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Rate adaptation for Buffer Underflow Avoidance in Multimedia Signal Streaming

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Summary

- The pre-roll buffering problem in multimedia signal streaming;
- Underflow probability analytical computation;
  - effect of parameter variations;
- Proposed rate adaptation method;
- Results

The buffering problem

- In multimedia signal streaming, if the media rate exceeds the channel resources, it is necessary to set a pre-roll time.
- During this time, packets are received but not played, and the amount of buffered data grows.
- This buffer will supply data if the network throughput is insufficient
  - but only for a *limited* amount of time.

The buffering problem

- The optimal pre-roll time depends on several factors.
- In particular, it should be:
  - long enough to ensure protection against buffer underflows;
  - short enough to avoid a long waiting before enjoying the content.

Optimal pre-buffering time

- If the media rate (R*) and the channel throughput (C*) remain constant during transmission, then the computation is straightforward:

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

- This is the shortest time that ensures enough data has been stored, and avoids media interruptions during the playout.

Buffer underflows

- Media playout interruptions occur when the buffer contains, at a given time, less data than what is needed by the decoder.
- This lack of data is usually partially concealed, if the decoder implements a concealment technique, but buffer starvation may lead to severe quality degradation.

Buffer underflow probability computation

- We consider time as slotted.
- We suppose the media rate (R*) to remain constant, and consider a channel without memory (samples are i.i.d.), whose average and variance are known.
- We set a pre-buffering time computed as shown before.

Buffer underflow probability computation

- After the pre-roll period, the buffer is full at the level:

$$B_s \simeq \mu_C n_B \delta_t$$

- After pre-roll, the decoder pops data from the buffer at rate R*, while the network delivers packet at its instantaneous rate

- At a generic time, the buffer fullness is:

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

Buffer underflow probability computation

- Using the *central limit theorem*, we can approximate the last term of the sum as:

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

- The probability of experiencing a buffer underflow at a generic time is:

$$P_u(n_t) = P\left(B(n_t) < 0\right)$$

Buffer underflow probability computation

- Substituting the complete expression, we obtain:

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

- and then:

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

Buffer underflow probability computation

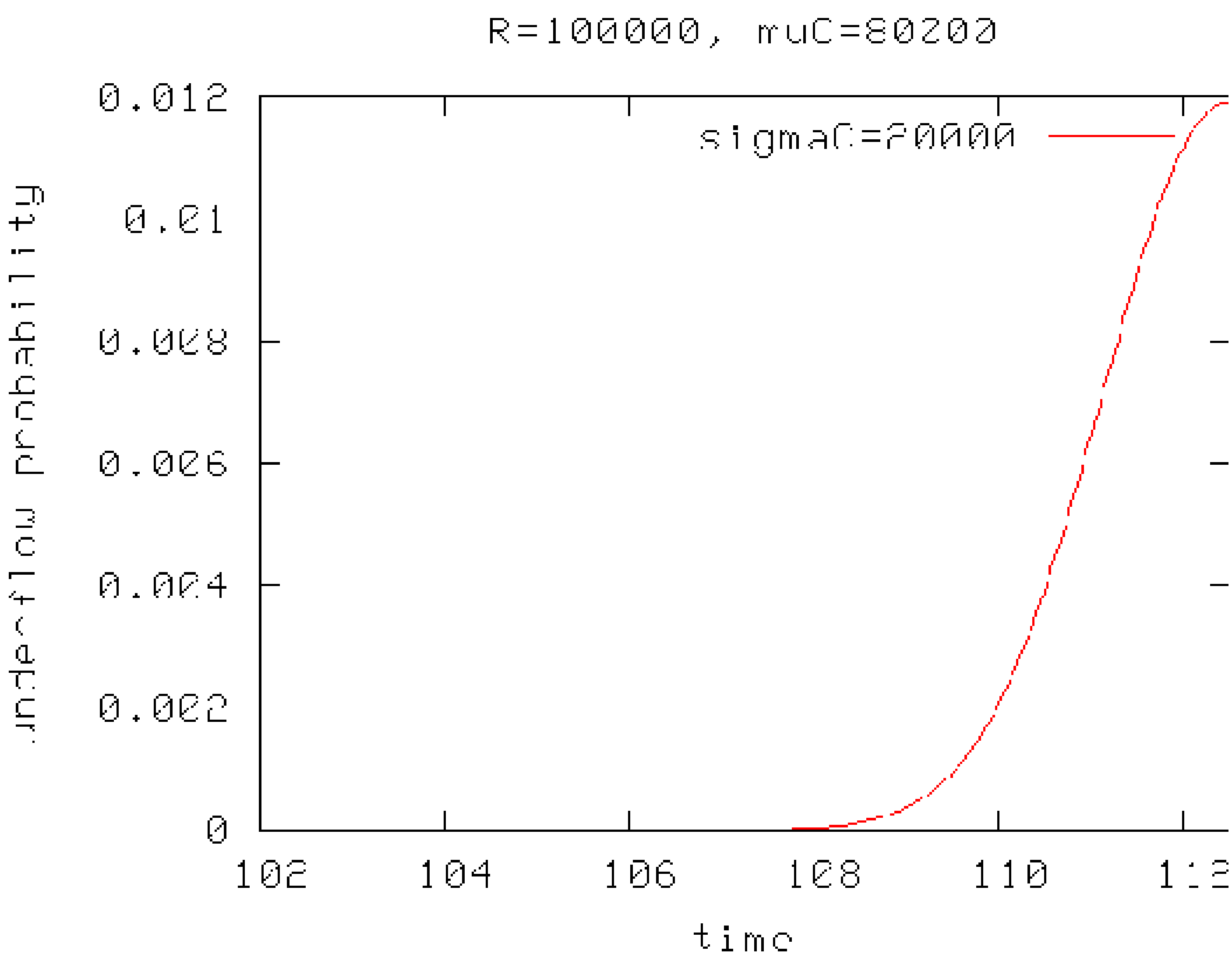
- P_u represents the cumulative distribution function (c.d.f.) of the buffer underflow event.
- To obtain the p.d.f., it is sufficient to derivate the expression of P_u:

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

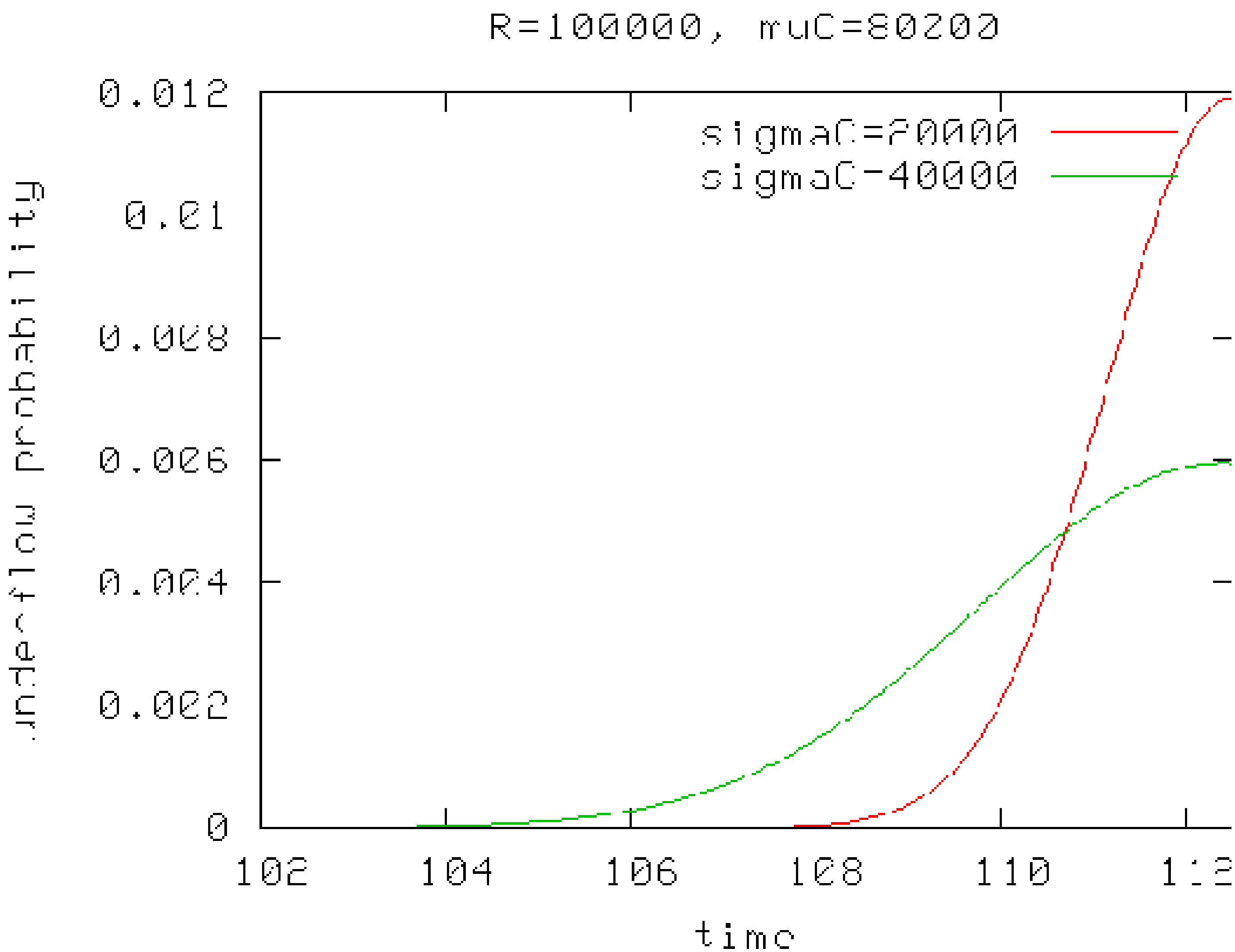
Effect of parameters on p_u

- Since we desire a low buffer underflow probability for all of the slots during the playout time, we studied how the parameters influence p_u
- In particular, we want to understand how to drive the parameter R^* when one or both of the network parameters (average and variance) change.

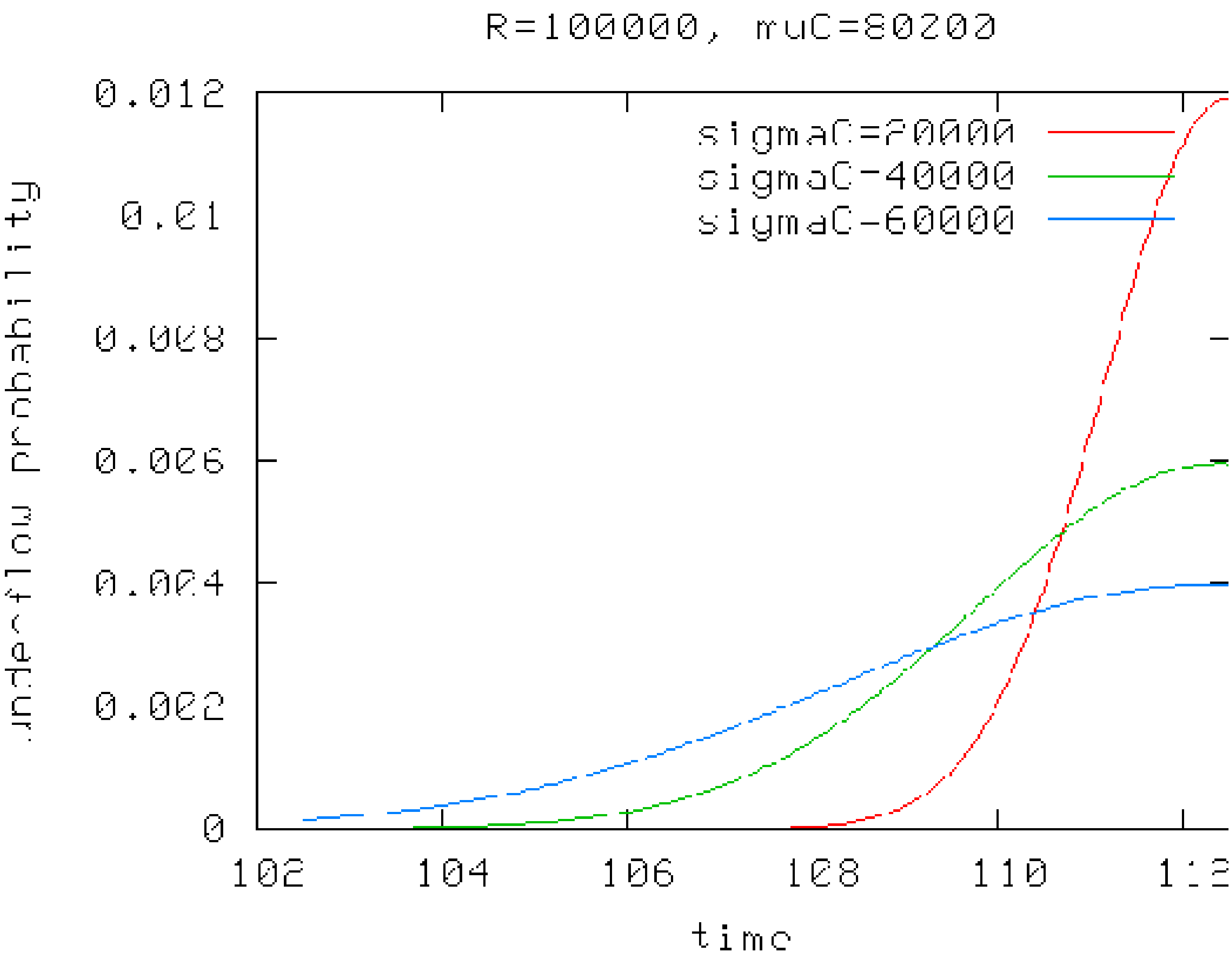
Effect of channel variance



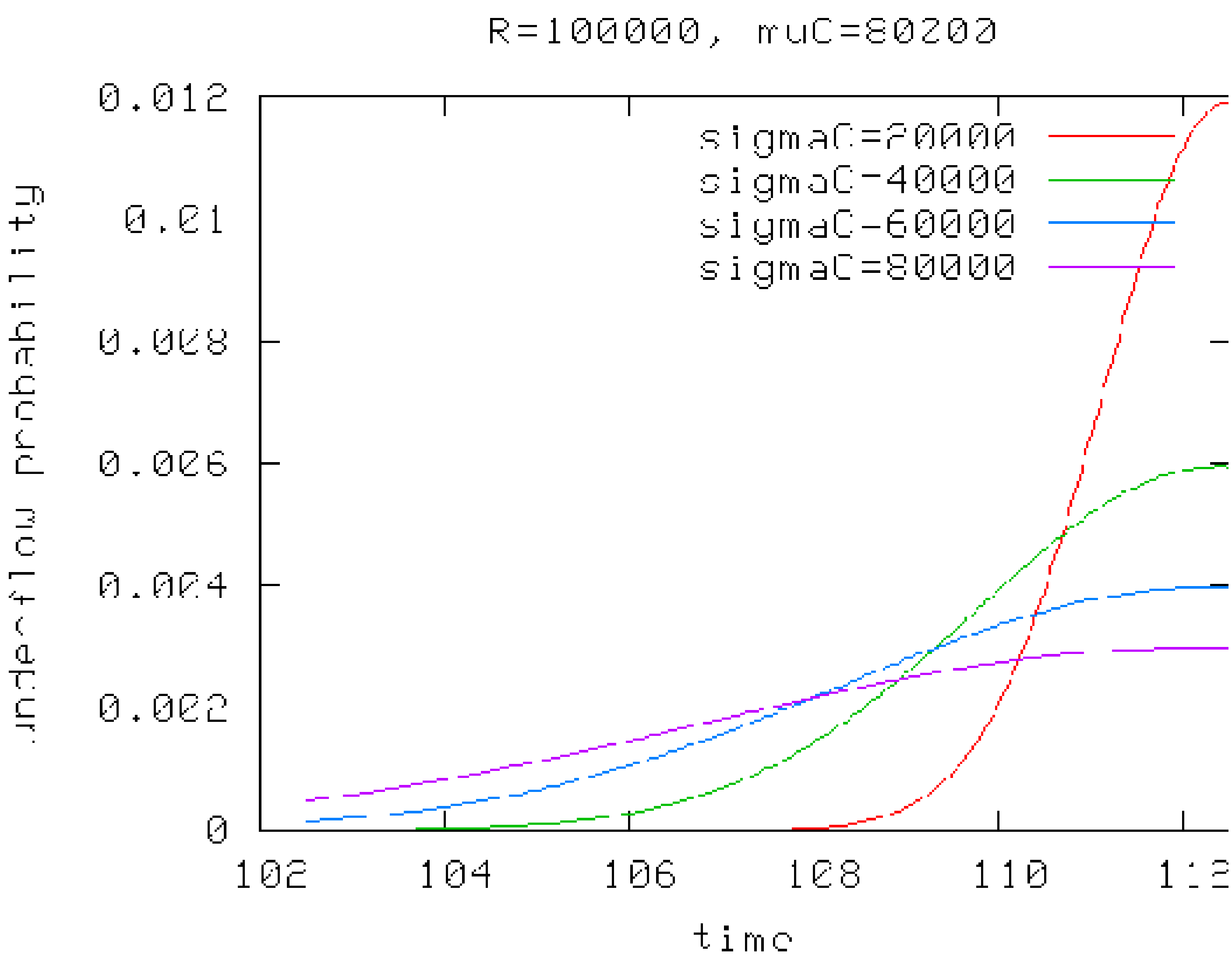
Effect of channel variance



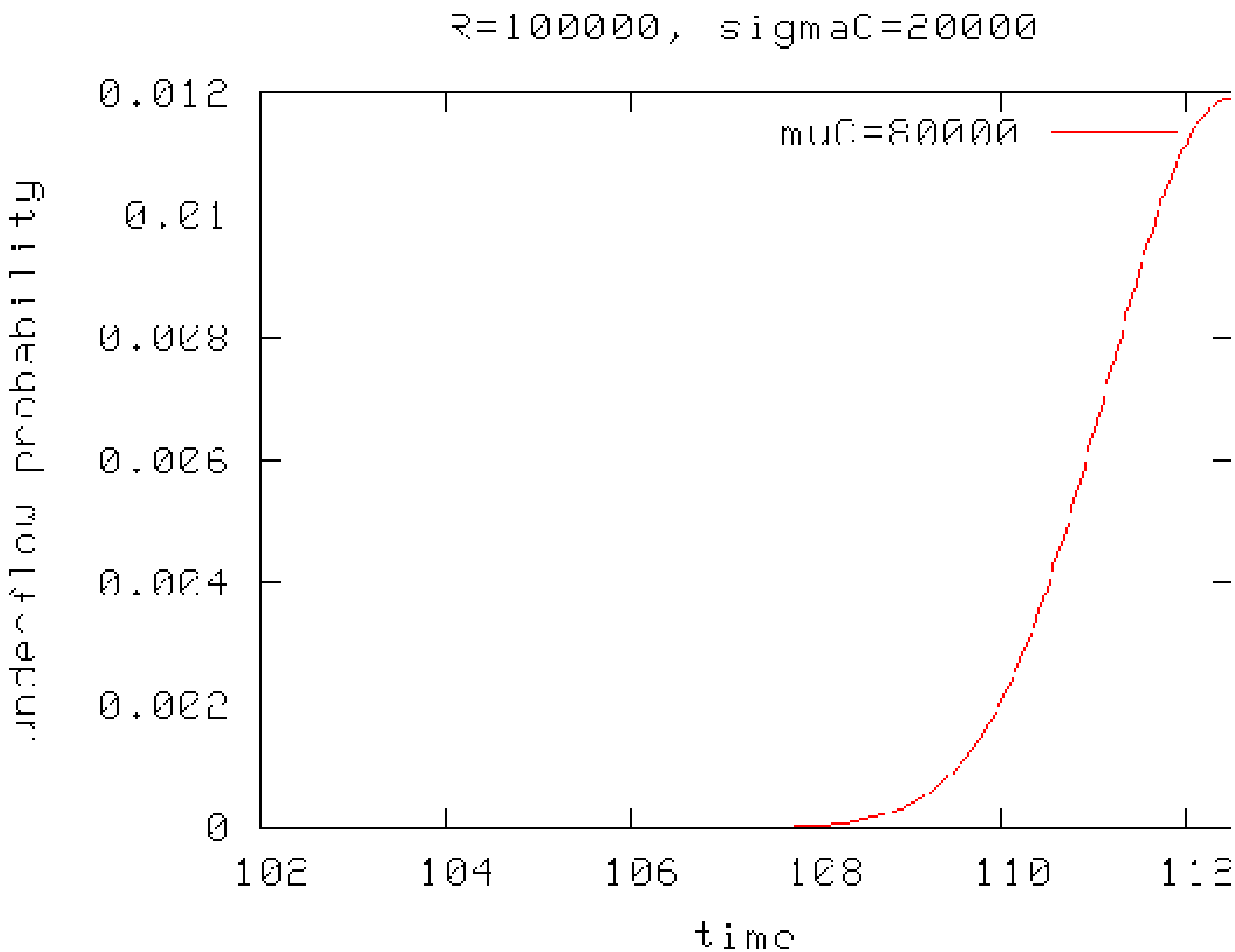
Effect of channel variance



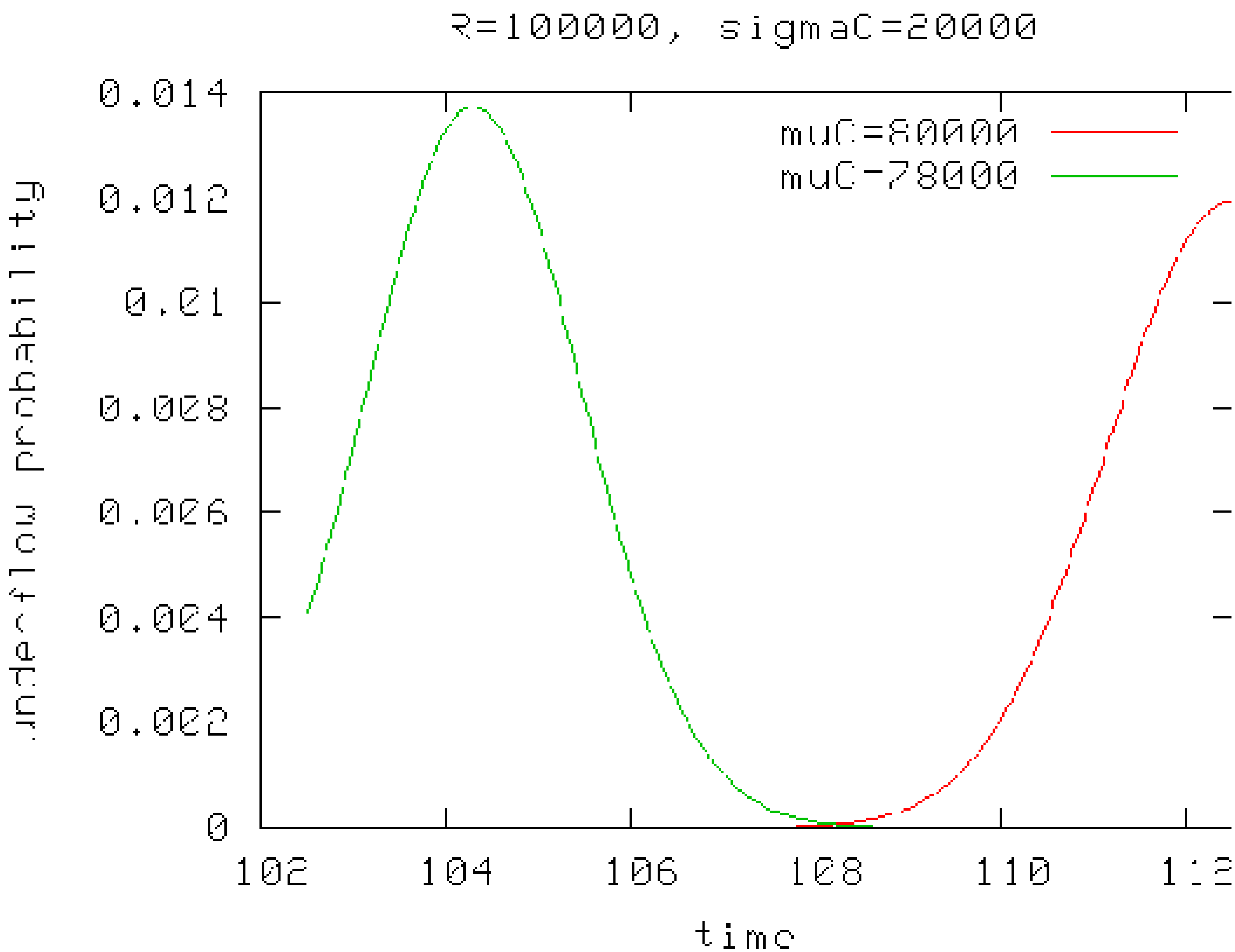
Effect of channel variance



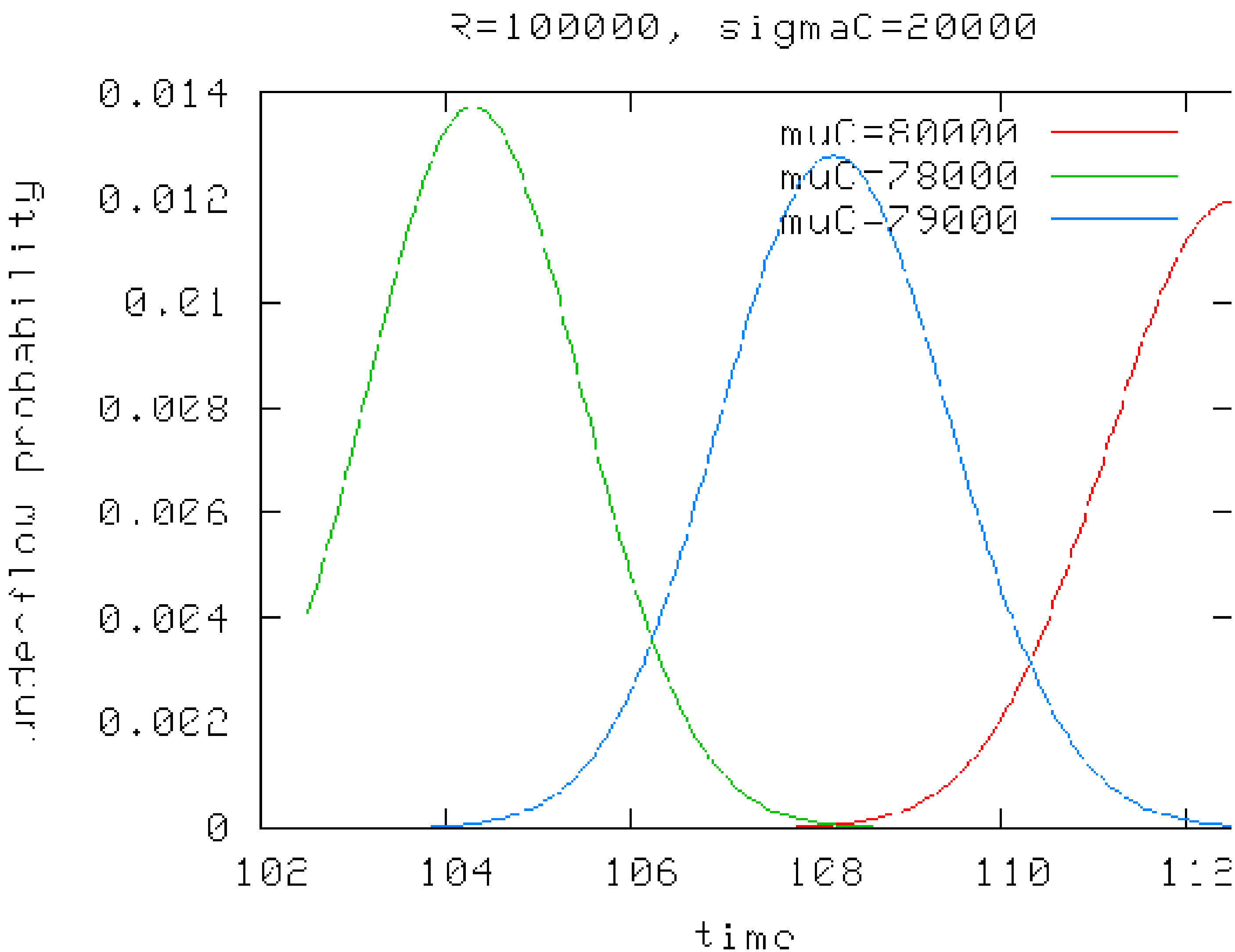
Effect of channel rate



Effect of channel rate



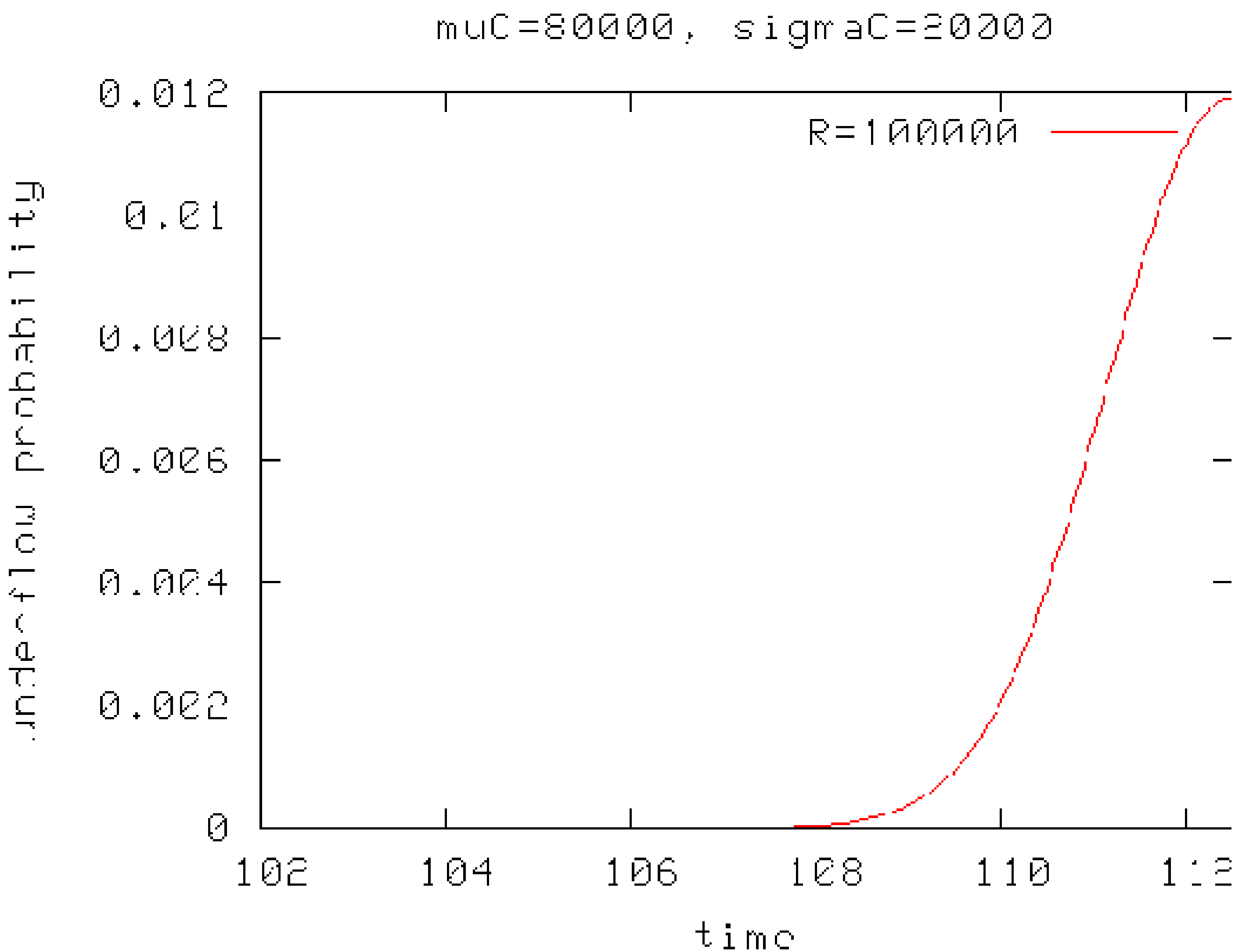
Effect of channel rate



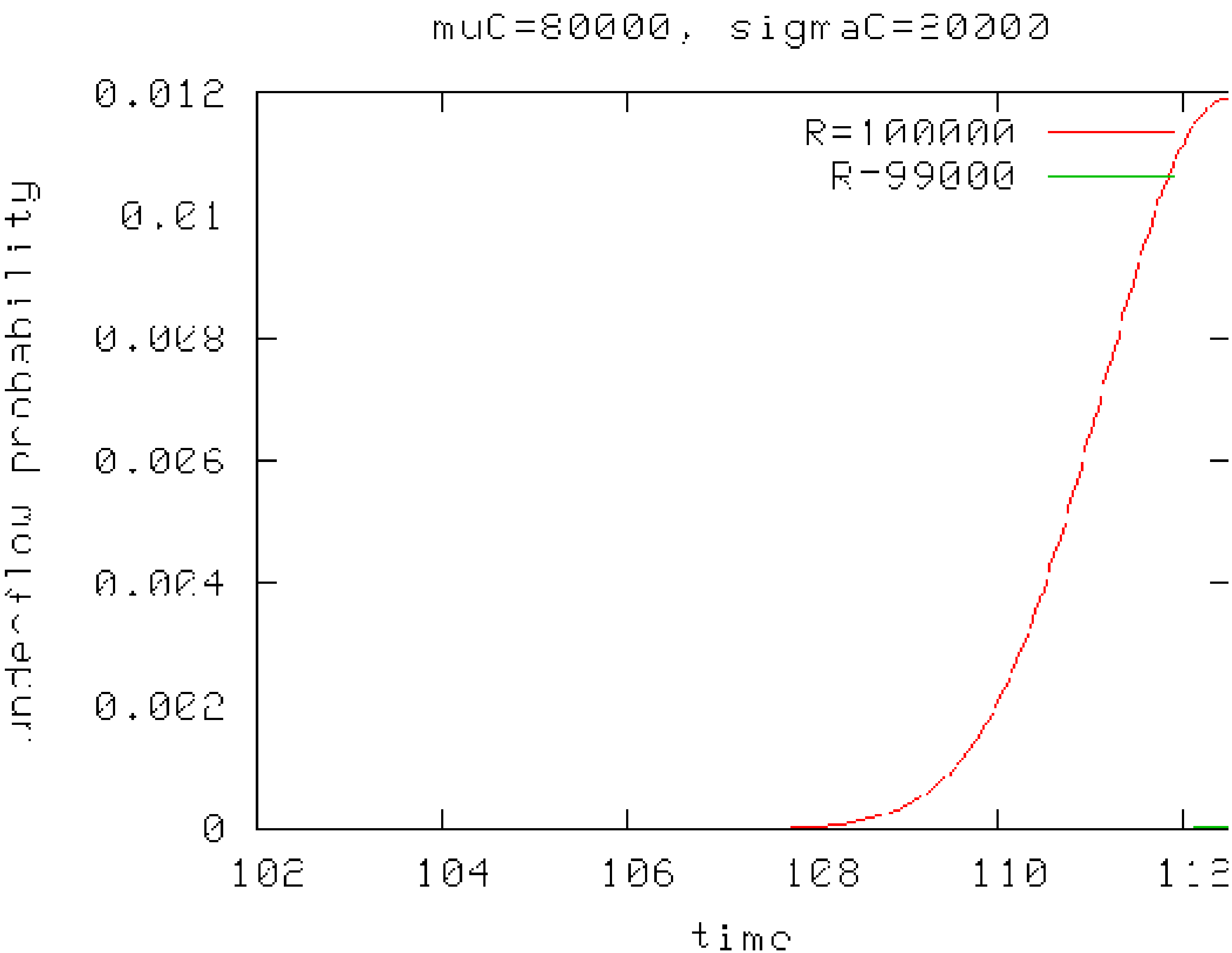
Effect of channel

- In general:
  - when the channel variance increases, the underflow probability increases also for time points far from the end of the sequence;
  - when the channel average rate decreases, the center of the underflow probability distribution is shifted left.
- We are interested in keeping the portion of distribution included in the playout interval as small as possible.

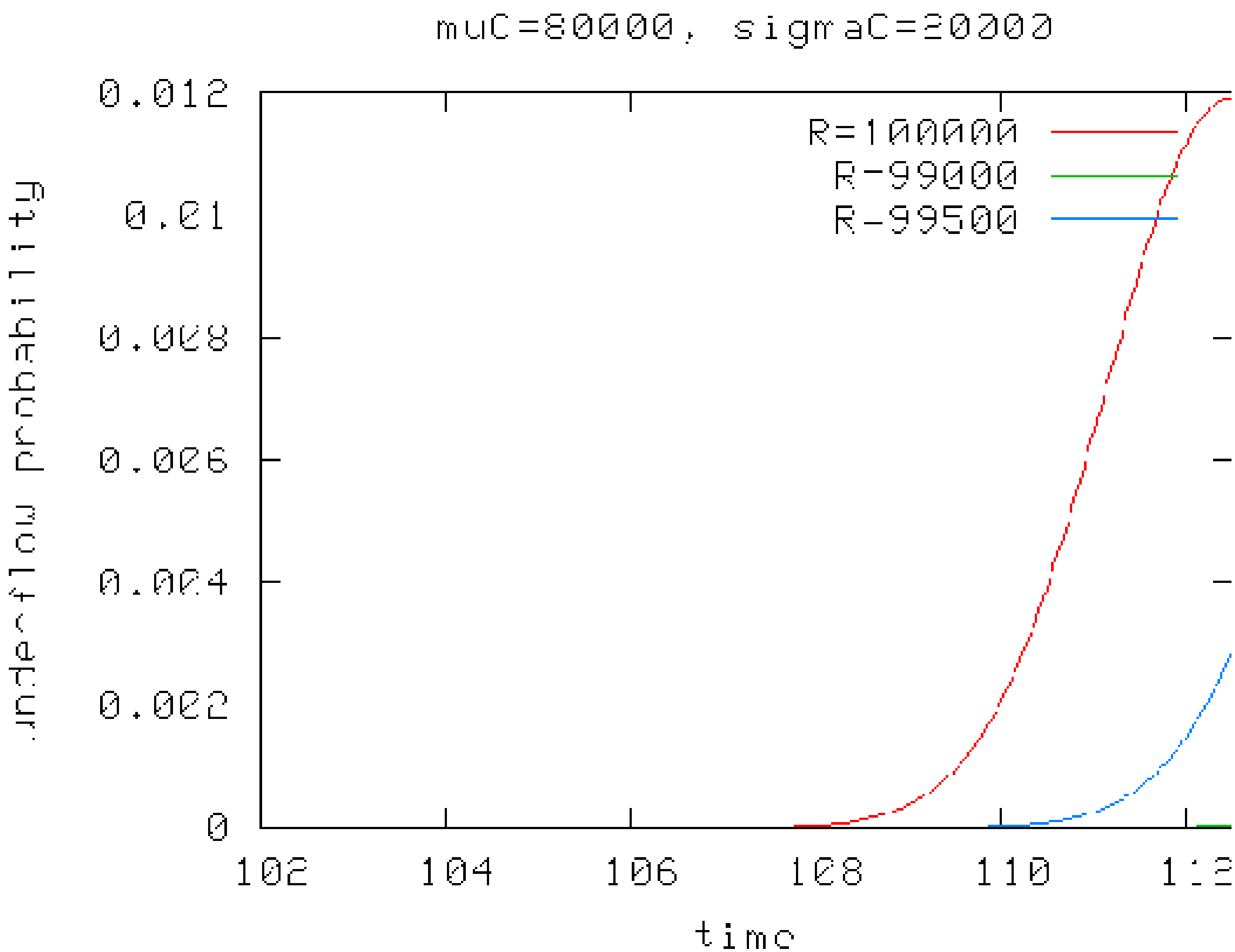
Effect of media rate



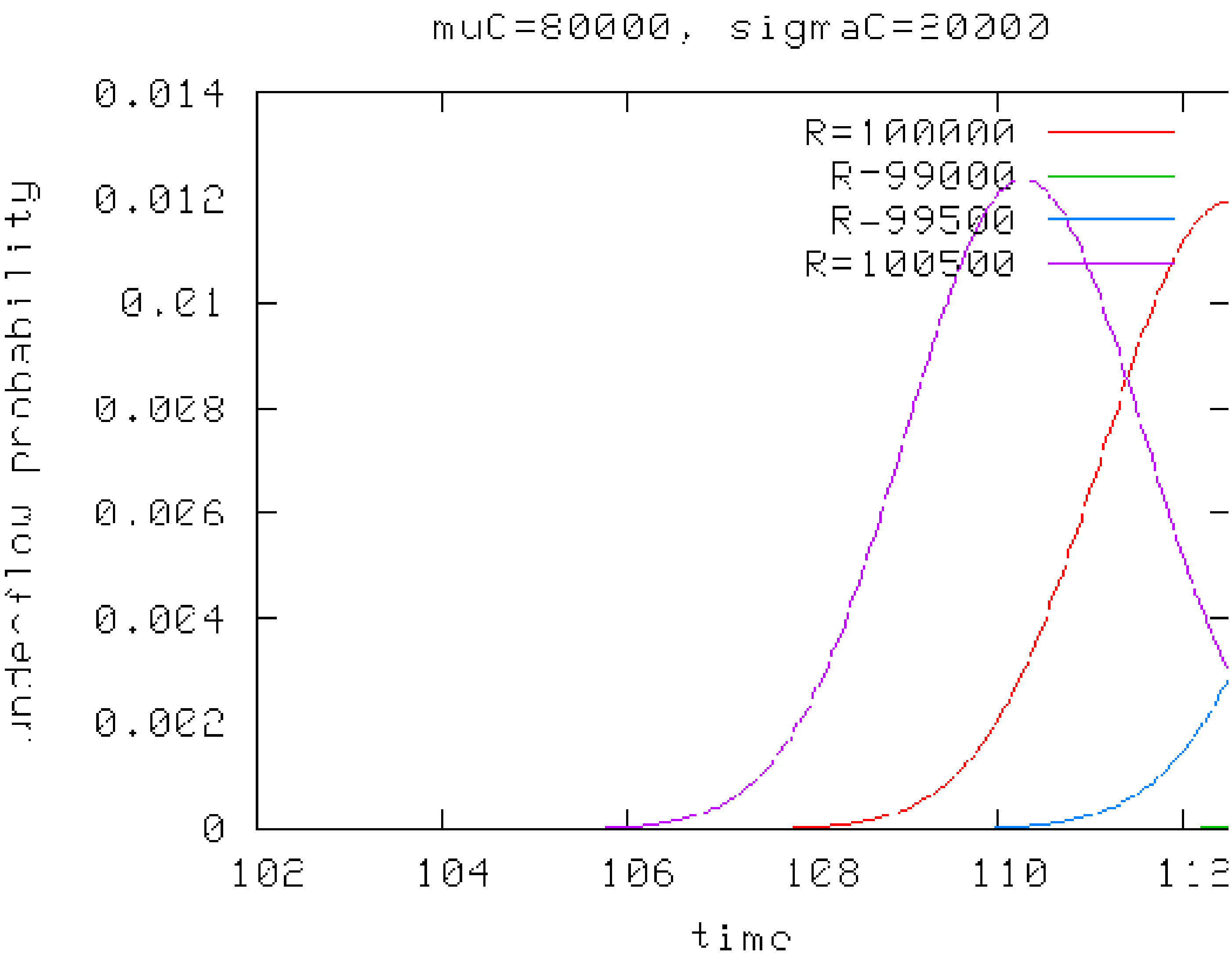
Effect of media rate



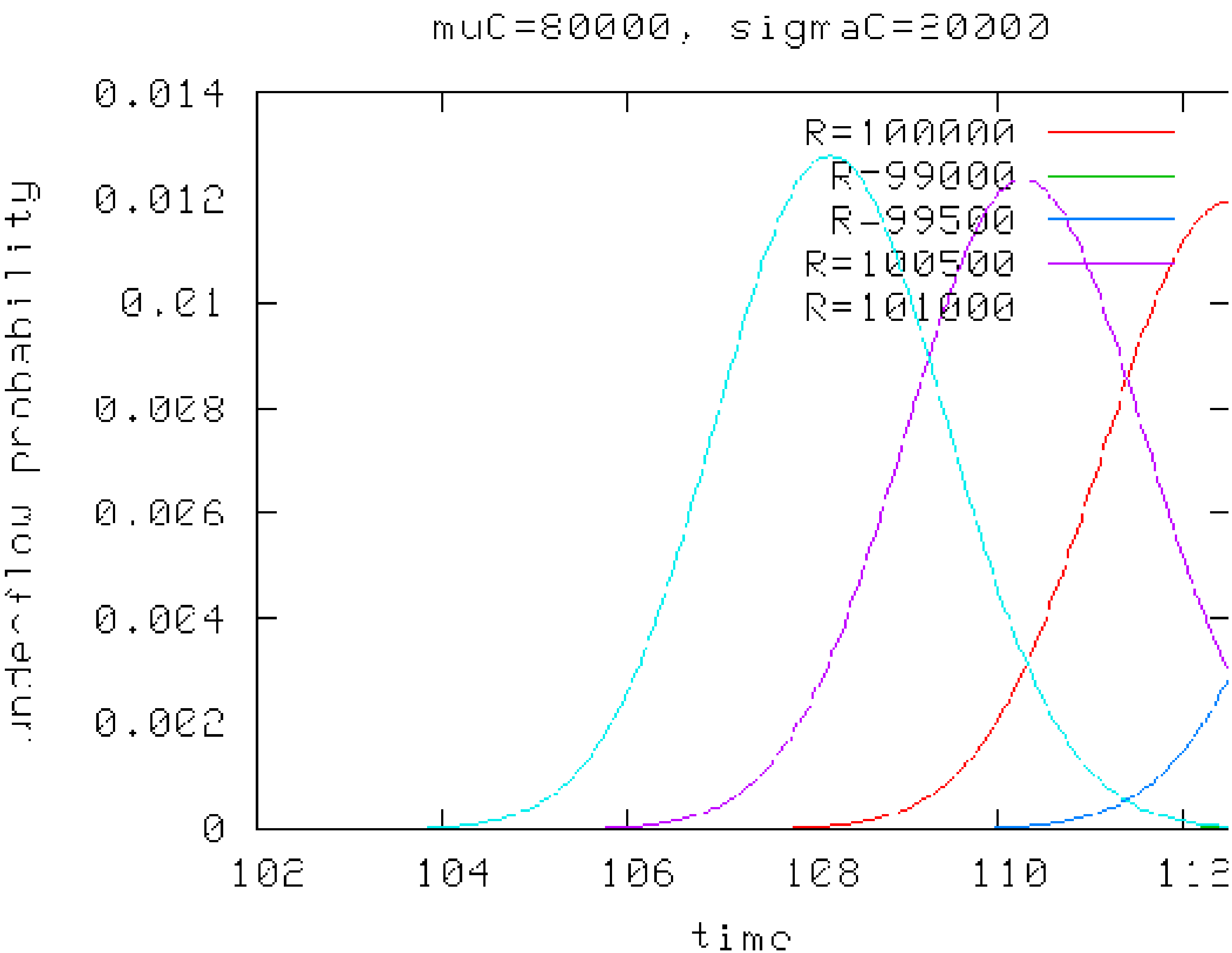
Effect of media rate



Effect of media rate



Effect of media rate



Effect of media rate

- Whenever a change occurs in the channel throughput, the portion of distribution within the playout interval may increase.
- To lower the probability of buffer underflows, it is necessary to modify the media rate;
  - decreasing the media rate, the distribution is shifted on the right and the portion inside the playout interval is decreased.

Proposed algorithm

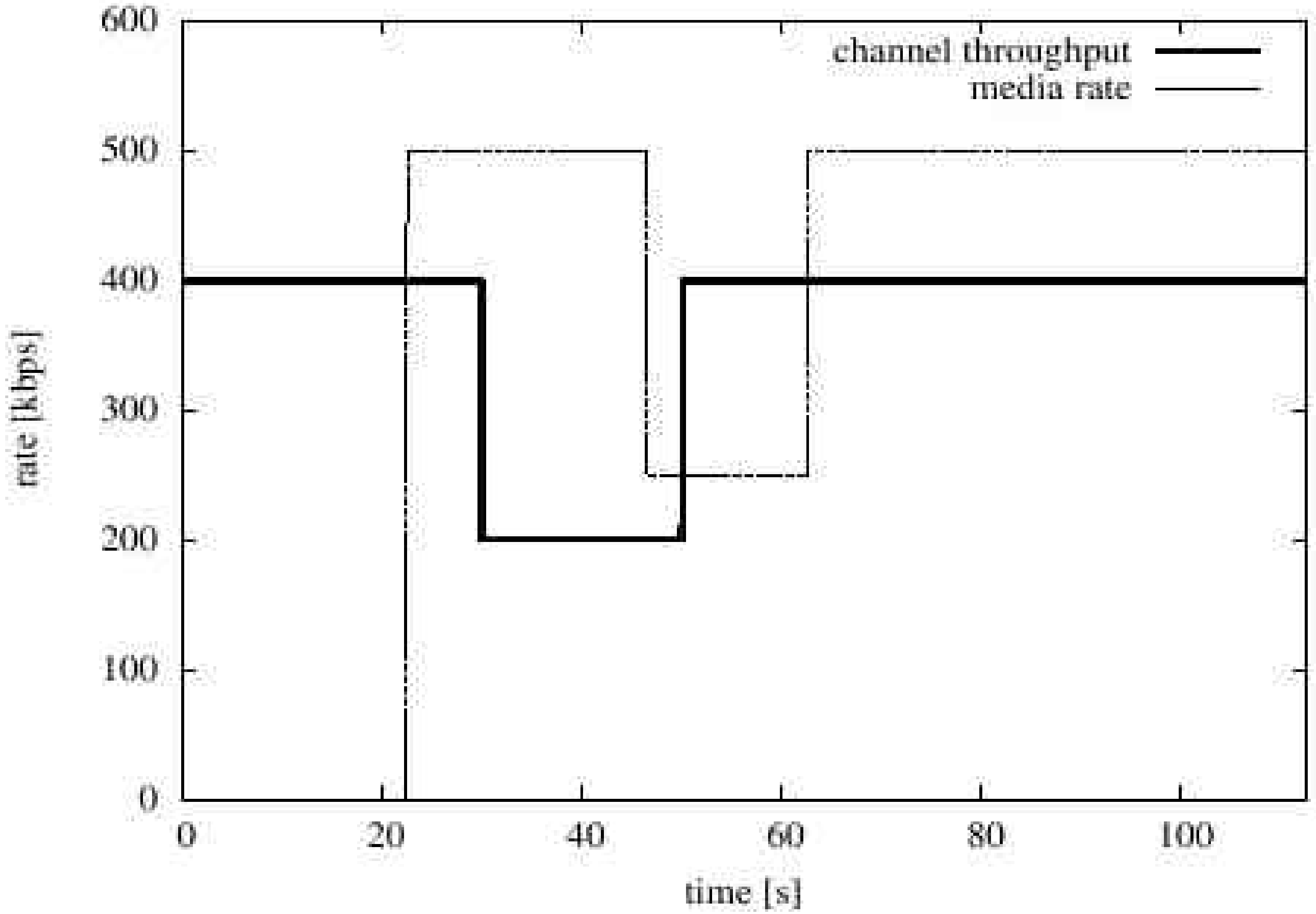
- In this work we consider the case of a piecewise constant channel.
- The pre-roll time is computed according to the initial channel rate value.
- Whenever a channel throughput variation occurs, the rate is recomputed according to the formula:

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

Proposed algorithm

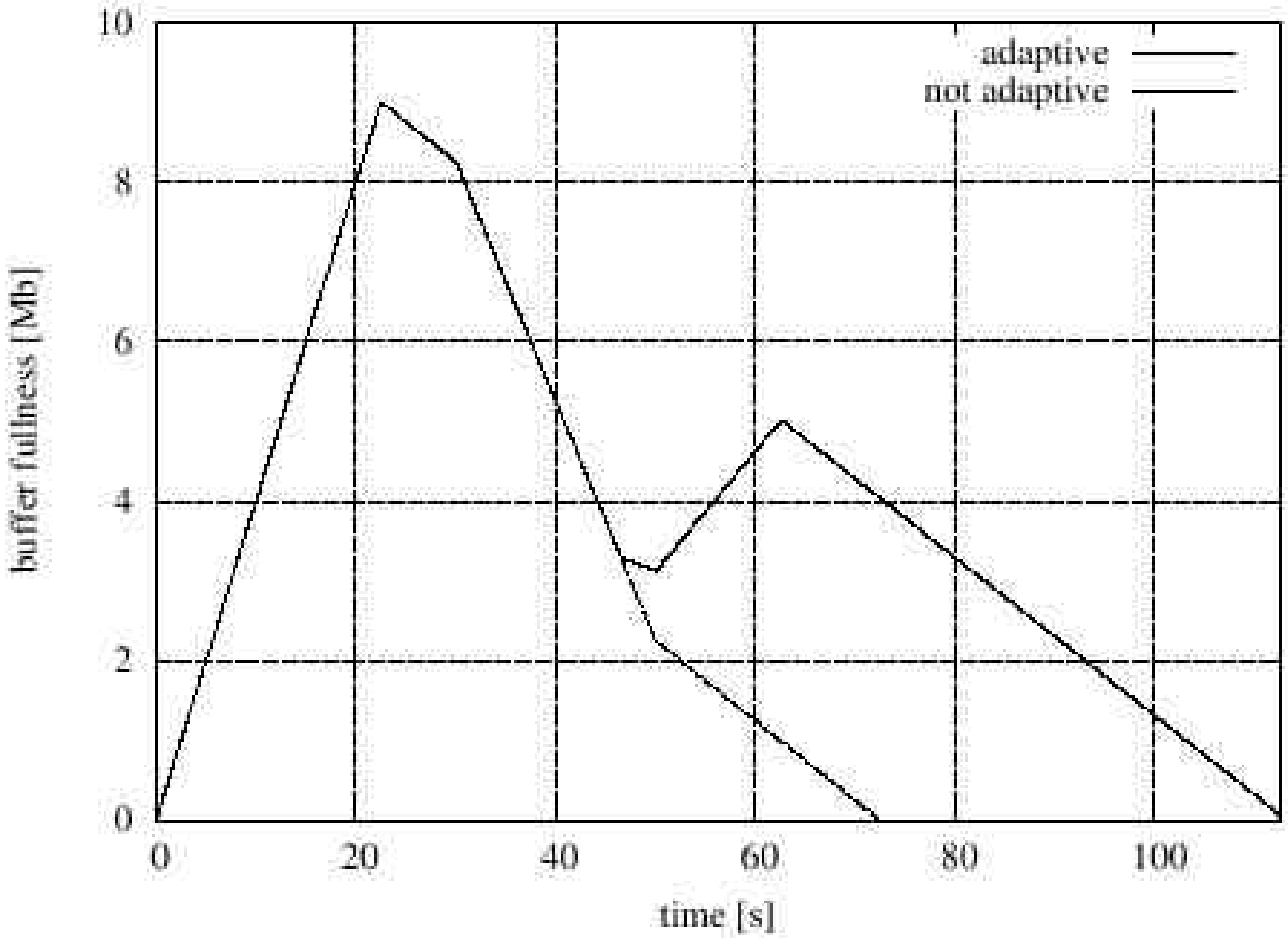
- This formula is a special case of the previous formulation.
- Working with zero variance, the underflow probability distribution is a delta function;
  - the initial pre-roll computation sets the center of the delta exactly at the end of the playout;
  - the re-computation formula places the delta again at the end of the playout, if a channel variation made it enter the playout interval.

Results



channel throughput
media rate
rate [kbps]
time [s]
0
100
200
300
400
500
600
0
20
40
60
80
100

Results



adaptive
not adaptive
buffer fullness [Mb]
10
8
6
4
2
0
0
20
40
60
80
100
time [s]

Conclusions

- It is possible to derive an approximate expression of the buffer underflow probability.
- The formulation can be used to determine at which rate to transmit in order to obtain a bounded buffer underflow probability.
- In addition, the proposed rate re-computation has a negligible complexity.

Thank you for your attention!