

# dPAM: A Distributed Prefetching Protocol for Scalable Asynchronous Multicast in P2P Systems

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**Abstract**—We leverage the buffering capabilities of end-systems to achieve scalable, asynchronous delivery of streams in a peer-to-peer environment. Unlike existing cache-and-relay schemes, we propose a distributed *prefetching* protocol where peers prefetch and store portions of the streaming media ahead of their playout time, thus not only turning themselves to possible sources for other peers but their prefetched data can allow them to overcome the departure of their source-peer. This stands in sharp contrast to existing cache-and-relay schemes where the departure of the source-peer forces its peer children to go the original server, thus disrupting their service and increasing server and network load. Through mathematical analysis and simulations, we show the effectiveness of maintaining such asynchronous multicasts from several source-peers to other children peers, and the efficacy of prefetching in the face of peer departures. We confirm the scalability of our dPAM protocol as it is shown to significantly reduce server load.

**Keywords:** Streaming; Peer-to-Peer Systems; Probabilistic Analysis; Simulations.

## I. INTRODUCTION

**Motivating Application:** In a large-scale peer-to-peer (P2P) network community, any node in the system may become the source of an interesting live, real-time feed that is (or may quickly become) of interest to a large number of nodes. For example, a P2P node may witness an interesting phenomenon or event, e.g., capturing a video feed from a webcam in a disaster scene, or capturing an interesting clip from a live event—say a super bowl entertainment “mishap.” In such a setting, it is unrealistic to assume that all requests for such a feed will arrive synchronously. Rather, it is likely that such requests will be distributed over time, with each node interested in receiving the entire feed (or a prefix thereof). Clearly, directing all such requests for the content to the “source” of the feed (which we would term as the “server” of the content) is neither scalable nor

practical. Also, using asynchronous multicast approaches requiring multicast capabilities (e.g., periodic broadcasts [24], [12], [14]) is not practical. For starters, the server may not even realize that the feed it is sharing with its P2P community is popular enough that it is “worth” multicasting! Finally, assuming that every node in the system is capable (or willing) to store the entire feed for future access by other nodes is not warranted since nodes may have limited storage capacities and/or nodes may opt to arbitrarily leave the P2P system. Rather, what is needed is a scalable protocol for the unicast dissemination of such content so that it is available on-demand to any P2P node requesting it, with minimal assumptions about the resources made available to such a protocol by the constituent nodes in the system. Specifically, it is prudent to assume that a node in the system is willing to contribute its limited storage and communication capacity as long as it is interested in receiving the content, *but not beyond*.

**Leveraging Local Storage for Scalable Asynchronous Multicast in P2P Systems:** Recently Jin and Bestavros proposed a scalable “cache-and-relay” approach [17] that could be used for scenarios similar to the one motivated above. Using this approach, a recipient of the feed would “cache” the most recently played out portion of the feed (after playing it out). Such cached content could then be used by other nodes in the system who request the feed within some bounded delay. This process of caching and relaying the content was shown to scale well in terms of server as well as network loads. In [6], a detailed analysis of this approach was presented.

There are two problematic aspects of the cache-and-relay approach. First, when a node leaves the system, any other nodes receiving the feed from that node are disconnected. This means that such nodes will experience a disruption in service. Second, to resume, such disconnected nodes must be treated as new arrivals, which in turn presents added load to the server (and network). This latter issue is especially significant because recent results by Jin and Bestavros [16] have shown that asynchronous multicast techniques do not scale as advertised when

content is not accessed from beginning to end (e.g., due to nodes prematurely leaving the multicast and/or when non-sequential access is allowed to support VCR functionality). Specifically, Jin and Bestavros showed that techniques that ensured asymptotic logarithmic server scalability under a sequential access model would in effect behave polynomially under non-sequential access models (e.g., in the presence of premature departures and/or jumps).

**Contributions:** In this paper we show that a more effective use of the local storage at P2P nodes (for purposes of asynchronous multicast) must involve prefetching. In particular, rather than caching content already played out, a node is also allowed to cache content that will be played out in the future. Such prefetching is possible if we assume that a node can retrieve content at a rate higher (even if by a small factor) than the playout rate. Such an assumption is common (and realistic) [24], [12], [14]. This “lookahead” buffering capability provides each node with an opportunity to recover from the premature departures of its source. Not only does this allow the node to avoid a disruption of its playout, but also it allows the node to resume the reception of the feed from sources other than the server—thus reducing the load on the server and improving scalability. In this paper, in addition to presenting the algorithmic underpinnings of our protocol, we also provide a complete modeling and analysis of its scalability under a workload with independent arrivals and departures. Our analysis is backed up by extensive simulations, which demonstrate the superiority of dPAM when compared to the cache-and-relay approach studied in [17], [6].

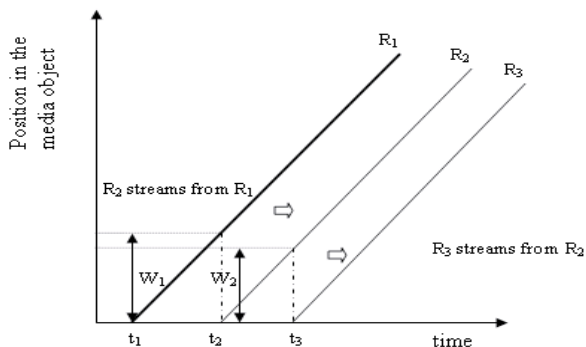


Fig. 1. Overlay-based asynchronous streaming

## II. PRE-FETCH-AND-RELAY: DETAILS

In this section, we present the *Pre-fetch-and-relay* strategy for asynchronous delivery of streams through overlay networks. We illustrate the *Pre-fetch-and-relay* strategy using Figures 1 and 2. Assume that each client is able to buffer the streamed content for a certain amount of time after playback by overwriting its buffer in a circular manner. As shown in Fig. 1,  $R_1$  has enough

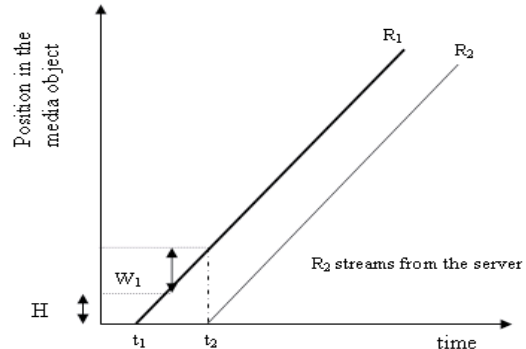


Fig. 2. Overlay-based asynchronous streaming

buffer to store content for time length  $W_1$ ; i.e. the data cached in the buffer is replaced by fresh data after an interval of  $W_1$  time units. When the request  $R_2$  arrives at time  $t = t_2$ , the content that  $R_2$  wants to download is available in  $R_1$ 's buffer and, hence,  $R_2$  starts streaming from  $R_1$  instead of going to the server. Similarly,  $R_3$  streams from  $R_2$  instead of the server. Thus, in Fig. 1, leveraging the caches at end-hosts helps to serve three clients using just one stream from the server.

In Fig. 2, by the time request  $R_2$  arrives, part of the content that it wants to download is missing from  $R_1$ 's buffer. This missing content is shown as  $H$  in Fig. 2. If the download rate is the same as the playout rate, then  $R_2$  has no option but to download from the server. However, if the network (total) download rate is greater than the playback rate, then  $R_2$  can open two simultaneous streams - one from  $R_1$  and the other from the server. It can start downloading from  $R_1$  at the playback rate (assuming that  $R_1$ 's buffer is being overwritten at the playback rate<sup>1</sup>) and obtain the content  $H$  from the server. After it has finished downloading  $H$  from the server, it can terminate its stream from the server and continue downloading from  $R_1$ . This stream patching technique to reduce server bandwidth was proposed in [13]. Assuming a total download rate of  $\alpha$  bytes/second and a playback rate of 1 byte/second, the download rate of the stream from the server should be  $\alpha - 1$  bytes/second. Hence, for this technique to work  $\alpha - 1 \geq 1 \Rightarrow \alpha \geq 2$ . Hence, we need the total download rate to be at least twice the playback rate for stream patching to work for a new arrival.

In the event that a client departs from the peer-to-peer network, all the clients downloading from the buffer of the departing client will have to switch their streaming session either to some other client or the server. The stream patching technique can be used by a client in this situation as well to avoid downloading from the server. As shown in Section II-C, unlike the case for a new

<sup>1</sup>We discuss the condition under which  $R_1$ 's buffer will be refreshed at the playout rate instead of the download rate in Section III.

arrival, the stream patching technique may work in this situation even when the total download rate is less than twice the playout rate, i.e.  $\alpha < 2$ .

When the download rate is greater than the playout

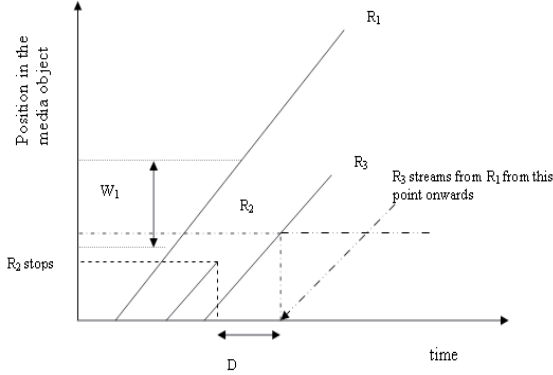


Fig. 3. Delay in finding a new download source

rate, a client can *pre-fetch* content to its buffer before it is time to playout that content. Pre-fetching content can help achieve a better playout quality in overlay multicast. In a realistic setting, there would be a certain delay involved in searching for a peer to download from; for example, consider the situation depicted in Fig. 3.  $R_3$  starts streaming from  $R_2$  on arrival. After  $R_2$  departs, as shown in Fig. 3, it takes  $R_3$   $D$  seconds (time units) to discover the new source of download  $R_1$ . If the pre-fetched “future” content in  $R_3$ 's buffer, at the time of  $R_2$ 's departure, requires more than  $D$  seconds (time units) to playout (i.e. the size of the future content is greater than  $D$  bytes, assuming a playout rate of 1 byte/second) then the playout at  $R_3$  does not suffer any disruption on  $R_2$ 's departure. If the size of the “future” content is smaller than  $D$  bytes, then  $R_3$  will have to open a stream from the server, after it has finished playing out its pre-fetched content, till it discovers  $R_1$ . In a *Cache-and-relay* strategy, clients do not pre-fetch content<sup>2</sup>. Thus, in the case of *Cache-and-relay*,  $R_3$  will have to open a stream from the server as soon as it realizes that  $R_2$  has departed and continue downloading from the server for  $D$  seconds (till it discovers that it can download from  $R_1$ ).  $R_3$  cannot know *a priori* when  $R_2$  is going to depart. Due to the delays involved in opening a stream from the server, it is quite likely that the playout at  $R_3$  would be disrupted on  $R_2$ 's departure in the case of *Cache-and-relay*. In the case of *Pre-fetch-and-relay*, if the time required to playout the pre-fetched content is larger than the delay involved in finding a new source to download from, the playout at  $R_3$  would not be disrupted upon  $R_2$ 's departure from the peer-to-peer network. Pre-fetching content is also advantageous when the download rate is variable. A client can absorb a temporary degradation

<sup>2</sup>It can be due to the fact that the playout rate is equal to the download rate or clients may choose not to pre-fetch content.

in download rate without affecting the playout quality if it has sufficient pre-fetched content in its buffer.

#### A. Control Parameters

In this paper, we analyze the importance and effect of the following three parameters in achieving scalable (in terms of server bandwidth), asynchronous delivery of streams in a peer-to-peer environment through analysis and simulations.

$$1) \alpha = \frac{\text{Download rate}}{\text{Playout rate}}$$

Without loss of generality, we take the *Playout rate* to be equal to 1 byte/second and, hence the *Download rate* becomes  $\alpha$  bytes/second. We assume  $\alpha > 1$ .

$$2) T_b : \text{The time it takes to fill the buffer available at a client at the download rate.}$$

The actual buffer size at a client is, hence,  $\alpha \times T_b$  bytes. The available buffer size at a client limits the time for which a client can download the stream at a rate higher than the playout rate.

$$3) \beta = \frac{\text{Future Content}}{\text{Past Content}}$$

$\beta$  represents the ratio of the content yet to be played out, “future content”, to the content already played out, “past content”, in the buffer.

Next, we discuss the constraints, in terms of  $\alpha$ ,  $\beta$  and  $T_b$ , that must be satisfied for a client to be able to download the stream from the buffer of another client available in the peer-to-peer network.

#### B. Constraints in the case of an arrival

For a new arrival  $R_0$  to be able to download from the buffer of  $R_1$ , the inter-arrival time between  $R_0$  and  $R_1$  should be less than  $T_b$ . If  $R_0$  arrives more than  $T_b$  time units after  $R_1$ , then part of the content that  $R_0$  wants to download would have been over-written in  $R_1$ 's buffer. If  $\alpha < 2$ , then in such a situation  $R_0$  has no option but to stream from the server. Hence, if  $\alpha < 2$ , a new arrival at time  $t = t_0$  can stream from only those clients that arrived during the interval  $T_D = [t_0 - T_b, t_0)$ . If  $\alpha \geq 2$  and  $R_1$  is over-writing the content in its buffer at the playout rate, then  $R_0$  can take advantage of the higher download rate (compared to the playout rate) and the stream patching technique [13] to possibly avoid a complete streaming from the server and instead download from  $R_1$  and only patch the missing content from the server. It is easy to verify that the size of the missing content that  $R_0$  needs to download from the server, in order to be able to stream from  $R_1$ 's buffer, cannot be greater than the size of the available buffer at  $R_0$ , which is  $\alpha \times T_b$  in our model. The following theorem sums up the above discussion:

**Theorem 1:** A newly arrived client  $R_0$  can download from the buffer of  $R_1$  if the following conditions are satisfied:

- The inter-arrival time between  $R_0$  and  $R_1$  is less than  $T_b$ , or
- If the inter-arrival time between  $R_0$  and  $R_1$  is greater than  $T_b$ , then  $\alpha$  should be greater than or equal to 2,  $R_1$  must be over-writing the content in its buffer at the playout rate and the size of the content missing from  $R_1$ 's buffer should be less than or equal to  $\alpha \times T_b$ .

The first condition ensures that the content needed by  $R_0$  is present in  $R_1$ 's buffer. The second condition defines the scenario in which the stream patching technique can be used by  $R_0$ .

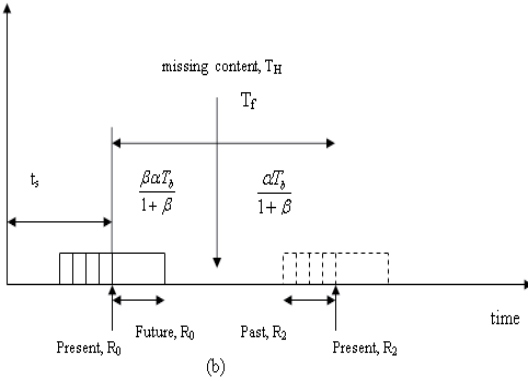
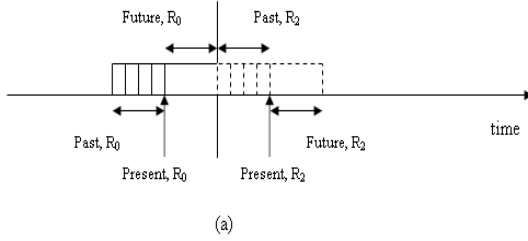


Fig. 4. Buffers of  $R_0$  and  $R_2$

### C. Constraints in the event of a departure

Let us assume that  $R_0$  was streaming from  $R_1$ 's buffer.  $R_1$  leaves the peer-to-peer network at time  $t = t_d$ . If the available buffer size at  $R_0$  is  $\alpha \times T_b$  bytes and at  $t = t_d$ , the ratio of “future” content to “past” content in  $R_0$ 's buffer is  $\beta$ , then  $R_0$  has  $\left(\frac{\beta \alpha T_b}{1+\beta}\right)$  bytes of the “future” content and  $\frac{\alpha T_b}{1+\beta}$  bytes of the “past” content in its buffer. At a *Playout rate* of 1 byte/time unit,  $R_0$  has  $\left(\frac{\beta \alpha T_b}{1+\beta}\right)$  time units to find a new source to download from after  $R_1$  departs. If  $\alpha = 1$ , then after  $R_1$ 's departure,  $R_0$  can download from another client  $R_2$ 's buffer if and only if the content in their buffers overlaps (partially). Fig. 4(a) shows the situation when the buffers of  $R_0$  and  $R_2$  are contiguous. Any client that is ahead of  $R_0$ , in terms of playing out the stream, would have some content that  $R_0$  needs to download missing from its buffer and hence, unsuitable for  $R_0$  to download from. Fig. 4(b) depicts such a situation.

Consider the situation depicted in Fig. 4(b). Let us assume that the ratio of “future” content to “past” content in  $R_0$ 's buffer is  $\beta$  and hence, it currently has  $\left(\frac{\beta \alpha T_b}{1+\beta}\right)$  bytes of pre-fetched data. Assume that the “missing” content is  $T_H$  bytes and that the playout rate is 1 byte/second. If  $\alpha > 1$ , then  $R_0$  can open two simultaneous streams, one from the server and the other from  $R_2$ , and terminate its stream from the server after it has downloaded the “missing” content and continue to download from  $R_2$  thereafter. Note that for this stream patching technique to work,  $R_2$  should be over-writing contents in its buffer at a rate less than  $\alpha$ ; in our model we assume that clients over-write the content in their buffer either at the download rate ( $\alpha$ ) or at the playout rate. Hence, in this case,  $R_2$  should be over-writing its buffer at the rate of 1 byte/second. If this is the case, then  $R_0$  can download from  $R_2$  at the playout rate of 1 byte/second and download the “missing” content from the server at the rate of  $(\alpha - 1)$  bytes/second.

The following constraints must be satisfied by the size of the “missing” content,  $T_H$  bytes, for  $R_0$  to able to stream from  $R_2$ 's buffer:

#### 1) Constraint imposed due to $\alpha$ :

The time taken by  $R_0$  to playout the pre-fetched content in its buffer and the “missing” content,  $T_H$  bytes, is equal to  $(\alpha \times T_b) \left(\frac{\beta}{1+\beta}\right) + T_H$  seconds (at the playout rate of 1 byte/second). The total time needed by  $R_0$  to download  $T_H$  bytes from the server at the rate of  $(\alpha - 1)$  bytes/second is  $\frac{T_H}{\alpha - 1}$  seconds. In order to have the “missing” content available at  $R_0$  before its playout time, the following inequality must be satisfied:

$$(\alpha \times T_b) \left(\frac{\beta}{1+\beta}\right) + T_H \geq \frac{T_H}{\alpha - 1} \quad (1)$$

The above inequality demands that the time taken to playout the pre-fetched and the “missing” content should be greater than the time taken to download the “missing” content. For example, consider the situation when  $\alpha = 1.2$  bytes/second, the playout rate = 1 byte/second and  $T_H = 1$  byte. Then the time needed to download the “missing” content from the server is 5 seconds. In this case if  $R_0$  does not have 5 bytes of pre-fetched data then the playout at  $R_0$  would be disrupted due to the fact that it would already be time to playout a portion of the “missing” content before it has been downloaded from the server. Note that if  $\alpha \geq 2$ , then the condition (1) is always satisfied. The stream patching can be used in the case of a departure even when  $1 < \alpha < 2$  if a client has sufficient pre-fetched content.

#### 2) Constraint imposed by the size of the buffer:

Suppose that  $R_0$  starts downloading simultaneously from the server and  $R_2$  at time  $t = t_d$ . Since it is downloading from  $R_2$  at the playout rate, at any

time instant the size of data downloaded from  $R_2$  is exactly equal to the size of the data played out by  $R_2$  after  $t_d$ . Thus, to store the content downloaded from the server,  $R_0$  will have to over-write the “past” content in its buffer. Hence,  $T_H$  cannot be greater than the size of the “past” content in  $R_0$ ’s buffer at  $t_d$ . By our assumption, the “past” content in  $R_0$ ’s buffer at  $t_d$  is  $\left(\frac{\alpha T_b}{1+\beta}\right)$  bytes and hence,

$$T_H \leq \frac{\alpha T_b}{1 + \beta} \quad (2)$$

### III. SERVER BANDWIDTH REQUIREMENT: ANALYSIS

We consider the case of a single CBR media distribution. The playback rate is assumed to be 1 byte/time unit. The client requests are generated according to a Poisson process with rate  $\lambda$ . The time spent by a client downloading the stream is exponentially distributed with rate  $\mu$ .

Let us assume that a client is able to determine whether it should download the stream from the server or from the buffer of some other client without any delays both in the case of a new arrival as well as in the event of a departure; i.e. we do not take delays like propagation delay and the delay involved in searching the peer-to-peer network for a suitable client to download from into consideration. We later incorporate such delays in Section III-D.

#### A. Arrivals

A new arrival,  $R_0$ , would have to download from the server in either of the following two cases:

- The inter-arrival time between  $R_0$  and the arrival immediately preceding  $R_0$ , say  $R_1$ , is greater than  $W$ ; where  $W = T_b$  if  $1 < \alpha < 2$  or  $W = (\alpha \times T_b) \left(\frac{2+\beta}{1+\beta}\right)$  if  $\alpha \geq 2$ . As mentioned in Theorem 1, Section II-B, if  $\alpha \geq 2$ ,  $R_0$  can use the stream-patching technique to “catch-up” with  $R_1$  iff the size of the content “missing” from  $R_1$ ’s buffer is less than or equal to  $\alpha \times T_b$ . If  $R_1$  maintains the ratio  $\beta$  in its buffer, then the maximum value of  $W$  for  $R_0$  is

$$\begin{aligned} W &= \text{max. “missing” content} \\ &\quad + \text{“past” content at } R_1 \\ &= (\alpha \times T_b) + \left(\frac{\alpha T_b}{1 + \beta}\right) \\ &= (\alpha \times T_b) \left(\frac{2 + \beta}{1 + \beta}\right) \end{aligned}$$

- Suppose  $R_0$  arrives at time  $t = t_0$ . It can be easily verified that as a consequence of Theorem 1, Section II-B,  $R_0$  can download from only those users that arrived during the interval  $T_D = [t_0 - W, t_0)$ . If all the users that arrived during the interval  $T_D$  have already departed from the peer-to-peer network by

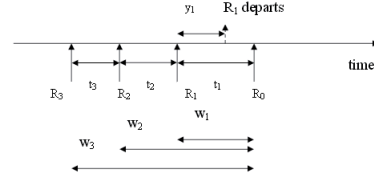


Fig. 5. Arrival of  $R_0$

$t = t_0$ , then  $R_0$  would have to download from the server.

Let  $w$  represent the inter-arrival time between any two client requests. Since the arrival process is Poisson with rate  $\lambda$ , the inter-arrival time is exponentially distributed with mean  $\frac{1}{\lambda}$ , hence,

$$P\{w > W\} = e^{-\lambda W} \quad (3)$$

Suppose a new client request,  $R_0$ , arrives at time  $t = t_0$ . Let  $N$  represent the number of arrivals in the interval  $T_D = [t_0 - W, t_0)$ . Let  $N = n$ . As mentioned earlier,  $R_0$  would have to download from the server if all the  $n$  users that arrived during the interval  $T_D$  have departed by  $t = t_0$ . Let  $R_i$ ,  $i = 1, \dots, n$ , represent the  $n$  users that arrived during the interval  $T_D$ . Let  $y_i$  be the time spent by client  $R_i$  downloading the stream before it departs. Let the inter-arrival time between user  $R_i$  and  $R_0$  be  $w_i$ . If  $y_i \leq w_i$ ,  $R_i$  would have departed by the time  $R_0$  arrives (see Fig. 5). Let **A** represent the event that  $R_0$  has to download from the server because all the users that arrived during the interval  $T_D$  departed before  $R_0$ ’s arrival. Then

$$\begin{aligned} P\{\text{Event A} | N = n, y_i, w_i\} &= \\ P\{N = n\} \prod_{i=1}^n P\{y_i \leq w_i\} \times P\{w_i\} \end{aligned}$$

where  $P\{w_i\}$  is the probability that the inter-arrival time between  $R_i$  and  $R_0$  is  $w_i$ .

Let the inter-arrival time between  $R_i$  and  $R_{i-1}$  be  $t_i$  with  $t_1$  being the inter-arrival time between  $R_1$  and  $R_0$  and hence,  $t_1 = w_1$ . Note that because of the memoryless nature of the Poisson arrival process, each  $t_i$  is exponentially distributed with mean  $\frac{1}{\lambda}$  time units. Now,

$$\begin{aligned} w_2 &= t_1 + t_2 \\ w_3 &= t_1 + t_2 + t_3 \\ &\vdots \\ w_n &= t_1 + t_2 + \dots + t_n \end{aligned}$$

Since each  $w_i$  ( $i \geq 2$ ) is a sum of  $i$  i.i.d. exponential random variables with mean  $\frac{1}{\lambda}$ ,  $w_i$  is an Erlang random variable of order  $i$  with mean  $\frac{i}{\lambda}$ . Hence,

$$P\{w_i\} = \frac{\lambda(\lambda w_i)^{i-1} e^{-\lambda w_i}}{(i-1)!}.$$

Since  $w_i \in [0, W) \forall i = 1, \dots, n$ ,

$$P\{\text{Event A}\} = \sum_{n=1}^{\infty} \int_0^W \dots \int_0^W (\text{Integrand}) dw_1 \dots dw_n$$

where the *Integrand* is

$$P\{N = n\} \prod_{i=1}^n P\{y_i \leq w_i\} P\{w_i\}$$

To keep the analysis tractable, we solve the above equation for  $N = 1$ . Then,

$$\begin{aligned} P\{\text{Event A}\} &= \int_0^W P\{N = 1\} \times P\{y_1 \leq w_1\} \times P\{w_1\} dw_1 \\ &= \int_0^W \lambda W e^{-\lambda W} \times (1 - e^{-\mu w_1}) \times \lambda e^{-\lambda w_1} dw_1 \\ &= \lambda W e^{-\lambda W} \times \left( \frac{\mu}{\lambda + \mu} + \frac{\lambda e^{-(\lambda + \mu)W}}{\lambda + \mu} - e^{-\lambda W} \right) \end{aligned} \quad (4)$$

Thus,

$$\begin{aligned} P\{\text{a new arrival goes to the server}\} \\ = P\{w > W\} + P\{\text{Event A}\} \end{aligned}$$

where  $P\{w > W\}$  and  $P\{\text{Event A}\}$  are given by equations (3) and (4), respectively.

### B. Departures

Suppose user  $R_0$  is downloading the stream from the buffer of user  $R_1$ . Let  $R_1$  depart from the peer-to-peer network at time  $t = t_d$  such that by this time  $R_0$  has been downloading and playing out the stream for a duration of  $t_s$  time units; i.e.  $R_0$  has been in the peer-to-peer network for  $t_s$  time units when  $R_1$  departs. We assume that  $t_s$  is long enough for  $R_0$  to have achieved the desired ratio,  $\beta$ , between its “future” and “past” content.

Fig. 4 presents a snapshot of buffer of  $R_0$  and another user,  $R_2$ , from whose buffer  $R_0$  can start downloading from instead of going to the server after  $R_1$  has departed. Since we have assumed a sequential access model for client requests, i.e. each client downloads the stream from the beginning and the playout speed is 1 byte/time unit, after  $R_0$  has spent  $t_s$  time units downloading the content, its “present” is  $t_s$  bytes away from the beginning of the content.

The difference, in terms of number of bytes, between the “present” of  $R_0$  and  $R_2$ , represented by  $T_f$  in Fig. 4(b), is

$$\begin{aligned} T_f &= \text{“future” content at } R_0 \\ &+ \text{“missing” content} + \text{“past” content at } R_2 \\ \implies T_f &= \left( \frac{\beta \alpha T_b}{1 + \beta} \right) + T_H + \left( \frac{\alpha T_b}{1 + \beta} \right) \end{aligned}$$

The size of the “future” content at  $R_0$  and the “past” content at  $R_2$  have been calculated using the assumption

that both  $R_0$  and  $R_2$  maintain the ratio  $\beta$ . As discussed in Section II-C the download rate  $\alpha$  and the buffer size impose certain constraints on the size of the “missing” content,  $T_H$ , for  $R_2$  to be suitable for  $R_0$  to download from. The maximum value of  $T_f$  is

$$T_f = \begin{cases} (\alpha \times T_b) \left( \frac{\beta - \alpha + 2}{(1 + \beta)(2 - \alpha)} \right) & \text{if } \alpha \leq \left( \frac{2 + \beta}{1 + \beta} \right) \\ (\alpha \times T_b) \left( \frac{2 + \beta}{1 + \beta} \right) & \text{otherwise} \end{cases} \quad (5)$$

Refer to the Appendix for a detailed derivation of (5).

After the departure of  $R_1$ , only those clients whose “present” is ahead of  $R_0$ ’s “present” by a value less than or equal to the maximum value of  $T_f$  are suitable for  $R_0$  to start downloading from, assuming that all such clients maintain the same ratio  $\beta$  in their buffers.

Since we have assumed the playout speed to be 1 byte/second, a client whose “present” is  $T_f$  bytes ahead of  $R_0$  must have arrived  $T_f$  time units before  $R_0$ . Hence, the constraint on the suitability of a client (mentioned in the preceding paragraph) can be re-stated as: *on  $R_1$ ’s departure  $R_0$  can download from only those clients that arrived at most  $T_f$  time units before  $R_0$ ; where the value of  $T_f$  is given by (5).*

Suppose the client request  $R_0$  arrives at  $t = t_0$ . Let

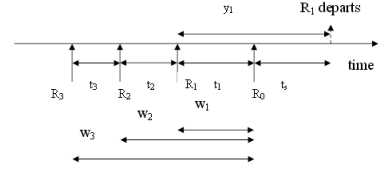


Fig. 6. Departure of  $R_1$

$N$  represent the number of arrivals in the interval  $[t_0 - T_f, t_0)$ . Let  $N = n$  and  $R_i, i = 1, \dots, n$  be the  $n$  client requests. Suppose  $R_0$  starts downloading the stream from  $R_1$  and  $R_1$  departs the peer-to-peer network at  $t_d = t_0 + t_s$ . Thus,  $R_0$  has spent  $t_s$  time units downloading the stream when  $R_1$  departs. Let  $w_i$  be the inter-arrival time between  $R_i$  and  $R_0$  and  $y_i$  be the time spent by  $R_i$  downloading the stream (see Fig. 6). If all the clients  $R_i, i = 2, \dots, n$  have departed by the time  $t_d = t_0 + t_s$ , i.e.  $y_i \leq t_s + w_i$ , then  $R_0$  would have no option but to download from the server on  $R_1$ ’s departure. Let event **B** represent the situation that  $R_0$  downloads from the server on  $R_1$ ’s departure. Then,

$$\begin{aligned} P\{\text{Event B} | N = n, t_s, y_i, w_i\} &= \\ P\{N = n\} P\{t_s\} \prod_{i=2}^n P\{w_i\} P\{y_i \leq t_s + w_i\} \end{aligned}$$

where  $P\{t_s\}$  is the probability that  $R_0$  has downloaded the stream for  $t_s$  time units by the time  $R_1$  departs,  $P\{w_i\}$  is the probability that the inter-arrival time between  $R_i$  and  $R_0$  is  $w_i$  and  $i = 2, \dots, n$ .

To keep the analysis tractable, let us assume  $N = 2$ ; this represents the scenario where  $R_0$  starts downloading from  $R_1$  on arrival and could potentially download from  $R_2$  on  $R_1$ 's departure. We have,

$$P\{\text{Event B}\} = \int_0^\infty \int_0^{T_f} (\text{Expression}) dw_2 dt_s \quad (6)$$

where *Expression*<sup>3</sup> is:

$$\left( \frac{e^{-\lambda T_f} (\lambda T_f)^2}{2!} \right) (\mu e^{-\mu t_s}) \left( \frac{\lambda (\lambda w_2) e^{-\lambda w_2}}{1!} \right) (1 - e^{-\mu(t_s + w_2)})$$

### C. Server Load

Let event **S** represent the situation that a client request downloads the stream from the server. From the preceding discussion in Section III-A and III-B,

$$P\{\text{Event S}\} = P\{w > W\} + P\{\text{Event A}\} + P\{\text{Event B}\}$$

where the right-hand terms are given by equations (3), (4) and (6), respectively.

In our model we have assumed that client requests are generated according to a Poisson process with rate  $\lambda$ . Hence, in steady state, i.e. after there are enough end-hosts in the peer-to-peer network, the average number of client requests that download the stream from the server is  $\lambda \times P\{\text{Event S}\}$ .

### D. Incorporating the delays

Let us revisit the scenario discussed in Section II-C. Suppose that  $R_2$  is a suitable client for  $R_0$  to start downloading from after  $R_1$ 's departure but it takes  $R_0$   $D$  time units after  $R_1$ 's departure to determine this.  $R_0$  needs to know what is available on each client's buffer and process that information to determine the suitability of each client. There will also be some propagation delay involved in the transmission of the *meta-data*<sup>4</sup> traffic amongst clients in the overlay network.

At the time of  $R_1$ 's departure,  $R_0$  has  $\left(\frac{\beta\alpha T_b}{1+\beta}\right)$  time units of "future" data. We assume that if  $R_0$  is not able to find  $R_2$  by the time it finishes playing out its "future" content ( $D > \frac{\beta\alpha T_b}{1+\beta}$ ),  $R_0$  starts downloading from the server. If  $D \leq \left(\frac{\beta\alpha T_b}{1+\beta}\right)$ ,  $R_0$  can absorb the delay without any disruption of its playout. In the next section, we present analytical results after incorporating the delay involved in the event of a departure into our model assuming that the delay is uniformly distributed.

<sup>3</sup>The integrand for computing  $P\{\text{Event B}\}$  is computed assuming that  $R_0$  starts downloading from  $R_1$  on arrival.

<sup>4</sup>The messages exchanged in determining the clients who are still present in the peer-to-peer network and their buffer contents.

## IV. PERFORMANCE EVALUATION

In this section, we evaluate the performance of *Pre-fetch-and-relay* based on the analytical model presented in this paper and compare it with simulation results. We also present analytical and simulation results after incorporating the various delays into our model (as discussed in Section III-D). We also compare the performance of *Pre-fetch-and-relay* with *Cache-and-relay* (CR) with respect to savings in server bandwidth. We refer to the protocols oStream [6] and OSMOSIS [17] by the generic term *Cache-and-relay* because they correspond to the situation when  $\alpha = 1$  (hence, a client cannot pre-fetch any content).

### A. Advantage of Pre-fetching

If a client does not *pre-fetch* content (as in *Cache-and-relay*), then on premature departure of its current source it has no option but to start downloading from the server till the time it is able to locate another source in the peer-to-peer network. In this section, we analytically compare the performance of our *Pre-fetch-and-relay* scheme against the *Cache-and-relay* scheme proposed in [6].

Suppose  $R_0$  starts downloading the stream from  $R_1$  upon arrival. Let the inter-arrival time between  $R_0$  and  $R_1$  be  $w_1$  and the time spent by  $R_i$  ( $i = 0, 1$ ) downloading the stream be  $t_i$ . Under the *Cache-and-relay* scheme,  $R_0$  will have to start downloading from the server at some point if  $R_1$  departs before  $R_0$ . In both *Cache-and-relay* as well as *Pre-fetch-and-relay*, a client that is forced to download from the server after the premature departure of its source can stop downloading from the server after it has found another source in the peer-to-peer network. To keep the analysis tractable, we have not considered this "optimization" in this analysis. But since this "optimization" technique to reduce the server bandwidth is possible under both schemes, the trend exhibited by the results presented in this section would remain the same. Using the same analytical model of Section III,

$$\begin{aligned} &P\{R_1 \text{ departs before } R_0\} \\ &= P\{t_1 \leq t_0 + w_1\} \times P\{t_0\} \times P\{w_1\} \\ &= (1 - e^{-\mu(t_0 + w_1)}) \times \mu e^{-\mu t_0} \times \lambda e^{-\lambda w_1} \end{aligned}$$

Thus, the probability that  $R_0$  will have to download from the server under the *Cache-and-relay* scheme due to the premature departure of its current source, represented by  $P_d$ , is

$$P_d = \int_0^W \int_0^\infty (1 - e^{-\mu(t_0 + w_1)}) \times \mu e^{-\mu t_0} \times \lambda e^{-\lambda w_1} dt_0 dw_1$$

In the *Pre-fetch-and-relay* scheme, as calculated in Section III, the probability that  $R_0$  downloads from the server after the premature departure of its current source is given by  $P\{\text{Event B}\}$ . Note that in computing  $P\{\text{Event B}\}$  we had not taken the various delays mentioned in Section III-D, into consideration. Let  $p$

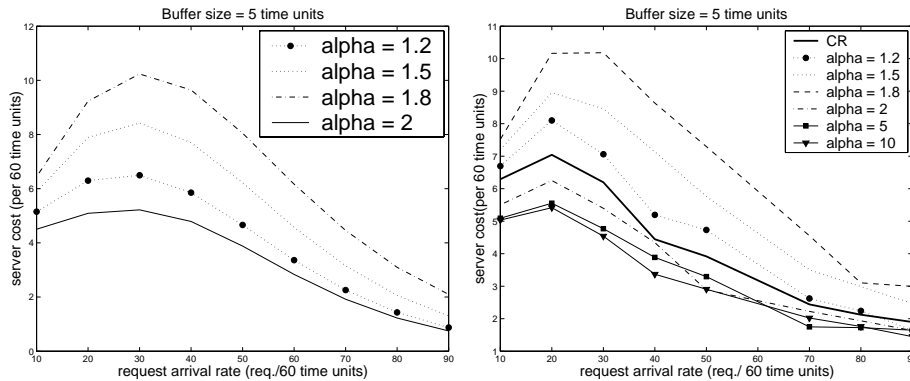


Fig. 8. (left) Analysis; (right) Simulation. Mean download time = 1000 time units, buffer size = 5 bytes

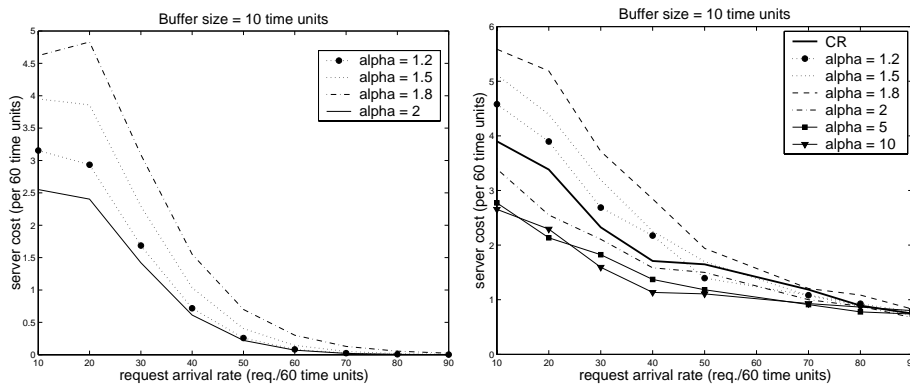


Fig. 9. (left) Analysis; (right) Simulation. Mean download time = 1000 time units, buffer size = 10 bytes

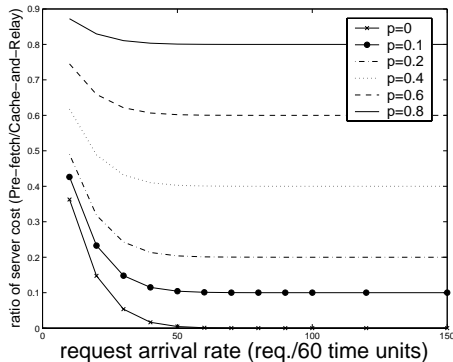


Fig. 7. Importance of Pre-fetched content

represent the probability that  $R_0$  is not able to locate a new source to download from, after  $R_1$ 's departure, before it finishes playing out its *pre-fetched* content.  $R_0$  would be forced to start downloading from the server in such a scenario. Thus, under the *Pre-fetch-and-relay* scheme, the probability that  $R_0$  will download from the server due to the premature departure of its current source, represented by  $P_F$ , is

$$P_F = P_d \times P\{\text{Event B}\} + P_d \times (1 - P\{\text{Event B}\}) \times p$$

where the first term represents the scenario where there are no new sources in the peer-to-peer network for  $R_0$  to download from after the departure of its current source, and the second term represents the situation where  $R_0$  finishes playing out its *pre-fetched* content before it is able to locate a new source.

The probability  $p$  can be thought of as the fraction of time when a client does not have enough *pre-fetched* content to be able to locate a new source after the departure of its current source. Fig. 7 compares the performance of *Cache-and-relay* and *Pre-fetch-and-relay* for different values of  $p$ . Note that  $P_F$  degenerates to  $P_d$  for  $p = 1$ . The ratio of the server bandwidth requirement for *Pre-fetch-and-relay* to the server bandwidth requirement for *Cache-and-relay* is plotted along the y-axis. The value of  $\alpha$  is 2 and  $\beta$  is 100,000. The buffer size is 10 time units. It is evident from the figure that in the presence of client departures, *pre-fetching* “sufficient” content by clients can help reduce the server bandwidth requirement significantly.

## B. Simulation Model

The following table presents the settings of the various parameters used to obtain the results presented

in this section.

Figure	Buffer size	$\beta$	$(1/\mu)$	Delay
8	5	100,000	1000	No
9	10	100,000	1000	No
10	20	100,000	1000	No
11	10	100,000	100	No
12	10	1	1000	Yes
13	10	100,000	1000	Yes

Figures 8(left), 9(left) and 10(left) represent the server bandwidth requirement obtained from our analytical model for the simplified cases when  $N = 1$  in Section III-A and  $N = 2$  in Section III-B. The quantity  $(1/\mu)$  represents the average time spent by a client downloading the stream. A “No” in the column labeled “Delay” indicates that the various delays discussed in Section III-D were not considered in calculating the server bandwidth requirement; a “Yes” indicates that the delays were considered. We assume that the total delay involved in switching the streaming session to another client after a departure is uniformly distributed over the interval  $[0,9]$  to generate the results of Figures 12 and 13. With a buffer size of 10 time units and  $\beta = 100,000$ , a client will have 9.999 time units of “future” content in its buffer after it achieves the ratio  $\beta$  whereas with  $\beta = 1$  it will have only 5 time units of “future” content. The total number of client arrivals during each simulation run was 3000. Each point on a graph represents an average over 10 independent runs.

### C. Summary of Observations

If the download rate is sufficiently high,  $\alpha \geq 2$ , dPAM has an advantage over *Cache-and-relay* in reducing the server bandwidth when the resources available for overlay stream multicast are constrained, for example when the buffer size is small or when the request arrival rate is low. The advantage stems from the fact that a higher download rate enables a client to open two simultaneous connections for a short duration to “catch-up” with the buffer of another client using the technique of stream patching. This advantage is more pronounced for higher client departure rate. If clients depart frequently from the peer-to-peer network, it reduces the caching capacity of the peer-to-peer network, thus patching content from the server becomes more beneficial. As the buffer size and the request arrival rate increase, the advantage of our dPAM protocol over *Cache-and-relay* is mitigated and for a given buffer size, at a sufficiently high request arrival rate, *Cache-and-relay* matches the performance of dPAM in terms of server bandwidth even when the download rate is very high.

Figures 8(right), 9(right) and 10(right) present the server bandwidth requirements obtained through simulations for different buffer sizes without taking into consideration the various delays discussed in III-D. For a fixed buffer size, the *Cache-and-relay* scheme is able

to match the performance of the *Pre-fetch-and-relay* scheme, for all values of  $\alpha$  considered in the simulation, once the client arrival rate increases beyond a certain threshold. This threshold depends on the size of the buffer—80/(60 time units) in Fig. 9(right) and 50/(60 time units) in Fig. 10(right). The reason for this is two-fold. In these simulations, on an average, a client downloads the stream for a duration (1000 time units) which is much longer than the time it needs to achieve the desired ratio  $\beta$  between its “future” and “past” content (approximately 5 time units for  $\alpha = 5$  and buffer size of 20 bytes when  $\beta = 100,000$ ). Thus, most of the clients have achieved the ratio  $\beta$  by the time the client they were downloading from departs from the peer-to-peer network. Since the value of  $\beta$  for these simulations is set very high ( $\beta = 100,000$ ), almost the entire buffer of a client is full of “future” content when the client it was downloading from leaves the overlay network. Hence, in the event of a departure, clients are not able to take advantage of the higher download rate, by opening two simultaneous streams and using stream patching, since condition (2), Section II-C is violated. Thus, on a departure, a client that needs a new source for download, can start downloading from another client only if their buffers overlap (partially). This situation is similar to the case for the *Cache-and-relay* scheme ( $\alpha = 1$ ) (refer to the discussion in Section II) and hence, the stream patching technique employed in the *Pre-fetch-and-relay* scheme does not provide any advantage over the *Cache-and-relay* scheme. If we set  $\beta$  to be small, say 2 or 3, then a client would be able to take advantage of the stream patching technique in the event of a departure.

The second reason for the *Cache-and-relay* scheme being able to achieve the same server load as *Pre-fetch-and-relay* beyond a certain request arrival rate threshold is as follows. A new arrival can take advantage of higher download rate and stream patching under the *Pre-fetch-and-relay* scheme but as client arrival rate becomes higher, a client rarely needs to resort to it. Hence, for a sufficiently high client arrival rate, the advantage that the *Pre-fetch-and-relay* scheme has over the *Cache-and-relay* scheme, due to clients being able to patch streams from the server (using a higher total download rate), is mitigated by the presence of a large number of clients in the peer-to-peer network enabling the *Cache-and-relay* scheme to match the performance of the *Pre-fetch-and-relay* scheme.

When  $1 < \alpha < 2$ , *Pre-fetch-and-relay* leads to a greater server load than *Cache-and-relay* for small arrival rates. As we increase  $\alpha$ , the time taken to fill the buffer at download speed,  $T_b$  decreases. For example, for a buffer size of 5 time units, for *Cache-and-relay* ( $\alpha = 1$ ),  $T_b = 5$ ; whereas when  $\alpha=1.8$ ,  $T_b=2.78$ . Thus, in the former case, a new arrival can reuse the stream from someone who arrived at most 5 time units earlier whereas in the latter case a new arrival can download from someone who arrived at most 2.78 time units earlier. Hence, in the latter case, more new arrivals have to download from the server. This effect can be mitigated by increasing the

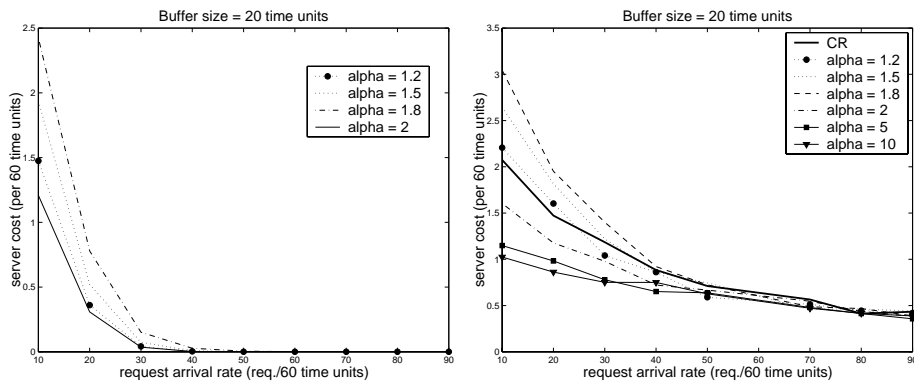


Fig. 10. (left) Analysis; (right) Simulation. Mean download time = 1000 time units, buffer size = 20 bytes

buffer size and also for increasing client arrival rate.

With the download rate at least twice as fast as the playback rate ( $\alpha \geq 2$ ) the *Pre-fetch-and-relay* scheme achieves a much lower server load than the *Cache-and-relay* scheme even for small buffer sizes and low request arrival rate (cf. Figures 8(right), 9(right) and 10(right)). When  $1 < \alpha < 2$ , the *Pre-fetch-and-relay* scheme does not have any advantage over the *Cache-and-relay* scheme. Since we ignore here the delay involved in searching for a suitable client to download from, the only advantage that the *Pre-fetch-and-relay* scheme has over the *Cache-and-relay* scheme, in these simulations, comes from the fact that it enables a client to “catch-up” with another client by downloading the “missing” data from the server. But for small  $\alpha$  ( $1 < \alpha < 2$ ), condition (1) discussed in Section II-C is often violated. For example, with  $\alpha = 1.2$  bytes/second, it will take 5 seconds to acquire a “missing” data of size 1 byte from the server. Hence, when the available buffer size is 5 time units, a client never opens two simultaneous streams because if it contains the required 5 bytes of “future” content to satisfy condition (1) of Section II-C, then its buffer is already full of “future” content and it has no available buffer space to download the “missing” content from the server. If it has less than 5 bytes of “future” content then condition (1) of Section II-C is violated. Combined with our observations in the preceding paragraph, it becomes clear why the *Pre-fetch-and-relay* scheme has a much higher server load compared to the *Cache-and-relay* scheme for  $1 < \alpha < 2$  when the client arrival rate is low.

The results presented in this section also show that as the available buffer at the client increases, the required server bandwidth to support a particular request arrival rate decreases, both in *Cache-and-relay* as well as *Pre-fetch-and-relay* (for all values of  $\alpha$ ). This observation is in agreement with the results obtained in [6].

The amount of time that clients spend downloading a stream is an important factor in determining server bandwidth requirements. Peer-to-peer network based asynchronous media content distribution is suited for situations in which the content being distributed is

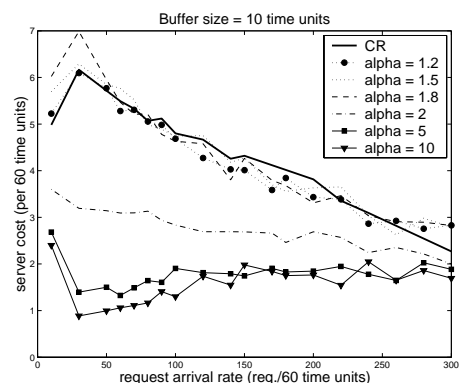


Fig. 11. Mean download time = 100 time units, buffer size = 10 bytes

large; so that the end-hosts participating in the peer-to-peer network are available for a long time. In a scenario where end-hosts keep departing after a short interval, the server load can be considerably high due to the fact that a lot of requests may have to start downloading from the server due to the departure of clients they were downloading from. Fig. 11 presents the simulation results when the mean time spent by a client downloading the stream ( $1/\mu$ ) is 100 time units. Compared to the case when  $1/\mu = 1000$ , the server bandwidth requirement is considerably higher even for very high client arrival rates. Fig. 11 illustrates the fact that the *Pre-fetch-and-relay* scheme with  $\alpha \geq 2$  performs better than the *Cache-and-relay* scheme, in terms of the server bandwidth requirement, even for very high client arrival rates (compare with Figures 8(right), 9(right) and 10(right)). Due to the shorter content download time of the clients, in a significant number of situations, clients are able to take advantage of the higher download rate through stream patching [13] in the event of a departure and hence, the server load is less in the case of *Pre-fetch-and-relay* scheme as compared to the *Cache-and-relay* scheme. Fig. 11 also shows that if the client request arrival rate keeps on increasing, eventually the *Cache-*

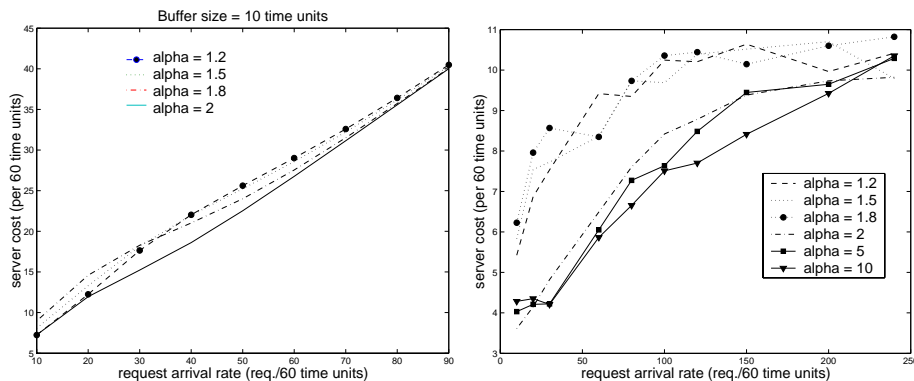


Fig. 12. (left) Analysis; (right) Simulation. With DELAY, Mean download time = 1000 time units, buffer size = 10 bytes,  $\beta = 1$

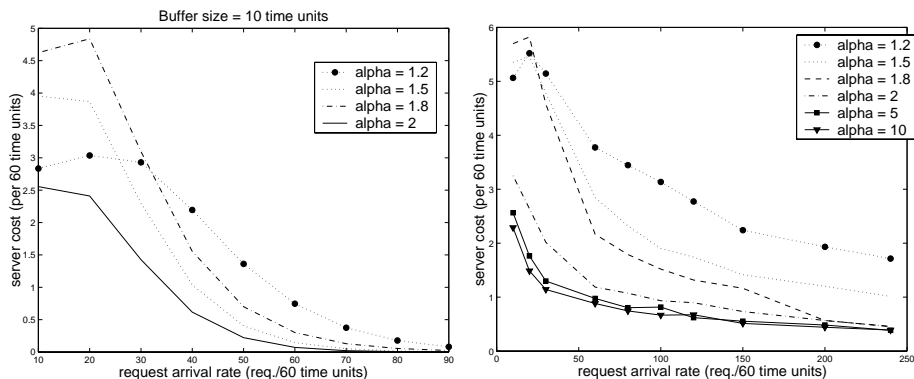


Fig. 13. (left) Analysis; (right) Simulation. With DELAY, Mean download time = 1000 time units, buffer size = 10 bytes,  $\beta = 100,000$

*and-relay* scheme will be able to match the performance of the *Pre-fetch-and-relay* scheme.

The server bandwidth requirements obtained from our analytical model (Figures 8(left), 9(left), 10(left)) display the same trends as observed through simulations. The analytical results are more optimistic than the results obtained through simulations because of the assumption that clients are able to achieve the ratio  $\beta$  in the buffer before they are forced to switch their streaming session because of client departures. If the ratio between the “future” and the “past” content in a client’s buffer is less than  $\beta$ , then not only does it have less time to discover another client to download from but the number of suitable clients available in the peer-to-peer network is also reduced because of a smaller  $T_f$ .<sup>5</sup>

Figures 12 and 13 present the results from analysis and simulations after incorporating the various delays into our model. In Figure 12, with  $\beta = 1$ , clients have 5 time units of “future” content whereas the delays involved are uniformly distributed over the interval  $[0,9]$ . Hence, in a significant number of cases, a client would be forced to download the stream from the server after a departure because it would be unable to find another client to

download from before it finishes playing out the “future” content. As a result, the server bandwidth requirement keeps on increasing even for high client arrival rate. In Figure 13, with  $\beta = 100,000$ , clients have 9.99 time units of “future” content. Hence, clients are able to absorb the delays without any disruption to their playout and are not forced to download from the server. As a result, the server bandwidth requirement displays the same trend as observed when delays were not considered. These results aim to underscore the importance of taking advantage of a higher download rate to *pre-fetch* content in achieving a better playout performance and lower server bandwidth requirement.

## V. IMPLEMENTATION ISSUES

There are two main components to dPAM: (1) Buffer Management: How should a node manage its buffer? and (2) Content Location: How does a node locate a set of potential sources from which to prefetch its content upon arrival, or upon being disconnected due to the premature departure of its current source? In this paper, we have focused on the buffer management component of dPAM. In this section, we briefly outline how the content location component of dPAM could be readily implemented.

<sup>5</sup>As discussed in Section III-B, the size of  $T_f$  is related to the “future” and the “past” content in the buffers of the clients.

First, we note that the content cached in any P2P node is uniquely identifiable by the absolute time of the first byte of the feed currently being played out at that node. For example, if at time  $t$  a node  $i$  is playing out the  $d^{\text{th}}$  second of a feed  $f$ , then the content of node  $i$  storage at any point in time is uniquely identified by the string  $f@t_0$  where  $t_0 = t - d$ . Thus one can treat  $f@t_0$  as the unique reference string for the content available at node  $i$ . Similarly, it also follows that a node  $j$  searching for content could uniquely generate the reference string for the cached content it is seeking by specifying the name of the feed and its own starting time for that feed.

Our ability to uniquely name the cache content at any node in the system gives us a simple mechanism for implementing dPAM's content location component. Specifically, this could be done using any number of existing P2P content location protocols, ranging from widely deployed controlled flooding protocols (e.g., gnutella [1]) to more scalable DHT-based protocols (e.g., CAN [21] and CHORD [23]), or recently proposed hybrids thereof [10]. Upon joining (leaving) the asynchronous multicast of a feed  $f$ , a node would advertise the availability of its cached portion of  $f$  by adding (removing) the appropriate reference string into (from) the pool of available content (this is akin to a node adding/removing a file to/from the set of files it is contributing to a P2P file sharing application). When a node needs to locate a new source of content, it would query the system using an appropriately formed reference string (possibly including wild-cards to allow for ranges, etc.)<sup>6</sup> If matches exist (or are locatable using the content location protocol), that node would receive a list of all candidate sources, possibly with additional meta information coded into the matching reference strings (e.g., load information, network distance, etc.) From such a candidate list, the node would select the most appropriate source. If matches do not exist, or if the candidate list is not responsive (perhaps due to delays in propagating departures through the P2P name space), the node will resort back to the server.

## VI. RELATED WORK

Delivery of streams to asynchronous clients has been the focus of many studies, including periodic broadcasting [24], [14], [12], [18] and stream patching/merging techniques [5], [11], [7], [8]. In periodic broadcasting, segments of the media object (with increasing sizes) are periodically broadcasted on dedicated channels, and asynchronous clients join one or more broadcasting channels to download the content. The approach of patching [13] allows asynchronous client requests to "catch up" with an ongoing multicast session by downloading the missing portion through server unicast. In merging techniques [9], clients merge into larger and larger multicast session repeatedly, thus reducing both the server bandwidth and the network link cost. These

techniques rely on the availability of a multicast delivery infrastructure at the lower level.

The idea of utilizing client-side caches has been proposed in several previous work [22], [20], [15]. The authors of [6], propose an overlay, multicast strategy, *oStream*, that leverages client-side caching to reduce the server bandwidth as well the network link cost. Assuming the client arrivals to be Poisson distributed, they also derive analytical bounds on the server bandwidth and network link cost. However, this work does not consider the effect of the departure of the end-systems from the overlay network on the efficiency of overlay multicast. As mention earlier, *oStream*, does not consider the effect of streaming rate, it is a *Cache-and-relay* strategy, and hence, does not incorporate patching techniques to reduce server bandwidth when the download rate is high. The main objective of the protocol, *OSMOSIS*, proposed in [17] is to reduce the network link cost. The effect of patching on server load has not been studied.

A different approach to content delivery is the use of periodic broadcasting of encoded content as was done over broadcast disks [2] using IDA [19], and more recently using the Digital Fountain approach which relies on Tornado encoding [4], [3]. These techniques enable end-hosts to reconstruct the original content of size  $n$  using a subset of any  $n$  symbols from a large set of encoded symbols. Reliability and a substantial degree of application layer flexibility can be achieved using such techniques. But these techniques are not able to efficiently deal with real-time (live or near-live) streaming media content due to the necessity of encoding/decoding rather large stored data segments.

## VII. CONCLUSION

We proposed a *pre-fetch-and-relay* protocol that allows a peer to serve as a source for other peers, while prefetching a portion of the stream ahead of its playout time. In contrast to existing cache-and-relay schemes, our scheme is more scalable in highly dynamic P2P systems. This is because a departure of a peer does not necessarily force its children peers (for whom it is serving as source) to go to the original server. Rather a child peer can continue its playout uninterrupted from its prefetched data until it discovers a new source peer. Through mathematical analysis and extensive simulations, we show that, if the download rate is sufficiently greater than the playout rate ( $\alpha \geq 2$ ), our distributed prefetching scheme significantly reduces the load on the server as it effectively increases the capacity of the P2P system. At the same time, clients can achieve a better playout performance.

<sup>6</sup>In our dPAM scheme, a node arriving at  $t_0$  needs to query a string range that spans the arrival times  $[t_0 - W, t_0)$ .

## APPENDIX

As discussed in Section II-C the download rate  $\alpha$  and the buffer size impose certain constraints on the size of the “missing” content,  $T_H$ , for  $R_2$  to be suitable for  $R_0$  to download from (see Fig. 4). If  $\alpha \geq 2$ , condition (1), Section II-C is always satisfied and hence, the maximum size of the “missing” content is given by condition (2), Section II-C. Hence, when  $\alpha \geq 2$ , the maximum value of  $T_f$ , assuming that both  $R_2$  and  $R_0$  maintain the ratio  $\beta$  in their buffers, is

$$\begin{aligned} T_f &= \left( \frac{\beta\alpha T_b}{1+\beta} \right) + \left( \frac{\alpha T_b}{1+\beta} \right) + \left( \frac{\alpha T_b}{1+\beta} \right) \\ \Rightarrow T_f &= (\alpha \times T_b) \left( \frac{2+\beta}{1+\beta} \right) \end{aligned}$$

Now consider the case of  $1 < \alpha < 2$ . Condition (1), Section II-C can be restated as

$$T_H \leq \left( \frac{\alpha\beta T_b}{1+\beta} \right) \left( \frac{\alpha-1}{2-\alpha} \right)$$

We can derive the condition on  $\alpha$  that determines whether condition (1), Section II-C or condition (2), Section II-C restricts the maximum size of the “missing” content; condition (1), Section II-C determines the maximum size of the missing content *iff*

$$\begin{aligned} \left( \frac{\alpha\beta T_b}{1+\beta} \right) \left( \frac{\alpha-1}{2-\alpha} \right) &\leq \left( \frac{\alpha T_b}{1+\beta} \right) \\ \Rightarrow \alpha &\leq \left( \frac{2+\beta}{1+\beta} \right) \end{aligned} \quad (7)$$

If (7) is satisfied then the maximum value of  $T_f$ , assuming that both  $R_2$  and  $R_0$  maintain the ratio  $\beta$  in their buffers, is

$$\begin{aligned} T_f &= \left( \frac{\beta\alpha T_b}{1+\beta} \right) + \left( \frac{\alpha\beta T_b}{1+\beta} \right) \left( \frac{\alpha-1}{2-\alpha} \right) + \left( \frac{\alpha T_b}{1+\beta} \right) \\ &= (\alpha \times T_b) \left( \frac{\beta - \alpha + 2}{(1+\beta)(2-\alpha)} \right) \end{aligned}$$

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