Adaptive Packet Video Streaming Over P2P Networks Using Active Measurements

Mubashar Mushtaq and Toufik Ahmed
LaBRI – University of Bordeaux 1
351, cours de la Libération
33405 Talence Cedex
FRANCE
{mushtaq, tad}@labri.fr

Abstract

In this paper we consider the problem of real-time streaming of IP packet video over Peer-to-Peer networks (P2P) from multiple senders to a single receiver. P2P networks are characterized by a potentially large and highly dynamic population of hosts that join and leave the network frequently. We present the design and evaluation of a quality adaptation streaming mechanism in a multi-source streaming to a single receiver. Multimedia streaming is a real time application so, the main challenges in the design of this mechanism are (1) selection of senders peers nodes; (2) stream switching among the peers; (3) optimizing video quality by active measurements of links; and (4) enhancing the overall Quality of Service (QoS). Our key technique to provide quality adaptation is based on active measurements of network links and selection of sender peers to enhance the overall throughput. We used video traffic organized as MDC (Multiple Description Coding) layers, which provides high error resilient. Our simulations results using ns2 show that our solution allows to efficiently utilize available network bandwidth of sending peers and allow maximizing streaming qualities at the reception peer.

Keywords: Peer to Peer streaming, Video Streaming, QoS, Quality Adaptation, Active Measurement.

1. Introduction

Content sharing between communities has revolutionized the Internet. During the last few years, we lived a new phenomenon that changed the Internet business model especially for ISP (Internet Service Provider). Peer-to-Peer (P2P) systems have gained tremendous intentions during these years. The Peer-to-Peer (P2P) phenomenon is facilitating information flow from and back to the end users. Unlike traditional distributed systems based on pure client/server model, P2P networks are self organizing networks that aggregate large amount of heterogeneous computers called nodes or peers (nodes and peers are used interchangeably in this paper). In P2P systems, peers can communicate directly with each other for the sharing and exchanging of data, besides this data exchange these peer nodes also share their communication and storage resources. The characteristics of P2P systems make them a better choice for multimedia content sharing/streaming over IP networks. P2P systems are dynamic in nature where nodes can join and leave the network frequently and that might not have a permanent IP addresses and observe dynamic changes over the inter connection links. Virtual networks are built on the top of these networks at the application level in which individual peers communicate with each other and share both communication and storage resources, ideally directly without using a dedicated server.

The main concept of P2P networking is that each peer is a client and a server at the same time. P2P media sharing uses two basic concepts. In the 'open-after-downloading' mode, the media content is played after downloading all the contents of the file from different participants, while the 'play-while-downloading' mode allows playing while downloading the content, which is commonly known as streaming. The 'play-while-downloading' has many advantages over 'open-after-downloading' as it requires lesser memory and client is not expected to wait for longer time to finish download first. In this paper, we consider the problem of Peer-to-Peer streaming defined as a content streaming from multiple senders

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to a single receiver in the P2P network, i.e. a single receiver peer is receiving same content from different peers present in the P2P network. Multiple sender peers are selected on the fact that a single sending peer may not be able or willing to share an outbound bandwidth of actual playback rate. Dynamic behavior of P2P systems is another reason of selecting multiple sender peers for media sharing, as it is possible that any sender peer sharing media can leave/crash without any prior notification.

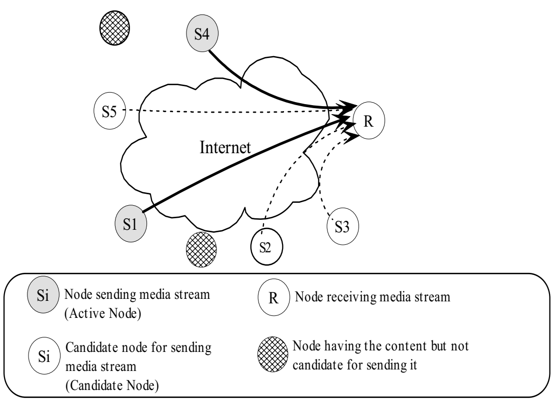

Figure 1: Peer-to-Peer Multimedia Streaming Architecture

Figure 1 illustrates our target architecture, which is composed of many senders and one receiver peer. In this architecture, several peers are connected logically and they form P2P network. In figure 1, we present only those peers which have the requested content. It is possible that there are many other peers present in the network which are not having the requested contents or they don't intended to share their contents. All the peers having the requested contents are named potential peer for sharing of contents. A subset of them is candidate for sending the content during next period of time. Furthermore, a subset of these candidate peers is selected called active peers. The receiver peer orchestrates the overall streaming mechanism by selecting potential candidate and active peers. It is worth noting that the overall quality at the receiver peer may not increase when additional sender peers are added because multiple sender peers may be connected behind the same bottleneck link. For this reason, we track each peer individually to measure its performances and capabilities, and then decide whether to activate sender peer or not, this selection mechanism is presented in section 4.1.

Real-time traffics are generally carried over Internet using Real-Time Transport protocol (RTP). P2P networks are widely used for multimedia streaming. Quality of the multimedia steams can be affected badly due to dynamic characteristics of P2P networks. Quality of video packets is influenced by available bandwidth, jitter/inter packets delay and packet loss rate. Inter packet delay/jitter plays major role in streaming applications. If jitter rate is high there will be distortion in the video which makes the user annoyed.

The topology for the construction of virtual P2P networks and signaling protocol are not considered in this paper. This topology can be ranged from single tree approach such as spreadIt, peercast, d3amcat, to multi-tree approach such as coopnet, splitStream, and finally mesh-based approach such as Narada and Yoid. Each approach has its advantages and weaknesses and we believe that the chosen topology is orthogonal to our adaptation mechanism that is presented in Section 4. However, peer selection approach is better applied to a centralized networks embedded in decentralized networks. This hybrid topology is realized with hundreds of thousands of peers in the Internet file-sharing system used in KaZaA [6] and Morpheus [8]. Mostly peers have a centralized relationship to a super peer called "supernode". All the queries are forwarded to this peer (super node) but instead of super nodes being standalone servers, they band themselves together in a Gnutella like decentralized network. Internet, email and SIP proxy also show this kind of hybrid topology. Mail clients have a centralized relationship with a specific mail server, but mail servers themselves share email in a decentralized fashion. By this way, each super node will have a matrix containing all peers actually connected to it. Each peer in the matrix is described by a set of QoS parameters (for instance available bandwidth, RTT delay, etc.).

The rest of this paper is organized as follow. Section 2 presents some related works, Multiple Description Coding scheme is described briefly in Section 3. In section 4, we present our proposed adaptation and Section 5 presents some performance evaluation and finally, we conclude in Section 6.

2. Related Works

P2P architecture is attracting many researchers and a lot of research activities are going on, in the domain of streaming over P2P networks. M. Hefeeda et al. [1] proposed a mechanism for P2P media streaming using CollectCast. They proposed an idea for collaborating with multiple sender peers for media streaming. A comparison is done for different selection techniques, i.e. "topology-aware selection" and "end-to-end selection”. In Topology-Aware selection technique all the shared communication links are considered for best selection of sending peers while in end-to-end selection technique these shared segments are not considered while selection of sender peers. Topology-Aware selection provides better results because it is based on congested links monitoring but it offers an overhead of considering each shared path in the network.

Reza et al, have proposed a framework PALS [2]. PALS is receiver centric framework, where a receiver coordinates delivery of layer encoded stream from multiple senders. A peer selection criterion has been proposed based on the overall effective throughput. There is no information available in the start so; initial peers are selected on random basis. They used layered coding video for the video transmission.

In [3] Padmanabhan et al. proposed system for the live and on-demand media streaming using MDC layers which presents better performance in flash crowd. Many other researchers have discussed different aspects of P2P media streaming. In [4] congestion control mechanisms using bandwidth estimation models have been proposed. In [5] a TCP-Friendly rate allocation algorithm for multiple sub stream coding combined with path diversity has been proposed where each sender sends different streams following different paths.

In this study, we proposed the mechanism of peer selection which is based on active measurements of network links. We follow the end-to-end selection technique. We introduce the cluster approach to avoid the risks arising from the sharing of same bottleneck link. The video is composed of MDC layers. We performed intensive simulations to test the adaptation mechanism. Results show that our proposed mechanism improves the overall throughput and enhance the overall received quality compared to system without adaptation mechanism. This enhanced throughput coupled with Multiple Description Coding improves the QoS remarkably.

3. Multiple Description Coding

Multiple description coding (MDC) [8] and Layered Coding (LC) are used for Audio/Video coding. In both schemes each description/layer contributes one or more characteristics of multimedia data [10]. Multiple description coding is a method of encoding the audio and video signals into many different streams. In Layered coding different layers are created. Base Layer is one of the most important layers while all other layers “enhanced layers” are referenced to base layer. The enhanced layers are not decodable independently to base layer. In MDC, a sub-set of descriptions is sufficient to decode the original file but there will be distortion if number of descriptions used is very small. By acquiring more descriptions, distortion can be lowered and it enhances the overall quality. MDC greatly improves error resilience because each description can be decoded independent to other descriptions. Even one description is sufficient to decode the Audio/Video signal with its minimum base quality. This feature of MDC makes it highly applicable for MPEG-4 video packets transmission over noisy networks/flash crowded networks when there is more possibility to loose more video packets. In CoopNet [3] as great efficiency for the use of MDC has been shown when used in flash crowd.

4. Adaptive mechanism for P2P packet video streaming

We are dealing with the problem of unicast, where a single receiver intended to receive media contents from many sender peers in P2P network. In this problem, selection of active peers become more important as, it is not feasible to select and coordinate with a larger number of active peers. It leads to extra overhead of establishing and monitoring of too many peers and also for reconstruction of all the video packets before decoding at the receiving end. In the start of the adaptive mechanism of P2P packet video streaming, the receiver node sends a query and get response from sender peers who intend to share the contents. It’s not necessary that all the nodes having requested contents must cooperate for content sharing. It’s a general observation that a large number of nodes present in P2P networks never intend to share their resources.

We assume that a lot of peers respond against the receiver peer query we named these peers as “Candidate Peers”. Receiver peer selects a sub-set of candidate peers to start streaming video packets. These selected peers are called active peers also shown is Figure 1. The mechanism for peer selection, peer activation, and stream switching is presented in the following sub sections.

4.1 Peer Selection Mechanism

Peer selection is an important part of media streaming in P2P networks as the dynamics and diversity between peers can vary with the passage of time which is effected by the facts 1) a sending peer crash/stop contributing the media content: 2) shared bandwidth is changed: 3) some new peer enter in the system providing better bandwidth share and low RTT (round trip time) value: 4) heavy traffic can cause more packet loss, high inter packets delay which ultimately causes low QoS (quality of Service). In the result of these factors, QoS is totally dependent on the intelligent peer selection and active monitoring of the network links between peers for detection of said changing and efficient stream switching to prevent its effects, stream switching is discussed in section4.2.

Receiver peer sends a query to search for the desire media content. In the response, receiver peer maintains a list of all the candidate peers with whom it can start streaming. For the selection of a sub-set of candidate peers, receiver peer diffuses a "*Hello*" packet to all the candidate peers. This "*Hello*" packet serves two purposes. First it behaves like a ping test to calculate the Round Trip Time "RTT" between receiver peer and targeted candidate peers and secondly it gets the information of targeted peer's super node. As we stated, we are considering the P2P architecture where some peer nodes are connected to one super node to form a cluster. All the requests pass through these super nodes. It's not necessary that this super node is not acting as server but it is used as transport node. This super node can be router or switch.

Receiver peer categorize all the candidate peers according to their "RTT" value and super node index. Receiver node selects a subset of these candidate peers having low "RTT" and which are belonging to different clusters. Selection of candidate peers belonging to different clusters is justified for the reason that all the peers present in this cluster (attached to same super node) share a common bottleneck link, so it is preferable to choose each sending peer from different cluster to avoid congestion over same link. This is an important feature in our adaptation mechanism.

Different weights w1 and w2 can be assigned for "RTT" and "TTL" between sender peers. If Pi represents candidate peers then, sender peer selection can be made using Eq 1.

$$\mathrm{P}i \approx \frac{RTT * w1 + TTL * w2}{w1 + w2} \qquad \text{(Eq. 1)}$$

For this study, we propose the selection criteria based on "RTT" after performing exhaustive tests to calculate some performance metrics such as "RTT" and "number of hops", details are not presented in this paper. "RTT" value is used in many mechanisms such as TCP-Friendly mechanisms and equation-based TCP throughput.

4.2 Stream Switching Mechanism

P2P networks are not reliable due to their dynamic nature i.e., any peer can enter or leave the network without prior notification. There may be problems of variations in available bandwidth and/or crashing of peers in the presence of flash crowd. Considering these problems receiver peer has to monitor all the active peers regularly for better performance of QoS.

Receiving peer monitor "RTT" value regularly between all active peers and itself. It's not desirable to switch for other candidate peer each time when there is low "RTT" to avoid oscillating effects. As multimedia streaming is real time application so for better QoS it's always encouraged to have low jitter rate, i.e., inter packet arriving should be constant and low. Our adaptation mechanism uses a low-pass filter to calculate a smoothed value of "RTT". Bursty traffic can cause a transient congestion. The "RTT" usage is not affected by this transient congestion since we shape this value. The low-pass filter is an exponential weighted moving average. The EWMA (Exponentially Weighted Moving Average) Chart is used when it is desirable to detect out-of-control situations very quickly. It is an Exponential Smoothing technique that employs one exponential smoothing parameter to give more weight to recent observations and less weight to older observations and vice-versa as presented in the Eq. 2. When choosing "λ", it is recommended to use small values (such as 0.2) to detect small shifts and larger values (between 0.2 and 0.4) for larger shifts [11].

$$X \leftarrow (1-\lambda) * RTT + \lambda * X \qquad \text{(Eq.2)}$$

A buffer is attached at receiver peer and before playing the media file, a reasonable amount of packets is received. A threshold value is set for the buffer. The value depends on the actual playing rate and packets arriving time. As we proposed MDC for data encoding ,so Receiver node receives different descriptions from active peers which are decoded after combining to achieve better quality.

We propose the stream switching for two cases, 1) if threshold value becomes lesser than that of desired value (50% of threshold value) receiver node must look for some other candidate peers. Stream switching is done by on/off mechanism new candidate peer is activated (on) sending a request and any of active peer which has now longer "RTT" can be deactivated (off). 2) For the second case stream switching can be done when any new peer node enter in the system having much lesser "RTT" value than that of existing active peers.

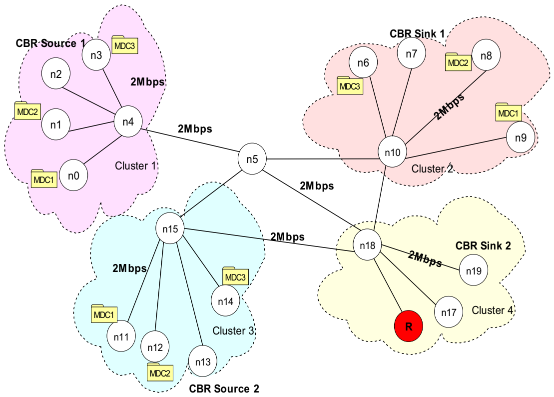

Figure 2: Simulation Topology

5 Performance Evaluation

This section presents the simulated results of the proposed adaptive packet video streaming mechanism. We performed intensive simulations to validate the results of our proposed scheme using NS-2 simulator [12].

5.1 Network Models

The network model considered for simulations is given in Figure 2. We distributed the original file equally among different cluster (cluster 1, cluster 2 and cluster 3). We attached a node "R" in cluster 4 to received real-time packet video. In this simulation, "R" tracks "RTT" value between each cluster super node and itself i.e. "RTT" from n4 to "R", from n10 to "R" and from n15 to "R". This enhances the scalability of the system rather than tracking each node individually. Each link in the topology is 2 Mbps bandwidth.

We activate a particular peer in one cluster depending on "RTT" value. Each sending peer sends different descriptions of original video file, which are reconstructed at receiver node "R". For our test cases, we have generated 3 different descriptions from MPEG-4 trace file containing different quality for the video [13]. These descriptions are generated by the fractions of DCT (Discrete Cosine Transform) matrix. We named these descriptions as MDC-1, MDC-2, and MDC-3. MDC-1 offers 50 % throughput of original file, MDC-2 offers 40 % throughput of original file, and MDC-3 offers 30 % throughput of original file. The overhead caused by MDC coding is about 20% of the original file. The overall video throughput of the different MDC layers is given in Figure 3.

We note that no source is providing 100% throughput but blessing of MDC scheme if receiver node receives all these three descriptions then it is possible to reconstruct original file with 100% quality. We attach all the descriptions to different sender and add CBR/UDP traffic to overload the network. "CBR source 1" is sending 512 bytes packet size with 1.5 Mbps. This source is attached to node n2 and sending UDP datagram to "sink 1" attached to node n7. "CBR source 2" is same as "CBR source 1" and it is attached to node n13. It sends UDP datagram to sink attached to node n19. "CBR source 1" is started at time 5 second, and stopped at time 55 second. "CBR source 2" is started at time 10 second, and stopped at time 50. The duration of the simulation is 60 seconds of time.

To compare the effect of our adaptation mechanism, we simulate two scenarios, also for making the scenarios simple, the topology is static. This means that no peer is leaving or entering the P2P network during data transfer.

Scenario 1: We run the simulation without applying any quality adaptation mechanism. In this case, there is no peer switching done even if a particular peer is going very congested by CBR traffic.

Scenario 2: We run the simulation with quality adaptation mechanism by providing peer switching based on active measurement of "RTT" between the super node and the receiver node. In this case, "R" selects and activates sending peers from different clusters.

5.2 Simulation Analysis

Figure 4 shows the received video traffic at node "R" in each scenario along with the expected video quality when using the three MDC layer. As we can see the adaptation allow maximizing the received throughput compared to scenario without quality adaptation. Even with quality adaptation, the received throughput is less than the expected one since the heavily stress the network with CBR/UDP traffic. The CBR traffic causes a lot of packet drops which are presented in Figure 5. The same comment is applied to this figure as the packet drop ratio is much lesser in scenario with quality adaptation compared to scenario without adaptation. Due to space limitation, we are unable to present the exact events when the peer switching is performed.

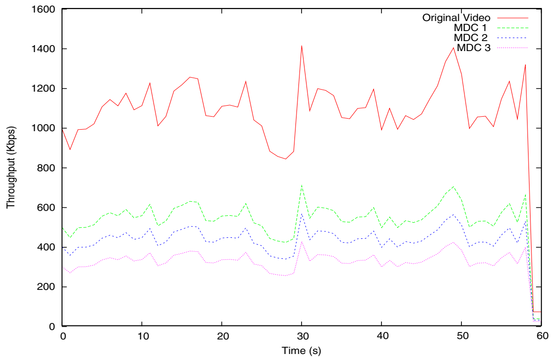

Figure 3: Video Throughputs for original and MDC Layers

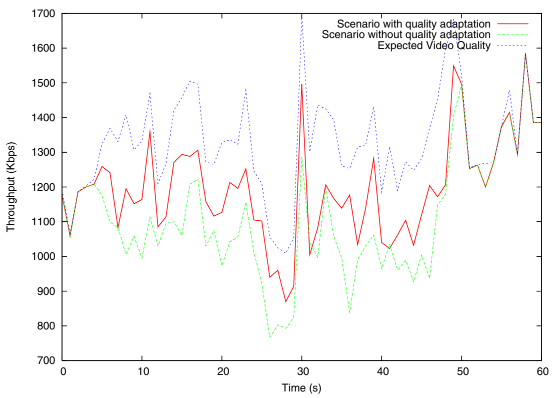

Figure 4: Received video at receiver peer in both scenarios

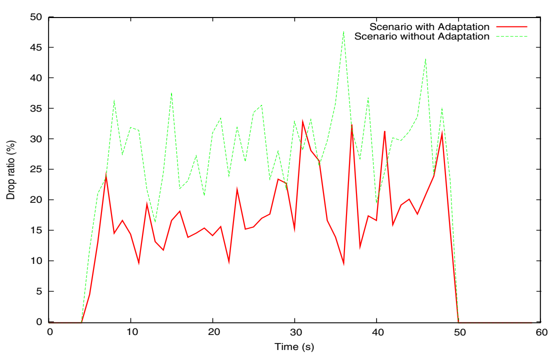

Figure 5: Drop Ratio of IP packet video in both scenarios

6 Conclusion

In this paper, we proposed a quality adaptive streaming mechanism for P2P networks. The presented solution based on active measurement of "RTT" value allows us to perform smooth quality adaptation for streaming of IP packet video. The mechanism used is receiver-centric i.e., receiver peer is in charge for selection of active peers and it also coordinates the overall streaming mechanism by switching from one congested node to other present in subset of candidate peers and offering better QoS. The simulation results show a noticeable improvement of received throughput and lower packet loss in the network. The network topology used in the simulation is static and does not take into consideration entering of new nodes that are offering better quality. Currently we are working on the dynamicity management in P2P streaming, where peers enter and leave the system more frequently.

7 References

[1] Mohamed Hefeeda, Ahsan Habib, Boyan Botev, Dongyan Xu and Bharat Bhargava, "Promise: Peer-to-Peer Media Streaming Using Collectcast", ACM MM'03, Berkeley, CA, November 2003.

[2] R. Rejaie and Antonio Ortega, "PALS: Peer-to-Peer Adaptive Layered Streaming", NOSSDAV'03, June 1-3, 2003, Monnterey, California, USA.

[3] V. N. Padmanabhan, H. J. Wang, and P. A. Chou, "Resilient Peer-to-Peer Streaming", IEEE ICNP, Atlanta, GA, USA November 2003.

[4] N. Aboobaker, D. Chanady, M. Gerla, and M. Y. Sanadidi, "Streaming Media Congestion Control using Bandwidth Estimation," In Proceedings o/MMNS '02, October, 2002.

[5] Joohee Kim, Russell M. Mersereau, and Yucel Altunbasak, "Network Adaptive Video Streaming using Multiple Description Coding and Path Diversity", ICME 2003.

[6] KaZaA: http://www.kazaa.com

[7] Morphous: http://www.morpheus.com/

[8] J. Kim, Russell M. Mersereau, and Yucel Altunbasak, "Network Adaptive Video Streaming using Multiple Description Coding and Path Diversity", ICME 2003.

[9] V. K. Goyal, "Multiple Description Coding: Compression Meets the Network" IEEE Signal Processing Magazine page 74-93, Sep 2001.

[10] A. Vitali, M. Fumagalli, and Cefriel, "Standard-compatible Multiple-Description Coding (MDC) and Layered Coding (LC) of Audio/Video Streams", Internet Draft, Network Working Group. July 2005.

[11] J. Stuart Hunter. "The Exponentially Weighted Moving Average" J Quality Technology, Vol. 18, No. 4, pp. 203-207, 1986.

[12] Network Simulator 2: http://www.isi.edu/nsam/ns/

[13] O. Rose, "Statistical properties of MPEG video traffic and their impact on traffic modeling in ATM systems", in Proceedings of the 20th Annual Conference on Local Computer Networks, Minneapolis, MN, 1995, pp. 397-406.