_t indicates the slot number. Similarly, we define the buffering time as

$$t_B = n_B \delta_t \tag{3}$$

and the media duration as

$$t_D = n_D \delta_t. \tag{4}$$

We suppose the media rate and the channel throughput to remain constant within each time slot, which is a reasonable assumption if the time δ_t is small (e.g., in the order of milliseconds).

After the pre-roll time t_B, the playout starts and the buffer is full up to a level of B_s bits. Under the hypothesis of average channel throughput μ_C during pre-roll, its value is

$$B_s \simeq \mu_C n_B \delta_t. \tag{5}$$

During slots $\in [n_B, n_B + n_D]$, information is taken from this buffer at constant rate R^* bps, while the arrival rate follows a stochastic process $C \sim f_C(x, n_t)$. After the playout starts, at a generic time slot n_t, the buffer fullness is:

$$B(n_t) = B_s - R^*(n_t - n_B)\delta_t + \delta_t \sum_{k=1}^{n_t - n_B} C(x,k), \tag{6}$$

where the random variable $C(x,k)$ is the stochastic process sampled at the given time slot k. According to the *central limit theorem*, and considering $C(x,k)$ independent and identically distributed (later, *i.i.d.*), it is:

$$\sum_{k=n_B}^{n_t} C(x,k) \sim \mathcal{N}((n_t - n_B)\mu_C, \sqrt{n_t - n_B}\sigma_C). \tag{7}$$

The probability of experiencing an underflow before a given slot is the probability of having less than zero bits within the buffer at any moment before t:

$$P_u(n_t) = P\left(B(n_t) < 0\right). \tag{8}$$

By definition, the buffer cannot contain a negative number of bits; the above condition represents the probability that the media playout requires, before a given time slot, more data than the amount contained in the buffer. In formulas, the probability of this event is:

$$\begin{aligned} P_u(n_t) &\simeq P(B_s - R^*(n_t - n_B)\delta_t + \\ &+(\sqrt{n_t - n_B}\sigma_C Z + (n_t - n_B)\mu_C)\delta_t < 0), \end{aligned} \tag{9}$$

where $Z \sim \mathcal{N}(0,1)$. It is straightforward to obtain the mathematical expression of this quantity:

$$
\begin{aligned}
& P_u(n_t) = \\
= \;& P\left(Z < \frac{R^*(n_t - n_B) - B_s/\delta_t - \mu_C(n_t - n_B)}{\sqrt{n_t - n_B}\sigma_C}\right) \\
= \;& \Phi_Z\left(\frac{R^*(n_t - n_B) - B_s/\delta_t - \mu_C(n_t - n_B)}{\sqrt{n_t - n_B}\sigma_C}\right) \\
= \;& \frac{1}{2}\left(1 + erf\left(\frac{(R^* - \mu_C)(n_t - n_B) - B_s/\delta_t}{\sqrt{2}\sqrt{n_t - n_B}\sigma_C}\right)\right), \quad (10)
\end{aligned}
$$

where $\Phi_Z(x)$ is the cumulative density function (*cdf*) of $Z \sim \mathcal{N}(0,1)$.

To obtain the probability density function of the buffer underflow (indicated in the following as $p_u(n_t)$), it is sufficient to derive Formula (10). Indicating with $g(n_t)$ the argument of the erf function, it is:

$$
\begin{aligned}
p_u(n_t) &= \frac{\mathrm{d}P_u(n_t)}{\mathrm{d}n_t} \\
&= \frac{\mathrm{d}P_u(g(n_t))}{\mathrm{d}g(n_t)}\frac{\mathrm{d}g(n_t)}{\mathrm{d}n_t} \\
&= \frac{e^{-g^2(n_t)}}{\sqrt{\pi}}\frac{\mathrm{d}g(n_t)}{\mathrm{d}n_t}. \qquad (11)
\end{aligned}
$$

The complete expression can be written easily from Formula (11).

To ensure that the buffer fullness does not reach zero at any time during transmission, it is sufficient to have p_u infinitesimal at each time slot $\in [n_B, n_B + n_D]$. To better clarify how to obtain this condition, we show some examples.

Let the initial (constant) rate of a media source be 100 kbps, and the channel throughput have statistics μ_C=80000 kbps and σ_C=20000. We suppose the sequence has a duration of 90 s, and from Formula (1) we estimate a pre-roll time $t_B = 22.5$ s. Under these conditions, the media stops the playout at time $t_E = t_B + t_D = 112.5$ s. Figure 1 shows the buffer underflow probability p_u around the point t_E.

It can be seen that, for $R^* = 100$ kbps, the buffer underflow probability starts being non-infinitesimal around time $t = 108$ s; this means that it is highly probable to have a freeze in the last 4.5 s of the playout. Figure 1 also shows the plots of $p_u(n_t)$ for rates in the range $R^* \in \{98.5, 99.0, 99.5\}$ kbps; we vary only the media rate, while the pre-buffering time is kept constant to 22.5 s. The effect of a reduction in the media rate is that the average point of $p_u(t)$ is shifted in the future, and therefore the buffer underflow probability for a generic instant $t \leq t_E$ diminishes. For $R^* = 98.5$ kbps, the buffer underflow probability is always infinitesimal inside the playout interval.

A. The CBR channel case

To further show the consistency of this formulation, we will compute the expression of the underflow probability as a function of time in the simplest case, when the channel is constant at rate C^*. This result cannot be obtained directly from Formula (11), since it corresponds $\sigma_C = 0$ and this quantity appears at denominator.

In this case, with constant media rate R^* and if we set the pre-roll time according to Formula (1), we showed that the underflow probability will be a delta function centered in t_E, since the first and only underflow event will be placed exactly at the end of the playout.

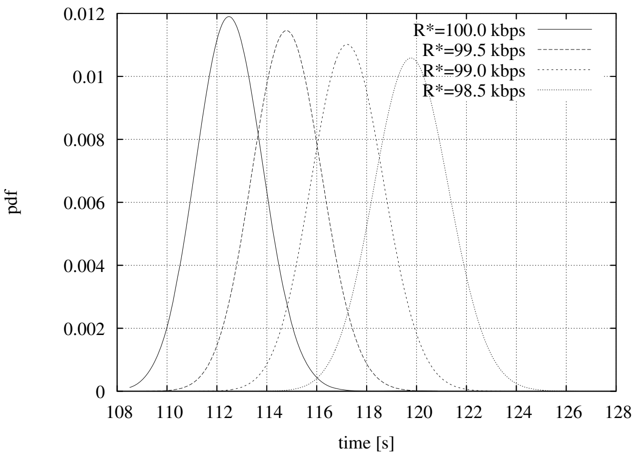

Fig. 1. Buffer underflow probability as function of time, for different values of R^*; the channel throughput statistics are $\mu_C = 80000$ and $\sigma_C = 20000$, the media lasts 90 s; pre-buffering time is set to 22.5 s and the media playout stops at 112.5 s.

Formula (10) needs some manipulation to be meaningful. If the channel is constant at rate C^*, then $f_C(x) \sim \delta(x - C^*)$; its variance is zero, the average is C^* and the argument of the *erf* is infinite. In particular, according to the sign of the numerator:

$$
\begin{aligned}
&\frac{(R^* - C^*)(n_t - n_B) - B_s/\delta_t}{\sqrt{n_t}\sigma_C} = \\
&= \begin{cases} +\infty \text{ if } (R^* - C^*)(n_t - n_B) - B_s/\delta_t \geq 0 \\ -\infty \text{ if } (R^* - C^*)(n_t - n_B) - B_s/\delta_t < 0 \end{cases} \qquad (12)
\end{aligned}
$$

Given the definition of the error function erf, the effect on P_u becomes:

$$
P_u(n_t) = \begin{cases} 1 \text{ if } (R^* - C^*)(n_t - n_B) - B_s/\delta_t \geq 0 \\ 0 \text{ if } (R^* - C^*)(n_t - n_B) - B_s/\delta_t < 0 \end{cases} \qquad (13)
$$

Remembering that in this case it is $B_s = C^* n_B \delta_t$ the condition to determine the sign of the infinity becomes:

$$
\begin{aligned}
&(R^* - C^*)(n_t - n_B) - C^* n_B < 0 \Rightarrow \\
&\Rightarrow (R^* - C^*)n_t - R^* n_B < 0 \Rightarrow \\
&\Rightarrow n_t < \frac{n_B R^*}{R^* - C^*}
\end{aligned} \qquad (14)
$$

Subtracting n_B at both members:

$$
n_t - n_B < n_B \frac{C^*}{R^* - C^*}. \qquad (15)
$$

We set the number of buffering slots at the minimum as defined in Formula (1):

$$
n_B = n_D\left(R^*/C^* - 1\right) = n_D \frac{R^* - C^*}{C^*}. \qquad (16)
$$

Substituting in Formula (14), we get:

$$n_t - n_B < n_D \Rightarrow n_t < n_B + n_D. \quad (17)$$

This means that:

$$P_u(n_t) = \begin{cases} 1 \text{ if } n_t \geq n_D + n_B \\ 0 \text{ if } n_t < n_D + n_B \end{cases} \quad (18)$$

Since $P_u(n_t)$ is a step function, with step in $n_D + n_B$, its derivative is

$$p_u(n_t) = \delta(n_t - (n_D + n_B)), \quad (19)$$

which represents the slot in which the transmission ends. This means that the buffer underflow probability is zero in all of the slots before the media playout end time, and this coincides with the result we expected.

III. Proposed rate adaptation algorithm

We showed that, if the channel varies over time and the media rate remains constant, the probability of having an underflow can be computed in closed form.

The proposed formulation involves four quantities, the channel average throughput μ_C and standard deviation σ_C, the media rate R^* (supposed constant) and the buffering slots n_B. The first two, related to the channel, are not dependent on the application and can only be measured; the media rate and the buffering time can be properly set to avoid buffer underflows. Using the same pre-roll time and channel characteristics, a decrease in the media rate is helpful in diminishing the underflow probability for time slots during the playout; an increase in the rate produces the opposite effect.

We also showed that, if the channel throughput is considered constant, the pdf of the underflow probability degenerates into a delta function; as in the case of variable channel, modifying the media rate it is possible to drive the peak of this delta outside the playout interval.

Until now, we considered that the channel characteristics (μ_C and σ_C) remain constant during the experiments. In this Section, we show that it is possible to apply the same formulation, if a channel variation is detected, at any point in time during the playout. The pre-roll duration is set supposing the channel will remain stationary during pre-roll and transmission; if for any reason the average channel throughput diminishes, or its variance increases significantly, and the media rate is not adapted accordingly, the quantity of buffered data may not suffice to ensure continuous playout. In this case, it is necessary to recompute the media rate *during* the playout.

In the following, we will suppose a CBR channel throughput; extension to the case with variance can be similarly derived and involves a heavier notation.

Since the computation is made my means of the formulation of Section II, the new rate at which the source should encode the stream takes into consideration both the new channel throughput value and the buffer fullness. When the channel change is detected, the quantities μ_C and σ_C are measured; the buffer fullness is known at the receiver and the remaining parameter R^* can be determined by means of Formula (11).

Packets already buffered at the receiver are not discarded, but the request for a new media rate is immediately sent to the source; therefore, since several seconds of data may be present within the buffer, the effects of a rate change are not immediate in the playout quality. The media rate must be requested at the time t_{req}, but will be effective under the decoder point of view at time t_{pl}, after all the packets at the old rate are played; for this reason, the buffer level to be used in our formulation is the one at time t_{pl}, which needs to be predicted.

Supposing the channel throughput will not change again during the interval $t_{diff} = t_{pl} - t_{req}$, then the buffer at time t_{pl} can be computed from the buffer at t_{req}, by means of a linear relationship:

$$B(t_{pl}) = B(t_{req}) - (R_{old} - C_{new}) * t_{diff}, \quad (20)$$

since data will be taken from the buffer at the old media rate R_{old} and added at the new channel throughput C_{new}. The rate which should be used to avoid buffer underflows after the time t_{pl} is given by:

$$R_{new} = C_{new} + \frac{B(t_{req}) - (R_{old} - C_{new}) * (t_{diff} + t_{RT})}{t_D + t_B - t_{pl} - t_{RT}}. \quad (21)$$

Formula (21) means that, to ensure the buffer goes to zero at the playout end time, the new rate R_{new} should be equal to the new channel throughput C_{new} plus a margin given by the quantity of buffered data at the moment the packets at the new rate will be played $(B(t_{req}) - (R_{old} - C_{new}) * (t_{diff} + t_{RT}))$ divided by the time remaining until the end of the playout time $(t_D + t_B - t_{pl} - t_{RT})$. The quantity t_{RT} represents the *round trip time*, and takes into account the propagation delay of the new rate request. In the following, this parameter will be set to zero for simplicity.

The new rate is the one that forces the peak of p_u at time t_E, under the assumption that no more channel throughput changes will occur until the end of the playout; if other changes happen, the above approach can be iterated and the rate modified again.

Formula (21) may be used also to increase the rate in case the channel throughput grows. In this case, the new rate allows better quality; the rate is again chosen to ensure the end of the transmission coincident with the end of the playout.

The delay introduced between the new rate request and its effect at the decoder is not critical if changes occur in the final part of the sequence, since t_{diff} reflects the quantity of already buffered data, which is smaller at the end of the playout. The only limitation to the successful application of this technique is when the channel changes at a time distance smaller than t_{RT} from the end.

IV. Results

To test the correctness of the proposed rate adaptation system, we performed several buffer simulations. We present here the behavior of the adaptive system, along with the results obtained with a non-adaptive approach.

A. Scenario I, short channel degradation during playout

In the first scenario, transmission is performed over a channel which diminishes its rate for some seconds during the media playout. The channel starts at 400 kbps, it drops

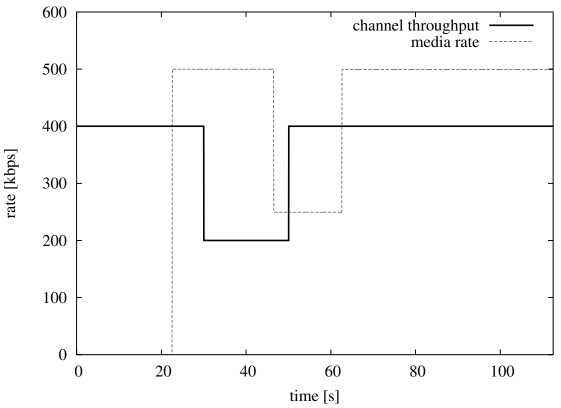

Fig. 2. Channel throughput and playout media rate obtained with the proposed rate control method; the channel diminishes for some seconds during the playout.

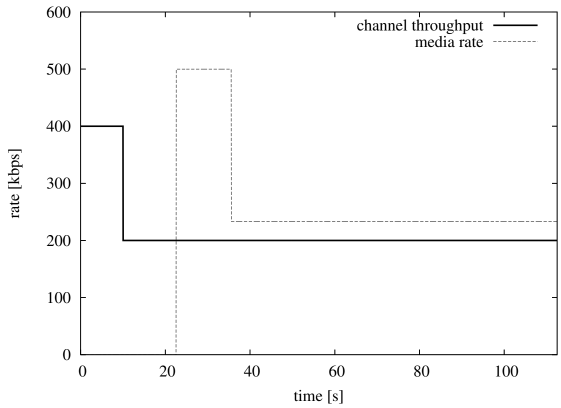

Fig. 4. Channel throughput and playout media rate obtained with the proposed rate control method; the channel throughput diminishes during the pre-roll and remains low until the end of the playout.

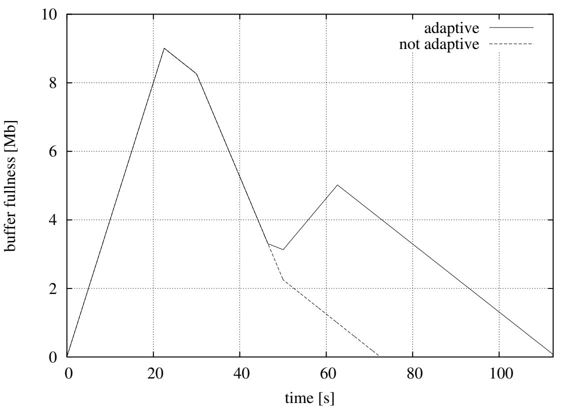

Fig. 3. Buffer evolution for the setting of Figure 2.

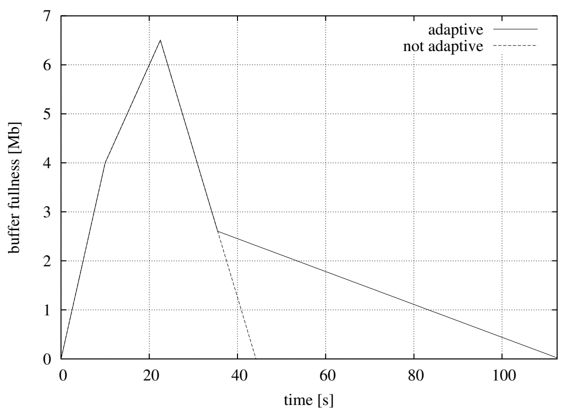

Fig. 5. Buffer evolution for the setting of Figure 4.

at 200 kbps after 30 s, and grows again to 400 kbps after 20 s; after this, it remains constant, as shown in Figure 2. The initial media rate is set to $R^* = 500$ kbps; from Formula (1), the pre-roll time is $t_B = 22.5$ s. This period is included in the interval in which the channel remains constant at 400 kbps, therefore no channel throughput changes occur during pre-roll. In Figure 2, the media rate seen at the decoder is shown. It remains zero during the buffering time, then it evolves adapting to the channel conditions.

The rate remains constant at 500 kbps for a certain time after the channel throughput changes; this is the effect of the packets already buffered at that rate. At the time the channel drops ($t = 30$ s), the buffer contains packets for 16 s; this quantity is equivalent to t_{diff} of Formula 21. When all of those packets are played, the decoder starts playing the part of the stream received at lower rate; according to the formulation, the new rate must be $R_{new} = 249.5$ kbps. When the throughput increases again, there are packets for $t_{diff} = 12$ s in the buffer. After this period, the rate increases again, and this time the rate value is set to $R_{new} = 499$ kbps.

The buffer fullness evolution under these conditions is shown in Figure 3. The buffer grows until $t_B = 22.5$ s at constant rate; at $t = 30$ s, the channel throughput changes and the buffer level starts decreasing fastly, because packets arrive at smaller rate while the decoder pops packets at the same speed as before. At $t = 46$ s, the decoder starts consuming information at the new rate, and the buffer decrease slows down. While the decoder is running at $R^* = 249.5$ kbps, the channel increases again, and information is received faster than it is consumed; in this interval, the buffer fullness grows up again, and this allows the buffer to avoid decoder starvation until the end. Note that we suppose that the source has always enough packets stored to always transmit at the channel rate throughput.

For comparison, Figure 3 shows the curve of buffer fullness in the case of a non-adaptive media rate, which remains constant at $R^* = 500$ kbps; in this case, the buffer goes in underflow well before the media playout end time.

B. Scenario II, permanent channel degradation during pre-roll

In the second scenario, the channel starts at a rate $C^* = 400$ kbps, and after 10 s it decreases to 200 kbps ans remains constant at this value, as shown in Figure 4. The media rate starts at $R^* = 500$ kbps and the pre-buffering period is $t_B = 22.5$ s, therefore the channel throughput change occurs during pre-roll. In this case, we decide to wait until the end of the buffering period before recomputing the rate. When the playout starts, the request for a new rate is sent to the source and, as in the previous scenario, after a certain time t_{diff}, it sets to $R^* = 233.5$ kbps until the end.

The buffer evolution in this scenario is shown in Figure 5. It can be seen how the buffer growth rate changes during the pre-roll period. After the beginning of the playout, the buffer fullness decreases fastly until information is not available at the new rate. Figure 5 also shows that if the rate is not adapted, the buffer underflow condition is reached about 20 s after the playback starts.

V. Conclusions

In this paper, we analyzed the receiver buffer underflow problem. First, we derived the analytical expression of the underflow probability at each time instant, given an initial buffering time and the channel statistics, and supposing the media rate constant. Based on this computation, we showed that whenever the channel average throughput or variance vary significantly, this probability may excessively grow and therefore produce poor-quality decoded streams several seconds before the media playout end time. We also showed that small modifications in the media rate value can be effective in decreasing the buffer underflow probability to infinitesimal values. We specialized the formulation in the case of a constant channel throughput, and demonstrated that in this case the underflow probability becomes a delta function.

We proposed a rate adaptation algorithm, based on the previous formulation, which ensures the center of the delta function to be located at the end of the playout. If packets have been buffered to compensate for a given throughput, and the channel changes with respect to this value, then the buffer fullness may decrease fastly and the media may freeze; we use the proposed algorithm to recompute the new media rate at which the source should encode the remaining part of the stream, to avoid decoder starvation. This approach can be reiterated every time a channel throughput change is detected.

The receiver computes the new desired rate according to the new channel condition, taking into account also the quantity of data already present in the buffer. the value is then fed back to the encoder.

This algorithm showed to achieve the goal of avoiding buffer underflows at the expenses of a variability in the bitrate. The proposed algorithm is effective also in case of multiple channel variations, even when they occur during the pre-buffering time, and introduces a negligible complexity into the system.

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