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Citation:

Mostafa M. Fouda, Zubair Md. Fadlullah, Mohsen Guizani, and Nei Kato, "A Novel P2P VoD Streaming Technique Integrating Localization and Congestion Awareness Strategies," Springer Mobile Networks and Applications (MONET), vol. 17, no. 5, pp. 594-603, Oct. 2012.

URL:

http://www.springerlink.com/content/463u11672g3v0471/

A Novel P2P VoD Streaming Technique Integrating Localization and Congestion Awareness Strategies

Mostafa M. Fouda · Zubair Md. Fadlullah · Mohsen Guizani · Nei Kato

Abstract The concept of the "future Internet" has evolved amongst researchers recently to relieve the tremendous pressure on the current Internet infrastructure to support the heterogeneous networking technologies, mobile devices, increased population of users, and also the high user requirements for real-time services and applications. Peer-to-Peer (P2P) Video on Demand (VoD) streaming technologies are expected to be a key technology in the future Internet. Because the existing P2P streaming techniques are attributed to a number of shortcomings, P2P VoD schemes need to be adequately redesigned for the future Internet. In this paper, we propose a scheme to effectively provide VoD by using P2P-based mesh overlay networks that may be suitable for the future Internet. Our scheme selects the most appropriate peers by exploiting domain-based localization and congestion awareness strategies. Through simulations, our proposed scheme is demonstrated to have scalability and capability of reducing the startup delay and total link cost, while maintaining high playback rate. The results are encouraging and show the importance of redesigning P2P VoD services in future Internet.

Keywords Peer-to-Peer · Video-on-Demand streaming · Future Internet

Part of the work has been presented in Chinacom'10 [1].

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1 Introduction

Over the last decade, the capacities of the Internet in both access and core networks have vastly improved. According to the estimation of Gartner, the worldwide broadband installations is expected to reach an overwhelming figure of 580 million households in 2013, i.e., the popularity rate of broadband subscribers will be around 25% [2]. As per a report from Cisco, the growth of the Internet in 2015 will be four times larger than in 2010 [3]. The fast deployment of broadband access networks enables massive data delivery at a fast pace. Instead of being restricted with delivery of only text and images, broadband Internet brings diverse audio and video streaming applications.

Also, recently, a huge variety of services have evolved in order to satisfy the ever-increasing demands for sharing information. However, because the existing Internet architecture was designed four decades ago, in the near future it may falter under the tremendous pressure created by the increasing users' requirements in terms of Quality of Service (QoS). In addition, the current Internet architecture also lacks a number of features like reliability, resiliency, security, mobility, context-awareness, and so forth. To address this issue, two projects were initiated by the National Science Foundation (NSF) in the United States called Global Environment for Network Innovations (GENI) [4] and Future InterNet Design (FIND) [5]. Both the GENI and FIND projects focused on addressing the need for cross-layer Internet design, network virtualization, dynamic switching of optical circuits, service discovery and composition, service management, traffic and routing engineering, along with many other directions. With the similar objectives in mind, the European Commission also initiated its own project termed as the Future Internet Research and Experimentation (FIRE) [6] and that is within the 7^{th} Framework Program (FP7). Indeed, the proposals in FIRE indicated the clear need to carry out early experimentation and testing over large-scale environments rather than solely relying on theoretical definitions of next generation Internet architectures, protocols, and services. The AKARI project in Japan was initiated by the National Institute of Information and Communications Technology (NICT) [7] and it stressed upon generating ideas, technologies, and a NeW Generation Network (NWGN) architecture by the year 2015. The above-mentioned projects and research endeavors have identified one of the serious shortcomings of the current Internet that consists in dealing with traffic volume. The network operators are required to offload traffic at optimal points of the current Internet framework.

Recent years have seen a rapid growth of video traffic over the Internet. According to [3], Internet video is now 40% of all consumer Internet traffic, and will account for 62% of all consumer Internet traffic in 2015. To this end, a major goal of the future Internet projects is to envision improved means of handling data traffic, substantial portion of which is currently generated by video streaming. Video streaming can be categorized into two streaming techniques, namely live video streaming and Video on-Demand (VoD) streaming.

Live video streaming is a broadcast of a live television channel or any other video content to multiple users at the same time. The traditional way to do this is to send the same video chunks to each client separately which requires huge bandwidth. In addition, this technique is not scalable. Another solution to do this is by using Internet Protocol (IP) multicast technology. In IP multicast, the server sends the video chunks to multiple assigned clients with only one traffic flow. Using this technology should avoid having a bottle-neck near the source since only one traffic is being transferred. The main problem with IP multicast is that not all the routers in the current Internet infrastructure support such kind of technology which makes it a rather restricted technology. To this end, a cheaper and reliable technology is needed to enable live video streaming over the Internet in an efficient manner.

On the other hand, Video on-Demand (VoD) streaming is more popular for Internet users since it provides user-interactive facilities (e.g., pause, random jumps, rewind, forward, slow motion, and so forth). One of the popular examples of VoD streaming is "YouTube" where Internet users can watch any part of a video anytime. With these more advanced features of VoD over live video streaming, multicasting cannot present itself as an adequate solution. As we mentioned earlier, sending separate chunks to individual costumers is also expensive. So, it is recommended to design a cheaper and reliable technology, which needs to be carefully designed to achieve low cost, decent Quality of Experience (QoE), and high scalability.

Three different ways do exist for sending such kind of traffic. The first technique is the traditional client-server paradigm whereby a single server sends the data to consumers. The second way consists in the Content Distribution Network (CDN), which is designed to avoid bottleneck near that server by having duplicated servers in different regions. The third way is the Peer-to-Peer (P2P) overlay networks where the resources of the clients are utilized for uploading the content without the need for a server. It should be noted that hybrid technologies also exist. However, the cheapest technology is based on Peer-to-Peer (P2P), since it is completely based on clients' resources.

P2P applications are employed for sharing files and streaming live/on-demand multimedia contents. These P2P applications involve a huge volume of information exchanged along both uplink and downlink directions amongst users/peers. Compared to the more commonly used client/server-based applications evolved in the traditional Internet, P2P programs often select one or more suitable candidates from a pool of peers, which offer the same services or resources. This unique feature of P2P applications leads to increased reliability and resiliency to a single point of failure. As a consequence, P2P applications are expected to be a vital part of future Internet.

The aforementioned points make the design and deployment of a P2P-VoD system in the future Internet rather difficult due to complex features of VoD services. Therefore, the research work on developing P2P-VoD architectures to fit the future Internet framework is a timely and important one. In order to effectively overcome this problem, we envision a novel P2P VoD scheme in this paper.

The remainder of this paper is structured as follows. Section II surveys several future Internet initiatives and also presents related research work on developing P2P VoD schemes. An overview of real time VoD P2P streaming systems is provided in Section III. Next, in Section IV, our envisioned P2P-based VoD system is presented. The simulation results are provided in Section V to demonstrate the effectiveness of the proposed scheme. Concluding remarks are presented in in Section VI.

2 Related Work

There is a number of recent initiatives in order to find the specific requirements of the future Internet. This section first describes several future Internet initiatives or projects carried out in the recent past. Following this, a number of P2P VoD schemes introduced by recent researchers that may be incorporated with the upcoming future Internet are surveyed.

2.1 Recently Conducted Future Internet Projects

The Internet has come a long way since its conception in the 1970s. The very concept that led to the fascinating breakthrough in Internet communications has become rather obsolete as it was not originally designed to address the current user-requirements for various services over heterogeneous networks. In fact, it is surprising to many that the Internet has survived so many decades of pressure from a wide variety and ever-increasing number of wired and wireless mobile users and applications, state-of-the art business models, latest networking devices and equipment (e.g., multiple wireless interfaces that enable users to connect to different networks at the same time [8]), and so on.

Therefore, these limitations of the Internet are taken into serious consideration by leading researchers which have prompted them to resort to a concept of providing a futuristic Internet framework.

Currently, a number of research projects are being carried out to develop the future Internet. One of these notable projects for dealing with the challenges of today's Internet and envisioning future Internet networks is referred to as FIND [5]. The FIND project was carried out by NSF. FIND stresses on the developing network architecture, security, advanced wireless and optical properties, economical principles, and in general, the means to effectively construct a global network in fifteen years from now. Another project called GENI [4] program was initiated to address the absence of security, reliability, evolvability, and manageability of the current Internet. In Europe, there were also similar projects. For example, the FIREworks (Future Internet Research and Experimentation - Strategy Works) project [9] and the EIFFEL (Evolved Internet Future for European Leadership [10]) Support Action (SA) for the 7$^{\text{th}}$ Framework Program (FP7) are prominent future Internet initiatives taken by European countries. Although these different projects have the same end goal (i.e., facilitating future Internet) in their focus, their means are not necessarily similar. In fact, the contrast in design goals is also present in the various Internet Engineering Task Force (IETF) work groups. For example, one IETF group envisioned Low Extra Delay BAckground Transport (LEDBAT) protocol [11] for minimizing the additional delay caused by sophisticated networking applications while another one introduced the Diffserv (Differentiated services) Code Point for bulk traffic to identify, mark, and manage congestion events arising from P2P traffic under Diffserv framework. Furthermore, to manage the P2P traffic volume within an acceptable level at the network operators' expensive links, two IETF groups were formulated, namely DECoupled Application Data Enroute (DECADE) [12] and Application Layer Traffic Optimization (ALTO) [13] working groups. DECADE identifies the problem with current Internet caches (e.g., P2P and web caches), which attempt to provide adequate storage capabilities within the network for reliable access of resources. However, DECADE is not able to explicitly support individual P2P application protocols or user access to the content providers' caches. The latter attempts at providing a simple means for conveying network information to the concerned P2P applications. This assists these P2P applications to mitigate overhead of measuring topology information. Furthermore, the work in [14] suggested, instead of random initial peer selection, adoption of traffic localization for P2P applications by introducing ALTO service.

2.2 Related work on P2P VoD schemes

To meet the requirements of future Internet, a number of research works have emerged recently in literature that have attempted to combine P2P strategies with the server-client streaming framework. A note-worthy example is (BASS) [15], which comprises an external media server and a modified BitTorrent [16] protocol. The clients in BASS are not able to download contents prior to the current playback time. Instead, these clients receive chunks from the media server sequentially. Also, they do not need to obtain the already/being downloaded chunks by the BitTorrent. However, the BASS framework lacks scalability when the system serves a large number of peers.

To deal with the above problem, no external media server was taken into account in the BitTorrent Streaming (BiToS) framework [17]. Rather than adopting the "rarest first" strategy, BiToS uses a selection mechanism, which ignores missing chunks (i.e., the chunks unable to meet their playback time). Due to these chunks being missed during the download process, the video playback is disrupted.

The work in [18] focuses on providing small startup delays in P2P-VoD systems by integrating network coding with segment scheduling. Thus, it achieves a high utilization of the available resources. However, this approach does not consider practically deployable P2P VoD environments. In order to overcome this problem, it is important to design a scheduling mechanism capable of instructing the involved peers to assist one another in streaming the VoD contents in a real-time manner. To this end, the authors proposed a VoD scheme in an earlier work [19] over P2P mesh-based overlay networks with user-scalability features.

3 Overview of conventional P2P VoD streaming

Along with advances in broadband technologies, multimedia communications such as interactive on-demand video streaming services have become popular Internet applications in recent time (e.g., YouTube). In the traditional client/server setting, videos are delivered to the users through a centralized entity/server. Two of the main drawbacks of such a VoD scheme are the lack of scalability (i.e., support of a limited number of subscribers) and the limited streaming rate at the end-users which is due to the limited bandwidth of the server. P2P-based streaming has recently emerged as an alternative to traditional server-based streaming (e.g., Youtube) for VoD provisioning. It already emerged as a promising technology for the future Internet targeting global users connected to heterogeneous networks. From the perspectives of a broadcaster/content provider, the P2P approach provides an added incentive as it allows the servicing of a large audience (i.e., many peers) without the need for investment in additional resources. On the other hand, a user also experiences improved delivery rate of the multimedia content while he/she can also upload his/her own content to other peers. The implication of use of P2P VoD in future Internet is immense, as it tolerates only minimal change to the existing infrastructure. In addition, P2P VoD approaches may also overcome bandwidth/processing load bottlenecks, reduce startup and end-to-end delays, and also improve the playback rate of the video. The existing P2P VoD systems can be broadly classified as either tree-based or mesh-based ones. In the tree-based systems, every node receives data from a source/parent node according to a tree-like structure. This intrinsic design of tree-based P2P VoD systems leads to a significant problem in future Internet dynamics whereby the nodes are expected to frequently change resulting in frequent re-construction of the trees. This will render the children nodes to fail to obtain the streaming feed until the tree is re-constructed. It is worth noting that Application Layer Multicast (ALM), which aims at replacing IP multicasting for content delivery, also suffers from this shortcoming [20,21]. ALM nodes, depending on the streaming application, construct a multicast tree, through which the stream is delivered. However, if a node leaves the tree, it is unable to further deliver the stream to its descendant nodes till the tree is recovered.

In case of the mesh-based overlay network, a peer, which has just joined the overlay, contacts the tracker (e.g., a directory server) to receive a list of a number of active peers. After receiving the same, this new peer immediately starts to request the video chunks from the peers in the neighbor-list. Also, note that a peer is connected with a small subset of active peers at any time, and it is permitted to exchange video chunks and control messages only with them. The overall architecture of the mesh-based P2P VoD is illustrated in Fig. 1. Also, it is worth-noting that the mesh-based P2P VoD networks are more resilient to node failures compared with their tree-based counterpart.

Another weak assumption made by most of the existing P2P-based VoD systems is as follows. A new user keeps watching the video stream until (i) the user leaves the mesh overlay, or (ii) the streaming session fails. This assumption does not take into account the user-interaction features (e.g., pausing, replaying, fast forwarding, and so forth) in order to simplify the system design. In practice, instead of continuously playing the video, the VoD subscribers tend to often jump to a more interesting section of the video either because they do not feel interested to watch that segment or do not have enough time to watch the entire video. The frequent interactions of the users in the P2P VoD applications present a significant challenge on the continuity of the video playback in terms of the "seeking delay", which refers to the time required since the request for a video segment till the segment becomes available. In order to ensure an almost zero seeking delay to watch the video without disruption, each video segment requires to be pre-fetched by a peer prior to the playback of the segment. This enhances the playback continuity. Most conventional P2P VoD systems, however, as mentioned earlier, based on their assumption that the users watch the video sequentially without any interactive "seek", perform the "pre-fetching" of video segments in a sequential manner. An alternative to this approach is to adopt a mechanism that pre-fetches random video segments. In this vein, we propose a pre-fetching mechanism for obtaining appropriate video segments *a priori*

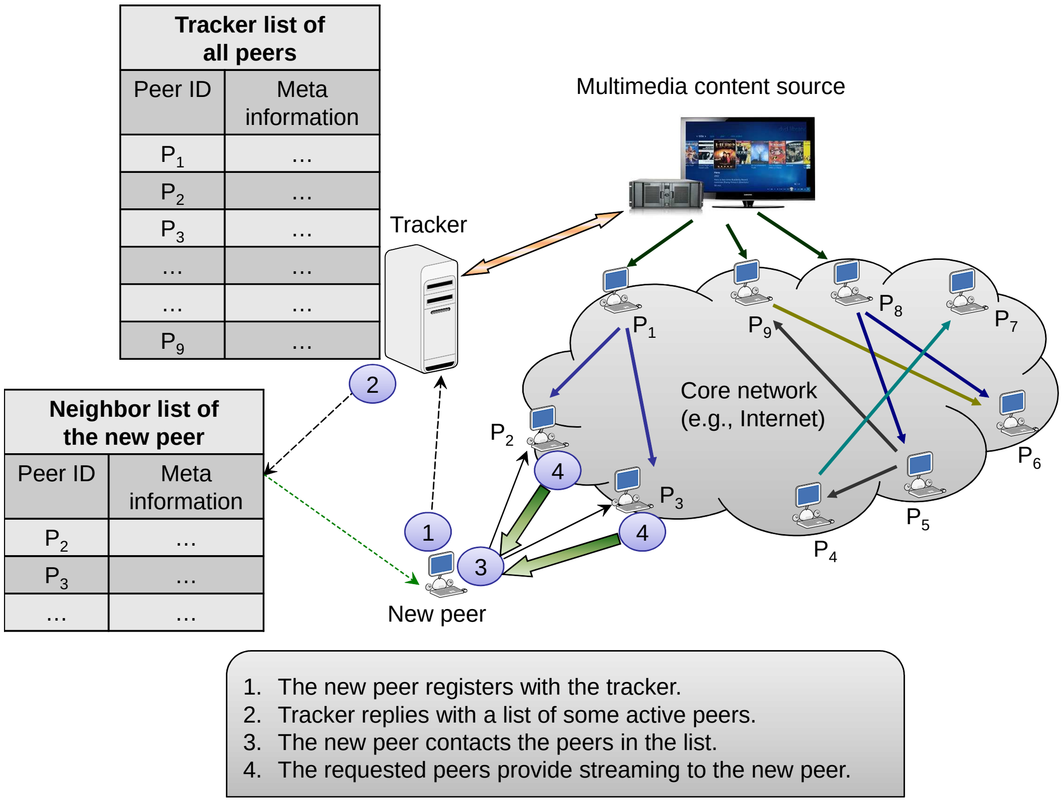

Fig. 1 The mechanism of a new host/peer joining the P2P-VoD overlay.

at the peers in a mesh-P2P VoD environment in the next section.

One of the design goals of a P2P VoD system is to maximize the aggregate throughput of all the peers. While throughput maximization in P2P applications has not been studied well in literature, by attempting to maximize throughput, a peer may obtain duplicate packets (i.e., same video segments or chunks) as it receives streams from multiple sources. As a remedy, the peer may negotiate with the source nodes and request for specific chunks. In addition, some contemporary P2P systems allocate the upload bandwidth to each of its outgoing links either equally or proportionally to the download bandwidth of the receiving peers. These conventional approaches may also suffer from inefficient usage of network resources in large and heterogeneous environments, as expected in the future Internet, and therefore, they may not be able to maximize the aggregate throughput.

4 Envisioned P2P VoD Scheme

In the earlier sections, we presented reasons behind the need to design an effective P2P-VoD system for overcoming the shortcomings of its conventional counterparts, and to fit the scope of the future Internet architecture and meet the high expectations of the subscribers. In this section, we propose a P2P VoD mechanism, which utilizes the overlay network resources by considering the available upload bandwidth of the source node and that of each peer, and designing an efficient pre-fetching scheme coupled with a scheduling mechanism for obtaining *a priori* the appropriate video segments. Note that our envisioned scheme presents a means for modifying the existing P2P VoD designs on the current Internet, and we stress on the fact that such enhancements are crucial for dealing with the new trends and design challenges of P2P VoD in the future Internet.

Similar in spirit to the ALTO service [14], the proposed system is composed of a "tracker", which contains the list of existing peers in the P2P overlay. The tracker contains also the information of the peers, i.e., their Internet Protocol (IP) addresses, port numbers, upload capacities, and chunk bitmap information. Also, the notations in Table 1 are used throughout the remainder of this paper. Let P_{src}, P_{rec}, and P_{srv} indicate the source node (i.e., the content distributor), the receiving peer, and a selected peer to serve the receiving peer, respectively. When a new peer arrives at the considered P2P mesh overlay, it expects to receive video streaming from other nodes. This new node is denoted by P_{rec}. A Neighbor List, NL, is provided to P_{rec} by the

Table 1 Definitions of the notations used in the proposed approach.

Notation	Definition
P_{src}	Source node
P_{rec}	Receiving node/peer
P_{srv}	A selected peer to serve the receiving node
NL	Neighborhood List
TTF	Time-To-Freeze
TTF_{th}	Threshold of Time-To-Freeze
K	Percentage of upload bandwidth assigned to requesting peers with $TTF < TTF_{th}$
D_i	Domain i, i=1,2,3,4
d_{P_i,P_j}	End-to-end propagation delay between two neighboring peers P_i and P_j
u_s	Upload capacity of media source
u_i	Upload capacity of peer i
N	Total number of peers
R_{max}	Maximum streaming rate
T	Time window

tracker. Rather than arbitrarily selecting P_{srv}, the proposed scheme adopts the following steps in the tracker.

1. The peers, which contain the video segments following the last playable chunk of P_{rec}, are chosen as potential P_{srv} candidates.
2. Among the above selected candidates, the tracker chooses the ones with adequate upload capacity and the minimum appearance in the NLs of the currently receiving peers. The reason behind this is to distribute the load over the participating peers as per their available resources.
3. P_{src} is not included in NL in case the number of already chosen P_{srv} is sufficient. Since the source node contains the entire video content, the inclusion of P_{src} is allowed only in critical situations (i.e., when there is no P_{srv} to serve P_{rec}, or all the serving peers, together, do not have the needed video chunks).

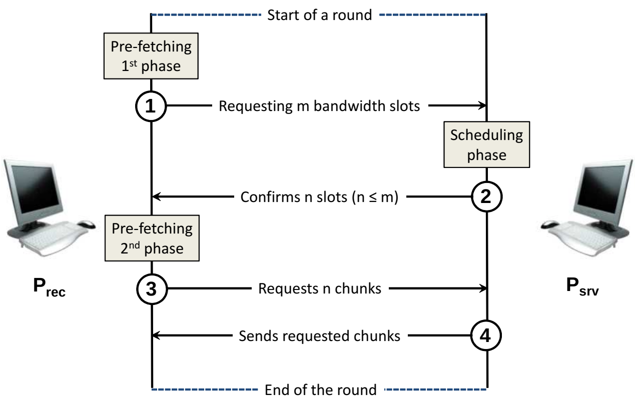

Fig. 2 Envisioned communication algorithm between P_{rec} and P_{srv} comprising pre-fetching and scheduling.

4. In case NL of P_{rec} is incomplete, random peers may be chosen as P_{srv} and they may be added to the NL.

Next, we describe the proposed pre-fetching and scheduling processes, which are implemented in P_{rec} and P_{srv}, respectively. The overall mechanism is demonstrated in Fig. 2. In the following, we present these two algorithms.

4.1 Pre-fetching Process

The pre-fetching process consists of two phases. In the first phase, P_{rec} requests each P_{srv} in its NL for specific bandwidth slots. Note that each bandwidth slot is sufficient for transferring a single video chunk. For enhancing the utilization efficiency of the available bandwidth, the number of bandwidth slots requested by P_{rec} is set directly proportional to the upload capacity of each P_{srv}. This is not allowed to exceed the maximum number of needed chunks available at the P_{srv}. During the same phase, P_{rec} transmits its own Time-To-Freeze (TTF) to P_{srv}. The TTF can be defined as the remaining time for the video playback to enter the freezing state. Fig. 3 shows this through an example whereby P_{srv} exploits the received TTF value for prioritizing requests during the scheduling process. The second phase of pre-fetching is processed after the scheduling process.

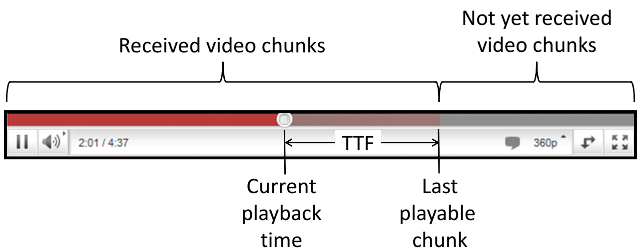

Fig. 3 Depiction of Time-To-Freeze (TTF) parameter in a typical example.

4.2 Scheduling Process

The proposed scheduling scheme implemented at P_{srv} allocates the available upload bandwidth of P_{srv} to the currently requesting P_{rec}. The steps associated with the proposed scheduling phase are as follows.

1. While giving the highest priority to P_{rec} with the minimum TTF, P_{srv} assigns up to K% of its available upload bandwidth to P_{rec}s with a TTF value less than that of TTF_{th} ($TTF < TTF_{th}$), where TTF_{th} is a pre-defined threshold.

2. The remaining bandwidth of P_{srv} is allocated according to the upload capacities of P_{rec}s. The rationale behind this lies on the observation that the requesting P_{rec} with higher upload capacities should be assigned more bandwidth by P_{srv} so that those P_{rec}s may obtain more chunks to utilize their upload capacities efficiently while serving contents to other peers.

It is worth noting that a TTF of a newly arriving peer that has not yet received any chunks is considered to be zero. This means that the highest priority is given to the newly joined peers. As a result of this, the startup latency of the new peers' playback can be lowered down to a minimum level.

Following the scheduling process, P_{srv} transmits a specific number of bandwidth slots to each requesting P_{rec}. This event initiates the second phase of pre-fetching (as described in Section 4.1). In this second phase of pre-fetching, P_{rec} determines the appropriate video chunks, which are to be requested from each P_{srv}.

Because the main objective of the envisioned system is to maintain smooth playback, the priority in requesting video chunks is given towards those particular chunks, which appear just after the playback time with a maximum interval equal to a pre-defined time window, T. After having all chunks within T being requested, P_{rec} assumes that the streaming quality will not degrade during the next T time window. Following this, there is no need to request chunks sequentially. At this point, if P_{rec} can request its P_{srv} for more chunks, it may consider requesting them in a random fashion to increase the chunks diversity in the overlay.

Now, we focus on the ability of the tracker to select the peers. In the envisioned P2P VoD scheme, the tracker is assumed to possess the knowledge of the network topology, i.e., the different domains of the network. We also assume that the tracker may acquire link congestion information over the different network domains (e.g., through deployment of monitoring agents). The domain-based localization refers to the selection of "nearby" peers belonging to the same domain. As selecting a nearby peer is not always an optimal decision (e.g., in case a segment of the link to a nearby peer is congested), link congestion should be also taken into consideration in the selection of peers. The two cases are depicted in a simple scenario in Fig. 4. There are four domains in this scenario, namely D_1, D_2, D_3, and D_4. Let us consider peers P_1, P_2, and P_3 belonging to domain D_1. Let us consider that the link between P_1 and P_2 is more congested than that between P_1 and P_3. While searching for its neighboring peers, P_1 may be assigned to either P_2 or P_3. P_2 and/or P_3 may be selected according to the domain-based localization scheme. However, at the event of congestion in the link between P_1 and P_2, the congestion awareness scheme will select P_3 based on the network domain and the link congestion. In the next section, we demonstrate the performance evaluation of adding such services to the tracker of the proposed scheme which is based on TTF values.

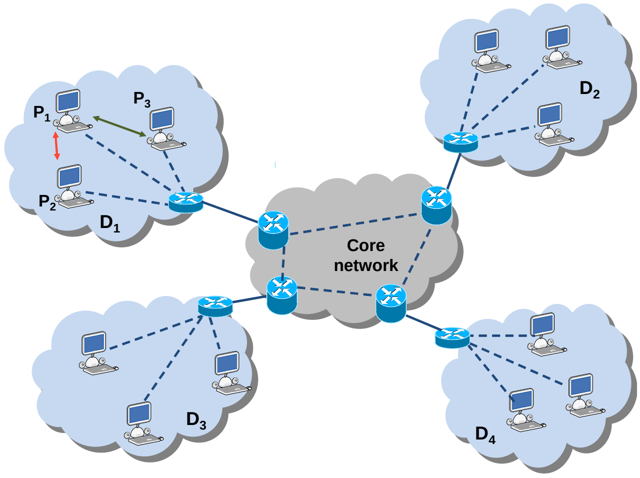

Fig. 4 Domain-based localization vs. congestion awareness strategies for peer selection.

5 Performance Evaluation

To verify the effectiveness of the proposed method and the peer selection strategies (domain-based localization and congestion awareness), we conducted simulations in Matlab [22]. Four different domains are considered for the simulation topology. A single peer is assumed to be the serving node at the beginning. The serviced video file size is set to 16MB, which is usually the average size of the mp4 video files found on sites such as YouTube. The chunk size is set to 8KB. The tracker is updated every 5 seconds. The peer list accommodates up to six peers. A population of 300 peers is used in the simulations.

Two performance metrics are considered for evaluating the proposed approach, namely the maximum supportable playback rate and total link cost. Given the startup delay (or initial buffering), the maximum supportable playback rate is defined as the maximum rate at which the video can be played without being frozen. On the other hand, the total link cost refers to the cost of transferring chunks to all the receiving peers for a particular VoD stream. The total link cost is represented by a weight depending on the congestion of the link, and the distance between the serving and the receiving peers.

Given a content source node and a set of peers with known upload capacities, the maximum/optimal streaming rate, R_{max}, can be formulated from the average upload capacity per peer as follows:

$$R_{max} = \frac{u_s + \sum_{i=1}^{N} u_i}{N} \quad (1)$$

where u_s, u_i, and N denote the upload capacity of the media source node, the upload capacity of peer i, and the number of peers in the system, respectively.

In our simulation, we address the heterogeneity due to the different bandwidth of the access network. In order to do this, the download and upload capacities of the considered peers are set according to the bandwidth distribution used in [23], which is derived from the actual distribution of Gnutella nodes. Specifically, 20% of the peers have upload capacity of 128 kbps, 40% have 384 kbps, 25% use 1 Mbps, and 15% employ 5 Mbps as shown in Table 2. The upload capacity of the source node is set to 5Mbps during the simulation. By using Eq. 1, we found the maximum/optimal streaming rate of the system, R_{max}, to be 1220Kbps for 300 peers.

We compare the performance of our former scheme, TTF-only-based [19] against the proposed enhancements in terms of domain-based peer localization and congestion aware peer selection. Also, another two naive approaches, namely the random and sequential algorithms mentioned in [18] are included in the comparison. We envision a flash crowd scenario, which represents the worst case scenario of a VoD streaming system over P2P overlay. The comparison results are presented in Figs. 5 and 6. In Fig. 5, the maximum supportable playback rate is plotted for the considered schemes over different startup delays. In this scenario, the value of $K\%$ is set to 40%. As evident from this result, both the domain-based peer localization and congestion-aware peer selection schemes outperform the TTF-only-based scheme while the sequential and random schemes give the lowest performance. With the congestion aware peer selection enhancement, the peers can watch the video stream at the maximum supportable playback rate when the startup delay is approximately 8s. The domain-based peer localization enhancement and our previous scheme achieve the maximum playback rate at a later time. This indicates that the user can enjoy smooth playback at an earlier time instance with the proposed enhancements. On the other hand, Fig. 6 depicts the total link cost associated with the proposed schemes. Our previous VoD scheme contributed to the highest total link cost, i.e., 6.2 units. This is because of the fact that the far-away peers may be chosen as neighbors in the previous scheme. The domain-based localization peer selection strategy overcomes this issue as evident by its lower cost (approximately 4.6 units). The congestion-aware mechanism improves the performance further by selecting the less congested peers. The effect of the percentage $K\%$ of the available upload bandwidth assigned to requesting peers with $TTF < TTF_{th}$ is shown in Fig. 7 where $K\%$ is varied between 20% and 100% in the congestion-aware scheme. The results shows that the increase of the percentage $K\%$ improves the performance of the P2P overlay based on the congestion-aware service which reflects the importance of categorizing the peers depending on their TTF value.

Table 2 Peer Classes

Peer's Class	Percent of Peers	Download BW	Upload BW
Class A	15%	10 Mbps	5 Mbps
Class B	25%	3 Mbps	1 Mbps
Class C	40%	1.5 Mbps	384 Kbps
Class D	20%	784 Kbps	128 Kbps

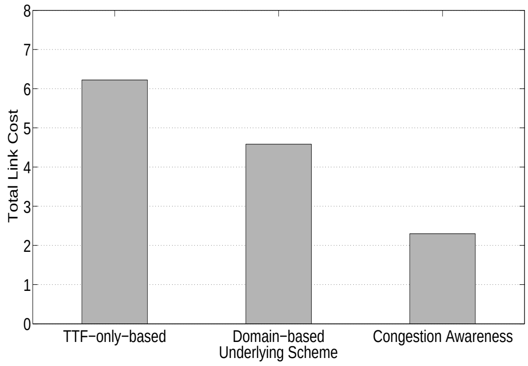

Fig. 6 Total link cost in the considered schemes.

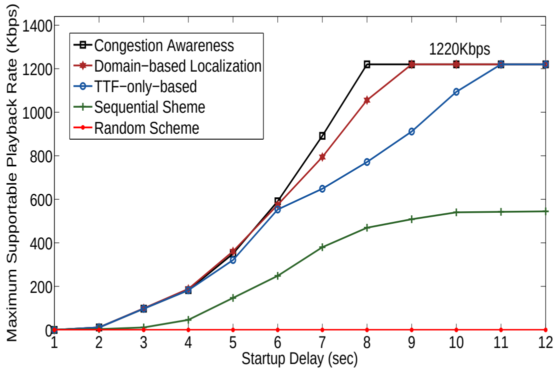

Fig. 5 The maximum supportable playback rates for different startup delays.

6 Conclusion

In summary, we stressed upon the importance of having effective Peer-to-Peer streaming of multimedia contents over the challenging dynamics of the future Internet. For ensuring quality of experience for subscribers

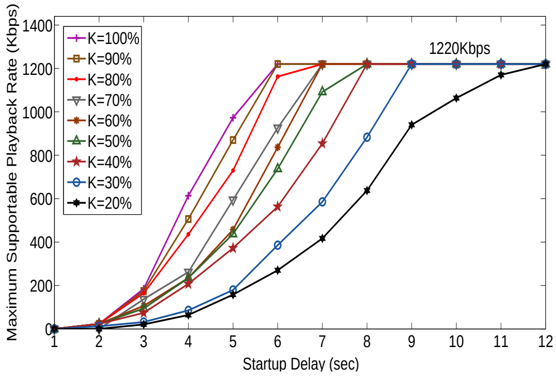

Fig. 7 The effect of the percentage $K\%$ on the performance of the congestion-aware scheme.

of such a P2P VoD service, it is necessary to ascertain short startup delay, smooth video playback, and scalability. Our work has pointed out that the upload bandwidth utilization of the involved peers may significantly affect these. In order to overcome this issue, this paper presents an effective pre-fetching mechanism for obtaining the necessary video segments, an efficient scheduling algorithm for allocating the upload bandwidth, and an appropriate peer selection strategy for selecting the best possible peers. The proposed P2P VoD pre-fetching and scheduling mechanism is integrated with domain-based peer localization and congestion-aware peer selection schemes. We verified the effectiveness of the proposed scheme by using computer simulations. The simulation results demonstrate encouraging performance even when the system is operating under worst case scenarios, i.e., servicing a potential flash crowd of users. Admittedly, further challenges may arise in future Internet settings. For example, a high number of users in future Internet may be expected to be mobile. While it is beyond the scope of our work presented in this paper, it may be interesting to study the effect of mobility of peers on the proposed scheme in future.

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