Adaptive Video Streaming: a Survey and Case Study

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Abstract

In the past decade, Internet traffic has seen a significant change from web browsing to video viewing. The ongoing trend raises a challenging problem: how to stream data to heterogeous peers?

The designer of such data streaming architecture should bear the following considerations in mind: QoE, server load, network resource efficiency, scalability, etc. The heterogeneous peer network condition makes the design more complicated. The underlying codec ranges from Multi Description Coding to Multi Layer Coding. The data deliver architecture ranges from unicast, multicast, to P2P network. Researchers have focused on different system settings and optimization objectives.

This paper will first sum up several works in the context of adaptive video streaming. At the same time, we do a case study on a commercial adaptive video streaming system, which combines Multilayer Codec and P2P technology. Possible improvements on this system are proposed with reasoning. Some of the conjectures are verified through a corresponding simulation platform based on NS2.

Contents

1 Introduction

1.1 Background

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1.2 Paper Structure

In section(2), we propose 5 metrics and present a very general model of system design as well as several variations. In section(3), we propose three problem scopes involved in adaptive video streaming. We address the full design from three aspects: codec, networking and playback. In section(4), we focus the networking on p2p technology, which is considered a very powerful tool to provide a system with huge aggregate bandwidth in a distributed fashion. A design consideration list is given from bottom to top. In section(5), we do a case study on a widely used P2P simulation platform[16]. There are not so many highlights from this case study. Nevertheless, we conclude some experience in section(6), and propose some potential directions in section(7).

2 General Model

2.1 Metrics

No matter what our target system is (live streaming/Vod; multicast/P2P; layered/or not), we propose 4 general system metrics:

(QoS) QoS is one of the traditional considerations. Typical metrics in video streaming is buffer ratio, startup delay, jitter, etc. In [39], Xiao proposed four metrics in the context of multilayer P2P video streaming: throughput, layer delivery ratio, useless packet ratio, and jitter(subscribe/drop layer events).

(QoE) Note that all systems should benefit their users in the first place. However, QoS can not directly reflect user experience. User experience can be measured using Mean Opinion Score[33] or User Satisfaction Index. The former one is subjective, which involves field test on a batch of users. The latter one is objective, which combines several QoS metrics and output one overall score. Current works in adaptive video streaming seldom consider QoE carefully. It's obvious that a QoE aware design is highly appreciated.

(Vendor cost) Cost of system vendor is often considered as the main optimization objective. Some recent VoD works like [37] and [43] fall into this category. In those works, server is modeled as a backup of any missing

chunks. Thus the QoE(only discontinuity here) is guranteed naturally. With this setting, people can try to lower the server load.

(User cost) User resources, like CPU cycle, network bandwidth, etc, are all precious. If QoE can be guranteed using less resources, it's not a good idea to consume more. Computing resource is prominent if we want to upgrade to a complex strategy. Some strategies look good in model, but may not be applicable due to very high time complexity. For example, in [41], Zhang proposed a Mincost Flow based technique, to solve the chunk selection problem. This technique is universal and can solve all similar problems with utility definition. However the time consumption in real deployment is not fully addressed. We believe it consumes more time than the Knapsack based technique proposed in [9], although the problem definition in the latter work is narrower. In many real deployments, we sacrifice the quality of solution for tractable computation resource. As is given in [9], an efficient heuristic based algorithm can perform as nearly equally good as the optimal algorithm. In this case, the heuristic basd algorithm may be more likely to be adopted by real system designers.

(System cost) In a P2P system, besides the bandwith consumption at server and peer side, the flow in intermediate netowrk infrastructure should also be taken into consideration. In [5], Chu use link stress as one of the metrics, which measures how much redundant flow tranverse a single physical link. This work is believed to be one of the pioneering work in P2P, so the author considers system cost thoroughly. Later, P2P technology has been proven an efficient method in many real deployments. The following works focus more on client and server side, rather than on the network system.

As far as we know, none of the current works consider all four metrics in design, although many works can achieve good result in those metrics even if they are not considered from the very beginning.

2.2 Model 1

One optimization abstraction is:

$$\begin{array}{rl} \text{maximize} & \text{QoE} \\ \text{minimize} & \text{Cost} \end{array} \tag{1}$$

QoE and Cost alone can be multi-dimensional. As is given in last section, cost can be categorized into vendor cost, user cost, and system cost. Note that this multi-objective model is intractable since the two groups of objectives are opposite to each other most time. Two simplifications are given in the following section.

2.3 Model 1(a)

By constraining cost to be less than a certain level, the original model becomes a single objective optimization. C_0 can be chosen such that the cost is approximately (but not necessarily) minimum.

$$\begin{aligned} \text{maximize} \quad & \text{QoE} \\ s.t. \quad & \\ \text{Cost} \quad \leq \quad & C_0 \end{aligned} \tag{2}$$

This philosophy is widely seen in many works. i.e. set C_0 to be the largest amount of resources available, and then optimize QoE(or QoS for simplicity) metrics. It is not seen that people constrain resources (i.e. only use half download bandwidth) when QoE is not the maximum.

2.4 Model 1(b)

By constraining QoE to be larger than a certain level, the original model becomes a single objective optimization. Q_0 can be chosen such that the QoE is approximately (but not necessarily) maximum.

$$\begin{aligned} \text{minimize} \quad & \text{Cost} \\ s.t. \quad & \\ \text{QoE} \quad \geq \quad & Q_0 \end{aligned} \tag{3}$$

This philosophy is widely accepted in the degisn of VoD. Zhou proposed a statistical model[43] to analyze the optimial replication of VoD. In that work, QoE is guaranteed naturally by assuming the server as a backup at any time. So the rest of that work aims at minimizing server load.

2.5 Comments on Models

Theoretical bound deduction is not seen in the scope of video streaming, let alone adaptive video streaming. In terms of adaptation, we want to know how the tradeoff between quality sacrifice and other metrics is, given certain resources. This is generally Model 1(a), with QoE decomposed into several parts.

3 Problem Scope

We divide the problem from bottom up into three parts:

- Codec. It determines how the video is coded, and what kind of adaptation it can have. This is not our research focus. However the choice of codes influences what we can do in the networking design.

- Networking. This part solves the problem that how we deliver data from server to all receivers efficiently, and meet certain playback requirements at the same time. Most of our work lies in this part.
- Playback. User actions and playback decisions are studied, so that we have a more clear direction when designing networking solutions. Again, this part is not our focus, but gives methodology and tool to facilitate networking design.

3.1 Codec

There are two types of adaptive codec:

- Multi-Description Coding(MDC). MDC has been studied long ago in the field of information theory[13]. The basic idea is to decompose video data into multiple descriptions. Descriptions are independent of each other. With more descriptions received, the receiver can playback higher quality video. For a good survey, interested readers are recommended to [34]. Some application of MDC in video streaming appeared around 2000, to name a few classical works, [2], [1]. In [1], Apostolopoulos combines MDC with CDN technology to provide multiple route for video streaming, thus higher achieving higher error-resilience.
- Multi-Layer Coding(MDC). MLC is different that data is organized layer by layer. Higher layers have dependency on low layers. H.264/SVC[30] is the extension of H.264/AVC[35], which provides three types of scalability: spatial scalability, temporal scalability, and quality scalability. In academic study, temporal scalability is frequently used, for the resultant chunk size is a pyramid look structure. The temporal scalability relys on temporal predictions, in a I-P-B frames fashion. I-frame and P-frame is often put in the base layer, and B-frame forms enhancement layer(s) typically. The typical frame size is: I-frame $>$ P-frame $>$ B-frame. In this case, base layer is usually larger than enhancement layer. In industrial deployment, spatial scalability is more preferred in heterogeneous device distribution. i.e. a cellphone users have smaller screens. There is no incentive for them to download full sized version. If mere temporal scalability is used, they have probably have no choice but to download the full sized base layer. The video will not playback smoothly. However, if spatial scalability is used, they can afford the smaller-sized version with better smoothness.

Some important characteristics of the two types of coding are summarized in table(1).

Another important consideration of codec is whether it's variable bitrate or not:

Table 1: Comparison of MDC and MLC

Item	MDC	MLC
dependency	independent	dependent
coding efficiency	lower	higher
real deployment	less	more

- Variable Bit Rate(VBR) may achieve higher coding efficiency. i.e. when the video is presenting a series of consecutive static scenery, it can be coded very efficiently using motion vector compensation. The resultant bit stream may consist of a big I-frame followed by many small P- and B- frames. Observed in time domain, the bitrate is variable.

- Constant Bit Rate(CBR) works at making the bitrate nearly a constant. A common way is to add a feedback buffer to VBR coding framework[19]. The encoder will adjust its quantization level according to buffer status, and make the overall bitrate oscillating around a constant. Two common problems are buffer overflow and buffer downflow.

According to our survey, CBR is commonly used in previous works. This is because the constant rate makes system design easier. i.e. in the desgin of adaptive video streaming, people always try to prevent jitter (frequently switch between higher and lower quality). If the codec is VBR, the receiver may observe a "better bandwidth" when it downloads some chunks faster. However, the real reason is the lower video bitrate during that period of time. If CBR is used, number of chunks downloaded in unit time can better reflect the network condition. Thus, better control over quality adaptation can be achieved.

3.2 Networking

We discuss three types of fundamental networking choices in this section: multicast, unicast, P2P. In real systems, designers may combine more than one methods together. i.e. in P2P VoD system desgin, peers can turn to server for help. Some systems treat server as a normal peer. This simpifies the system and is widely accepted in many studies. However, server exihbits much larger capabilities than normal peers. Distinguishing the server from normal peers and do some specific optimiztion may enhance the system. In one measurement study of popular online video websites[47], people found one of the largest vendors combines unicast and P2P together. For hot movies, content is mainly streamed from peer to peer using UDP. For cold movies, content is mainly streamed from server to peer using TCP.

3.2.1 Multicast

The first stream of adaptive video streaming comes in the IP multicast streaming. The reason is multicast has long been believed to be the solution of streaming large amount of data. It has the property that no redundant data tranverse a single link more than once.

In 1996, McCanne [24] did some pioneering work in the desgin of adaptive multicast streaming. He utilizes multiple multicast groups, which transmit different layers. Peers subscribe and unsubscribe layers explicitly by joining and leaving corresponding groups. Peers do joining experiments in a shared learning manner.

According to the number of multicast groups, we categorize those works into three classes:

- Single group. The server sends video data through a single multicast group. In this case, every receiver will get the same quality of video. This is unfair for users who have better network condition. Adaptation is generally performed at server side. Users can feedback their receiving rate/ratio to the server.

- One group per receiver. This is another extreme case compared with single group. The system uses as many groups as receivers. In this case, the multicast mechanism degrades to unicast. We'll discuss more about the adaptation of unicast in the following section. With this design, largest fairness is achieved. However, the intermediate resource utilization is not very high, since many redundant flows tranverse a single physical link.

- Several groups. This is called Simucast in [19], and also the fundamental choice of [24], [3], [4], etc. This method seeks for tradeoff between fairness and efficiency. The adaptation can be sender-driven, like [3], or receiver-driven, [24], and is generally performed by subscribing/unsubscribing different groups.

For a full survey till 2003, interested readers are recommended to [19].

However, due to lack of globally deployed multicast infrastructure, the multicast based adaptation technique is not used to stream video data to public audience.

3.2.2 Unicast

With the rapid growth of Internet, networking infrastructure becomes more and more powerful, so that they can support large amount of simultaneous unicast streams. One of the advantages of IP multicast, efficiency, is no longer a major concern. Besides, unicast has other extra advantages over

multicast: easy to control, can be connectionless or connection-oriented, better error resilience, etc.

If unicast is used as the networking solution. The problem of adaptive video streaming degrades to a flow control problem, which has been well studied, like TFRC[10].

In 1999, Rejaie designed a mechanism in the context of unicast to stream scalable data to receiver[28]. They utilize layered coding and Additive-Increase-Multiplicative-Decrease mechanism. Their design works without any assumpations of loss patterns.

In 2001[2] and [1], respectively, Apostolopoulos designed another series of representative unicast adaptation. Underlying codec used in their works is MDC. Since MDC itself is error resilient, the emphasis of those works is to design multi-path mechanism. If the data is streamed through a single path, loss of data ususally happens in a batch. Multi-path can make the streaming service more robust. Three of the proposed solutions are summarized below:

- Source routing[2]. Ordinary IP route is destination based, but the source can specify a series of intermediate nodes in the option section of IP header. Multiple paths can be achieved with the knowledge of network topology. However, this can never be a real story, since topology is confidential to ISPs.

- Relay network[2]. This is a kind of proxy based approach. Video data will first be streamed to multiple intermediate nodes. Those nodes can be semi-intelligent nodes, which simply decapsulates the IP packet and drop it back to the network again. Or the nodes can be full-intelligent nodes, forming an application level overlay. The latter one is the preliminary form of P2P, so we'll discuss about it later. Relay network is applicable, but it creates multi-path at the cost of higher delay(for extra hops of packets).

- Content Delivery Network(CDN)[1]. CDN emerged in the early 2000s. With thousands of edge content servers, the connections between client and the CDN is naturally a multi-path structure. In [1], the authors deploy different descriptions across different content servers. Clients will connect to several near content servers. This framework is error resilient.

A very simple but non-trivial example of unicast adaptation is like Youtube [44]. It provides several quality choices for the users, like low definition, medium definition, and high definition. This is rigid adaptation using priori knowledge by users. While flexible technical adaptation solutions are wide studied, the most widely deployed adaptive streaming follows this trend.

3.2.3 P2P

Since the pioneering work of [5], P2P technology emerged in the past decade. Several works like [6] extend those ideas used in mulilayer multicast streaming and use P2P technology to form the underlying overlay.

In [7], Dai maximized throughput of P2P network using commodity flow algorithm. Although the title contains "layered", their work is more likely applicable with MDC. This is because dependency between layers are not formulated in the model. Also, they address the inter-layer fairness by allocating resources proportional to the demand of the layer(better known as chunk size of each layer). In this work, each layer has its own overlay network. This differs from most counterparts.

In 2002, Khan proposed the algorithm called HEU[18]. He model the adaptive multimedia system using Multi Dimension Knapsack Problem(MMKP). HEU is a fast heuristic algorithm which can give suboptimal solution within 6% sacrifice to the optimal one. HEU proceeds by first choose one item from each group to get an initial feasible solution and iterates by exchanging item in each group to approximate optimal solution.

PALS[29], proposed by Rejaie, follows their prior work in the context of unicast in 1999[28]. Many following works follow the framework used in this paper. A 2D sliding window is used to smooth the transient bandwidth variation. How to fill this sliding window(also known as buffer map in other works) is the chunk/peer selection strategy. Chunk selection and peer selection is closely related with each other. People come up with different design. Some of the works prefer peer selection first to present clear picture of chunk availability for chunk selection strategy. Other works prefer chunk selection first and assign chunk requests to available peers according to measured network condition. The latter one is fine-grained, and is adopted in PALS. As for overlay construction, PALS simply uses a random based initialization and random based update.

In 2007, Liu used MDC to provide incentives in an open P2P video streaming system[20]. Their work roots from the observation that even in a short period of time, there are bilateral flows between each pair of peers [15]. In this case, the mature Tit-For-Tat strategy used in BT can be adapted to the context of adaptive video streaming. With the help of MDC, peers can get something to playback while being frequently choked or unchoked. Thus poor peer won't starve to death as is seen in many P2P file downloading systems. One of the following work is [21], which utilizes MLC rather than MDC. Compared with [20], [21] claims MLC has higher coding efficiency. In 2009, the same group of researchers extend those work to address a full layered P2P system[22].

In [27], Qin presents some work in the context of mobile adaptive video streaming. Difference from wired network comes in many aspects: users are more interested in general content rather than specific picture quality;

mobility models like random waipoint or random walk should be considered; need to support certain wireless infrastructure characteristics, like 802.11 Auto-Rate Fallback.

LION[42] uses alternative paths and network coding to enhance layered P2P streaming. One of the drawbacks is that layers are considered in isolation. Note that their only optimization objective is throughput, this may results in some higher layer useless chunks, since layer dependency is not fully considered.

In 2008, Xiao designed a finer tuned overlay called OCALS [40]. Traditional overlay construction is done with the help of a centralized tracker. i.e. Chainsaw [26] a random based overlay construction method. In OCALS, peers enther the system using a few landmark nodes, and explore the overlay in a BFS-like manner. They construct different overlays for different video layers from bottom up. Neighbours in lower overlay networks are reused if they also subscribe to higher layers. They overlay is expanded with the consideration of both network condition and data availability. One edge is added only when this edge can help either side of the peers, or there are no other choices.

In 2009, Xiao proposed the full design of a layered P2P system[39]. The overlay construction is OCALS[40] described in above paragraph. This work focus on buffer management and chunk selection. They proposed a 3-section based framework. In each section, the strategy targets at different optimization goals(metrics). The biggest contribution in this paper is that they proposed 4 metrics specialized for adaptive video streaming. Metrics like throughput is widely adopted in many P2P system evaluation. Others like layer delivery ratio, useless packet ratio, and jitter(layer subscribe/unsubscribe events). However, this work is majorly heuristic. The definition of utility is crucial in the optimization process.

Recently, Eberhard [9] followed the Knapsack Problem based algorithms proposed in [31]. They add some simulation and evaluation to the original work. They aims at optimizing a given utility function. Although there is no significant improvement compared with baseline (PALS,[29]), they have one highlight that greedy algorithm of KP performs nearly as good as more complicated algorithms but time complexity is much lower. There are some further improvements to this algorithm. First, the greedy algorithm doesn't require complete sort. Second, with a clear definition of utility, the chunk selection priority can be deduced directly, and no sorting is needed. Either ways can lower the client side cost.

3.2.4 Miscellaneous

Besides those discussion above, other related topics are:

- Where to perform data drop? There are two major approach. One is to drop data at end hosts. Peers keep measuring network condition and

decide how much packets to send to/ request from other peers. This is active flow control, studied in many previours works. The other approach relys partly on intermediate devices. If the intermediate devices can distinguish between different priorities, we can set lower layer data to higher priority and vice versa(assume MLC is used). This is not possible in current infrastructure, but we've seen a chance in IPv6, which use traffic class to support priority differentiated service.

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(reference above)

3.3 Playback

The last concern in adaptive data streaming is playback. It can be divided into two parts:

- Playback decision. Playback decision influences the design of networking. However, the author of this paper never see complete study of playback strategy, especially in the context of adaptive video streaming. Many of the works presented above use simple rules for playback decision. Most of them even don't mention it in the design of strategies. One common example is: wait for a chunks are ready in the buffer; pause the video when buffer downflows; after the buffer is refilled with b chunks, continue to play again. If adaptive methods are introduced, say multilayer, the decison becomes more complex. Basically, the player should decide to wait for higher layer chunks or just play in sacrifice of certain quality.

- User model. In order to make rationale playback decisions, QoE study is essential. In [33] Wang uses subjective tests to map QoS parameters to QoE metrics, Mean Opinion Score. The drawback of this kind of approach is lack of scalability. Instead, Dobrian initiates the use of mass objective evaluation in video streaming[8]. They measure correlation and information gain between different QoS paramters and user engagement. Although [8] is scalable and the result is more reliable, their user model is preliminary. The only measure of user satisfaction level is engagement. Since there are no widely deployed adaptive video streaming system, we lack user action data. With those per action recorded data, maturer model from other disciplinaries like web serach, portal website, etc, can be borrowed to analyze QoE in adaptive video streaming. This helps us better understand the preference of users, and thus make more reasonable strategies.

4 Design of Adaptive P2P VoD System

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4.1 Codec Choice

4.2 Transimission Protocol Choice

4.3 Overlay Construction

4.4 Peer Selection

4.5 Buffer Management

4.6 Chunck Selection

4.7 Playback Decision

4.8 User Model

5 A Case Study and Simulation

Our simulation platform roots from a commercial deployment of P2P system[46], which has upgraded from P2P file downloading system, to P2P live streaming sytem, and then to the current version of multilayer P2P VoD system. Interested readers are recommended to [16] for further information. This platform has been validated against real trace data collected from 2008 Olympic period. It is widely used in P2P related research and pre-deployment testing. Course IERG5270[47] has adopted this platform as a simulation foundation for several years.

5.1 System Description

To get a concrete view of the situation, we draw the version tree in fig(1). Blue blocks represent the real deployment of ASTRI. White blocks represent corresponding simulation platform. Red dash stands for strong equivalency. Green block shows the position of this project. The trial version of multilayer P2P VoD is forked from the original downloader platform and is developed by Zheng Wen in 2010.

Precaution: the results obtained in this project may not reflect real system in every aspect(there is no red dash showing this kind of relationship). Nevertheless, methodology wise speaking, those results do provide some insights in Multilayer P2P system design.

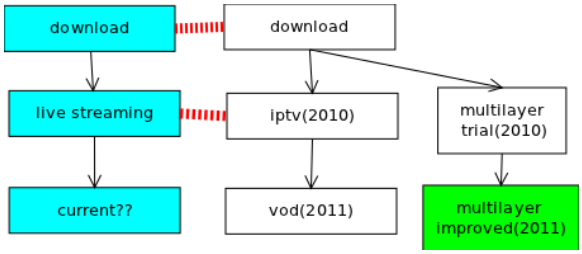


Figure 1: Version Tree of Real and Simulation

5.2 System Benchmark Test

Before we start, system benchmark test of simulator performance is conducted.

Fig(2) shows how simulator running time varies with different number of cores. Note that PDNS [49] is the distributed version of NS[48]. In this paper, we just run the simulation on one machine with 16 cores. The original configuration of the platform involves 8 cores per simulation instance. Leaving some system recourses for other purpose, we can only run one simulation at a time, which significantly lowers our experiment efficiency.

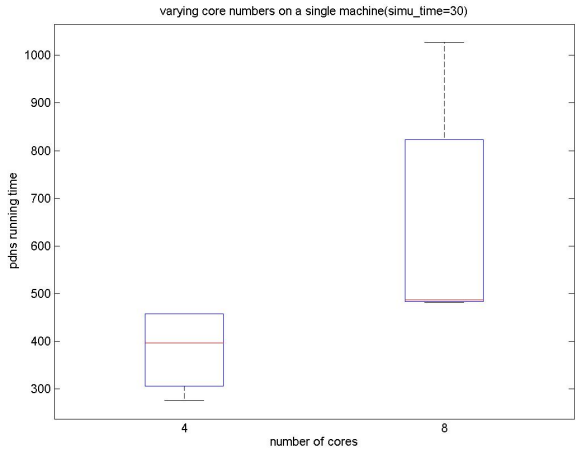


Figure 2: Running Time v.s. Number of Cores

We find that, when simulation time(measured in NS seconds) is small, 4 cores appear to be more efficient than 8 cores. Thus the following experiments are all issued to 4 cores to enhance our distributing level.

Fig(3) shows how simulator running time varies with the number of nodes simulated. It appears to be linear. The fit parameters are given in the title of the plot.

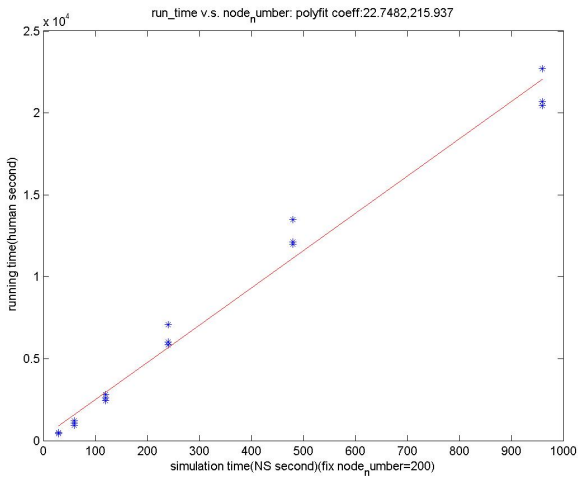


Figure 3: Running Time v.s. Number of Nodes

Fig(4) shows how simulator running time varies with the simulation time(in NS seconds).

With those system test, the final parameters we choose are:

- Number of Nodes: 160
- Simulation Time: 300 (NS seconds)

It's obviours that the number of nodes and simulation time is much smaller than those in a real VoD system. With those two configurations, the simulator can finish in about 2 hours using 4 cores. We sacrifice some equivalency to the real system to make the simulation tractable.

5.3 Experiment Configuration

In this study, we follow the same configuration given by Zheng Wen. The topology is summarized as follows:

- Star like. Peers are equally divided into 4 subnets. Every node connects to a single central router.
- Subnet parameters:
  - Subnet1: down:10Mb; up:10Mb.
  - Subnet1: down:3Mb; up:0.5Mb.
  - Subnet1: down:3Mb; up:0.5Mb.
  - Subnet1: down:3Mb; up:0.5Mb.

The layer configuration is summarized as follows:

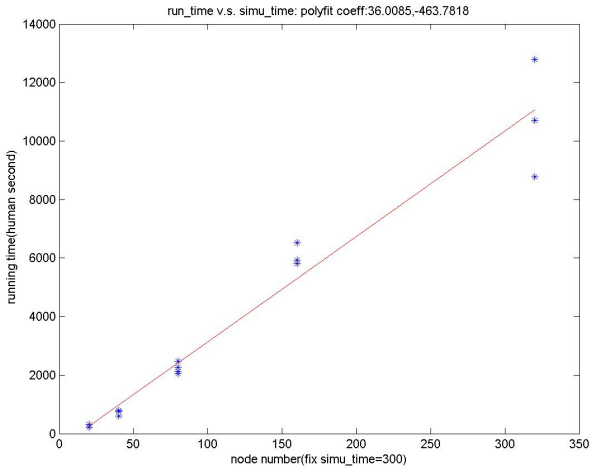


Figure 4: Running Time v.s. Simulation Time(NS)

- 3 Layers.
- Layer parameters:
  - Layer1: 256Kbit. (per piece)
  - Layer2: 256Kbit. (per piece)
  - Layer3: 512Kbit. (per piece)

Note that the cumulative data amount is 256Kbit, 512Kbit, and 1024Kbit. This is consistent with Wang's QoE study[33]. Thus we can rely on their QoE model to measure our system.

5.4 QoE Model Implementation

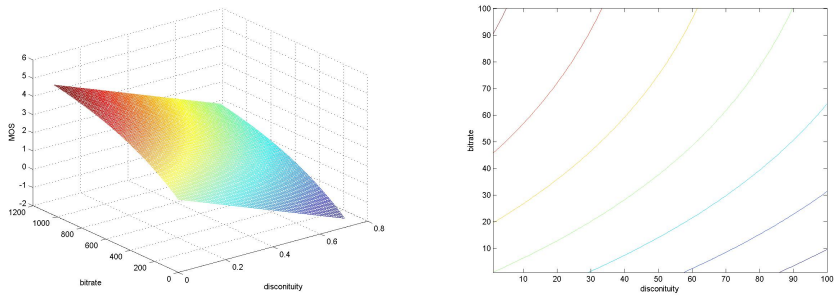
In Wang's paper[33], they seek for the tradeoff between bitrate(influenced by current layers subscribed) and discontinuity. They first do large amount subjective tests, and then fit the data into a continuous version formula, as is given in eqn(4).

$$MOS = c_1 \times d + \alpha \times (1 - e^{-b \times \lambda}) + c_2 \tag{4}$$

The parameters are $c_1 = -5$, $c_2 = 2$, $\alpha = 4$, $\lambda = 0.0015$.

In that paper, the maximum discontinuity considered is 50%, so we plot the MOS curve again to give the readers a better impression of how it looks like. See fig(5).

Fig(5(a)) is the 3D version and Fig(5(b)) is the contour version. We want to point out that by definition, MOS should be a score between 0 and 5. However, using this formula, we'll get negative scores sometime. Especially



(a) 3D (b) contour

Figure 5: Conclusion, QoE and Performance

when the discontinuity is 100% and bitrate is 0, the MOS can be as low as -3. We don't regard this case as a bug. On the contrary, it is reasonable to assign negative opinions. In our implementation of this QoE model, peers report their statistics every 20 seconds. If the -3 score happens, that means the user is purely waiting in the whole 20 seconds, which causes a negative opinion naturally. Our system should work towards reducing this kind of disturbance.

5.5 Baseline Test

In this section, we evaluate the baseline using the QoE records reported by peers. We compute MOS using eqn(4), and take the average.

Fig(6) shows the result when total simulation time varies from 30 NS seconds to 960 NS seconds. It's interesting that with the increase of simulation time, the QoE gets improved, and the boxplot shows the statistics of 5 simulations with the same configuration. This means the improvement is statistically significant.

By examing the detailed trace file, we find that there are many negative opinions at the beginning of those simulation. Later after about 100 NS seconds, the reported MOS gets better and better. It makes sense that the system needs some bootstraping time, when all of the peers lack chunks. They can not help each other efficiently during this time.

We eliminate those negative records and make the boxplot again. See Fig(7). This figure verifies our conjecture. The last two boxes stand for 480s and 960s simulation time. There is little difference in mean within this range. Longer simulation time just makes the variance smaller. To explain, after some bootstraping period, the system approaches steady state and the average opinions become nearly a constant. The first box seems an outlier. This is because peers are joining the system gradually at that time. The number of reported records are very small. Only those peers connected

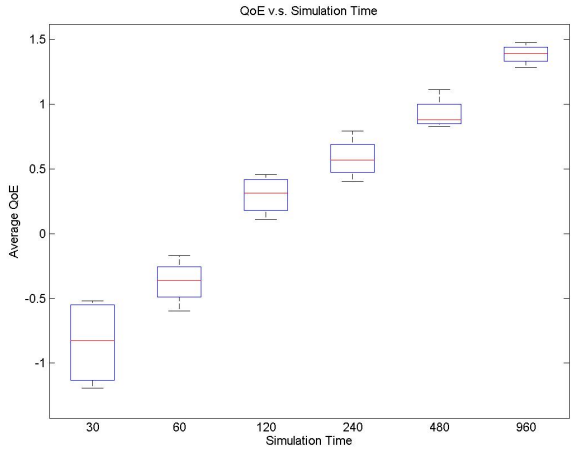


Figure 6: QoE v.s. Simulation Time

directly to the source can get something to playback. Others all get negative opinions. By eliminating the negative opinions, the resultant score is very high.

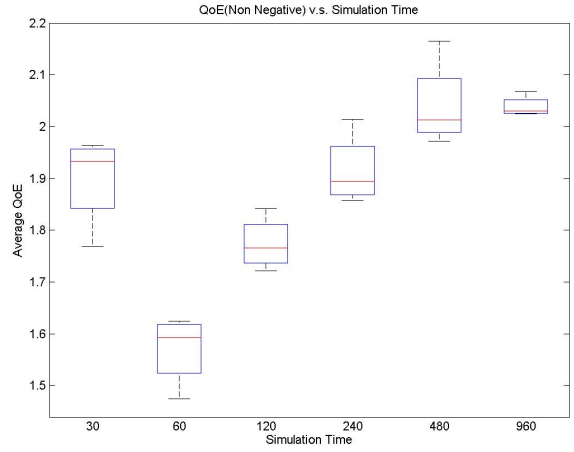


Figure 7: QoE(nonneg) v.s. Simulation Time

Next, we pick a sample run out of all these baseline simulations. The global parameters and statistics are: simulation time is 480s; number of nodes is 160; number of QoE reports is 3749; number of nonnegative QoE reports is 2601; average QoE socre is 0.87; the nonnegative version of average QoE score is 2.02.

Fig(8) shows how QoE varies with curret time section. We can see the

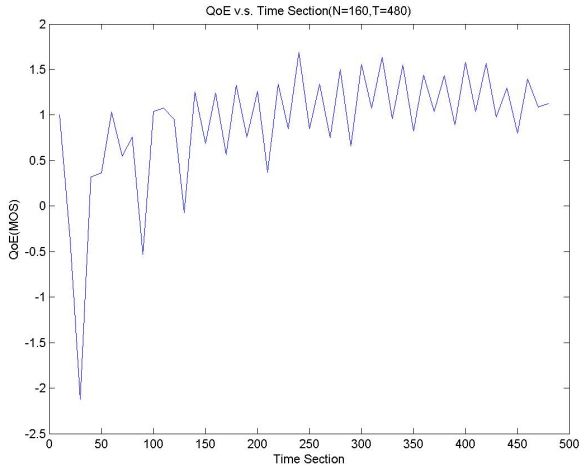


Figure 8: QoE Varies with Current System Time

per section averaged QoE gets higher and higher in the first 150 seconds. After that, the section averaged QoE becomes steady. This again proves our bootstraping conjecture.

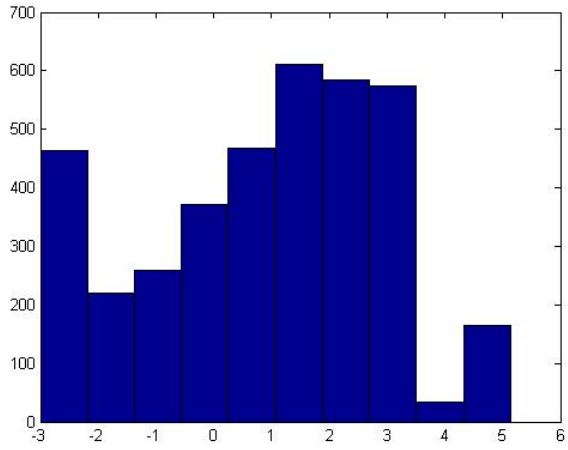


Figure 9: Histogram of QoE Distribution

Fig(9) shows the historgram of QoE reports from this sample run. We can conclude that, some powerful peers can get full 5 score experiene, whereas large amount of peers report negative scores. Our future improvements should mitigate this part(purely buffer for 20 seconds and play nothing).

5.6 Chunk Selection Architecture Reconstruction

In this section, we modify the original architecture to make it more clear and unified. This helps us develop strategies on this architecture later.

Original design separates chunk selection and peer selection. It is shown in Fig(10).

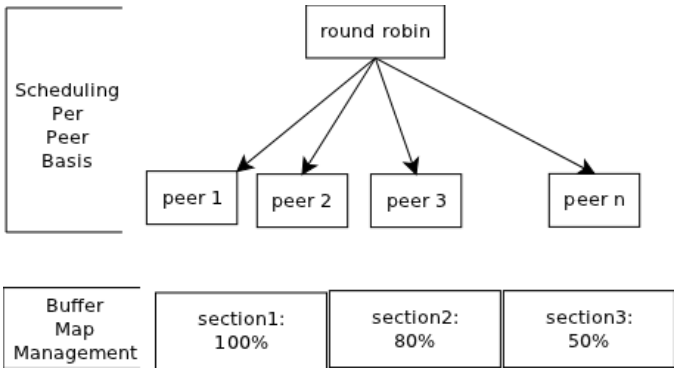


Figure 10: Original Architecture

In this design, every peer is considered in a round robin fashion. After one neighbour is chosen, this peer selects chunks according to a three section based strategy. After one section is filled up to a certain ratio, the peer will consider the next section. Chunks in the same section are selected randomly. In our multilayer situation, layers are considered from lower ones to higher ones. That is, after the peer get 50% of the 3rd section, it will start to to fill the next higher layer. In order to reduce the network burden, peers are constrained to fetch chunks from no more than 20 neighbours simultaneously.

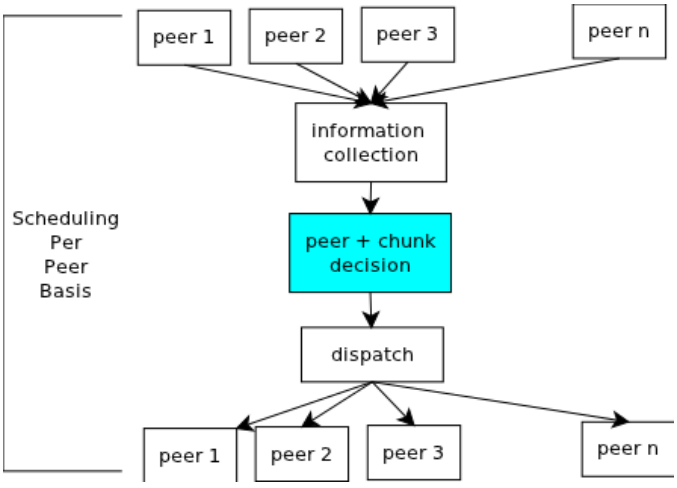


Figure 11: Improved Architecture

The improved architecture is illustrated in Fig(11). To make our decision more rationale and make the architecture clear, we collect all information first, and do peer selection and chunk selection together.

In this version, we adopt a very simple strategy, as is used in PALS [29]. This strategy makes decision on a chunk basis. It first consider base layer chunks and then moves to enhancement layers. In one layer, the chunks closer to playback points are considered first. The overall picture is like a snake-shaped move on the 2 dimension buffer map. Fig(12) illustrates this strategy. If there are more than one peers available for one chunk, we just randomly choose one peer.

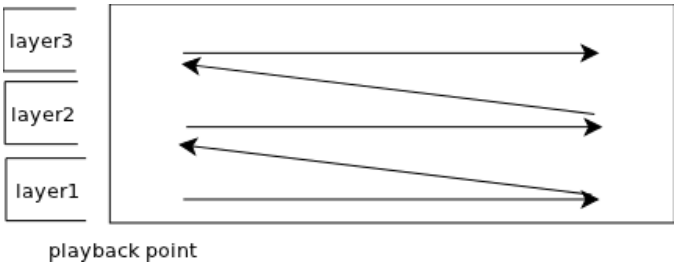


Figure 12: PALS Style Chunk Selection

After each modification, we evaluate the result from two aspects:

- QoE. The formula is given in eqn(4). In our study, we didn't make any aggressive strategies like scheduling all requests to the server. So we don't subtract server's contribution from this metric intentionally. Another reason is, in the underlying overlay construction(which is not modified in this study), the server is treated as a normal peer. Not all peers can turn to server for help when the lack chunks.

- Performance. Another important thing we care is the performance of simulator. This kind of packet level simulation is very time consuming. One iteration of strategies calls for several hours of running typically. In order to fulfil a short-term project, we can not make strategies of high time complexity. What's more, the running time of simulator reflects the computation resource consumption in real deployments. User experience will degrade if their player consumes extra large amount of resources, to limit other process.

The result of modification in this section is:

- QoE: $0.7 \rightarrow 3.3$

- Performance: nearly doubled.

5.7 Priority Based Upgrade

In this section, we do more engineering upgrades. We abstract the chunk selection strategy using a priority table. One sample table with window size equal to 5 is given in Table(2). This table is equivalent to PALS.

Table 2: A Sample Priority Table with Window Size = 5

	t1	t2	t3	t4	t5
layer3	11	12	13	14	15
layer2	6	7	8	9	10
layer1	1	2	3	4	5

Further investigation on optimal priority table can be done easily in this framework. Result of this modification is:

- QoE: nearly the same.
- Performance: decrease slightly.

5.8 Scalable Window Size

After previous upgrades, we examine the trace files from several simulations. From the trace, we find many powerful peeers can get full 3 layers in the whole window(60 seconds, 3 layers). The restriction on window size is a special desgin of video streaming, no matter live or VoD. This is because chunks very far from playback point is not of high interest to the peer. Every peer only needs to store enough chunks in the buffer to guarantee smooth playback. However, there is a side effect. Capable peers who are playing the early part can not help others who are playing the later part. Thus we want to give those peers a chance to help others. This is the rationale behind scalable window size.

Table 3: A Sample Priority Table with Window Size = 5+5

	t1	t2	t3	t4	t5	t6-t10
layer3	11	12	13	14	15	...
layer2	6	7	8	9	10	...
layer1	1	2	3	4	5	...

Table(3) illustrates a scaled window with the size doubled. Using this table, less capable peers will act as they are using table(2). For those powerful peers, after they finish the first section, as is in table(3), they can move to the second section. This is the simplest illustration of the idea of scalable window size. Ideally, the window size should be computed on recipient of heartbeat message. When a peer updates the network connection measurements, it can compute a proper window size based on those data. Last

but not least, the priority table should be recomputed if the window size changes.

In this version, we fill the second section of the window using the same strategy as used for the first section. The result is:

- QoE is improved a little.
- Performance degrades sharply.

5.9 Performance Optimization

After last experiment, one simulation needs about 4.5 hours to complete. This is really time consuming. In order to do more iterations, we try to optimize the simulator performance right away. At first, optimization described in this section only aims at the performance of simulator. However, it turns out the QoE is also improved. We'll talk about this phenomenon in conclusion section.

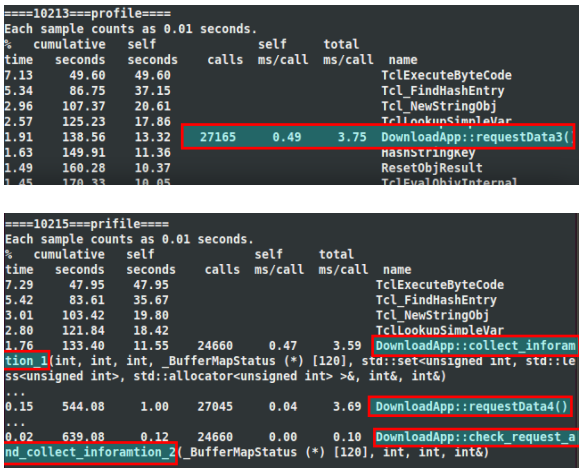


Figure 13: GNU Profile, Output

Fig(13) shows two screenshots of the output of 'gprof'. We first locate the most time consuming part that we can control. It turns out to be the 'requestData3()' function, which implements the strategy mentioned in last section. We then separate some sub functions from this one and find that the information collection is most time consuming. Note that we do not send probes to collect information. Instead, peer information is updated on recipient of periodical notification message. The information collection sub function only iterates over some map structures and fill the availability information into one unified 2D buffer map.

According to the analysis of the profile, we optimized those iteration process. For more details, please refer to the code repository of this project. Difference can be found between the following two commitments:

```
b45038404d8970c8ee1654d9bdbb10703432b8cd
5d3aabdb1546d88da6e7a026d2d1fe67b04ad2f1
```

Result of the optimization is:

- QoE: improved a little.
- Performance: 4.5h $\rightarrow$ 3.5h.

Note that this is not the end of the performance optimization. One of the possible directions is to optimize the sampling method using a reservoir[32]. However, this optimization is conducted with the knowledge that we'll only random select one peer if there are multiple peers available for one chunk. This kind of optimization is too aggressive and will limit the design space of strategies, so we don't do them in this research version. Designers can make notes and do this kind of optimization when all strategies are fixed in a real release version.

5.10 Introduce Randomness in Second Window Section

Due to time issue, our last trial based on the already established mechanism is to introduce randomness in second window section. As is known, randomness is a good property in the sense that it helps the system to spread chunks uniformly. The first window section is scheduled using a snake-shaped priority, this design considers user-perceived experience. However, the second window section is for those powerful peers. Those chunks are not hurry for playback. Thus we make the second section of the priority table randomly initialized. Every peer will have different priority table.

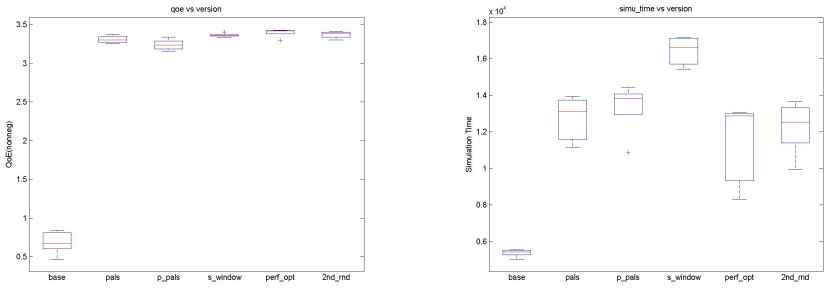
The result is:

- QoE: get worse.
- Performance: get worse.

5.11 Conclusion of the Case Study

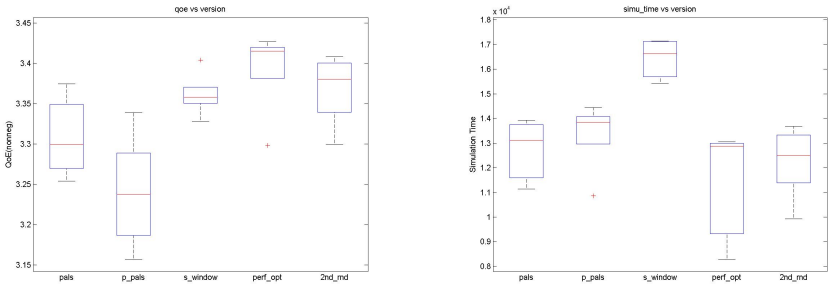
In this section we put together the result of 6 big versions. See Fig(14). In terms of QoE, version 2 improves a lot from baseline. After that, there are not so much difference in QoE. In terms of simulator performance, all modified versions are slower. The slowest version is the scalable window one. From the plot, we know our performance optimization really helps to reduce running time.

Fig(15) gives a closer look at the 5 modified versions. The most interesting observation is the difference before and after performance optimization. Note that in section(5.9), we did nothing about strategy. It is simply equivalent code reconstruction. However, while we observe better performace of



(a) qoe (b) time

Figure 14: Conclusion, QoE and Performance



(a) qoe (b) time

Figure 15: Conclusion, QoE and Performance

the simulator, we also find the QoE is improved a little. From the box plot, this improvement seems statistically significant(there are two ourliers).

Similar phenomena have been observed in our other VoD simulations using the same series ASTRI platform[16]. When the server is idle, the results tends to be better. Just like in our experiment, after performance optimization, the computation burden is lowered, thus we see better results. One conjecture is that this phenomenon is due to underlying timing facilities, either NS[48, 49] layer or the p2p simulator layer[16]. The verification of this point is left as future work.

6 Conclusion

Here we conclude two pieces of experience from this study:

- Engineering approach v.s. academic approach: where is the biggest cake? From section(5.11), we see the most significant improvement lies in the first engineering upgrade. Although the modification itself

reads simple, it makes the strategy framework more clear. Upon this framework, one simplest PALS like strategy achieves very good result. By examining the simulator carefully, we believe more engineering optimization can be found. Without an ideal underlying platform, the verification of algorithms on it is questionable. At same time, the benefit we tried exhaustively to get using better strategies may come easier with some engineering upgrades.

- Time distribution of this project:
  - 70%, literature survey.(30+ papers)
  - 15%, environment setup, bugfix of the platform.
  - 5%, first unified version(QoE:0.7→3.3, biggest improvement in this study)
  - 10%, scalable window, performance optimization, random 2nd section. (little outcome)

  Sometimes we try very hard, to get little outcome. Sometimes, ideas flash through our minds, and suddenly we get significant achievements.

7 Future Works

For multilayer P2P VoD system, we list some potential research topics:

- Degree of composition and decomposition.
- If we use the framework given in section(5.7), what's the optimal priority table. In order to achieve QoE aware design, models like [33] should be considered. Due to time limit, we haven't deducted the optimal table in this study.
- As for the simulator, how to explain the relationship between QoE and simulator performance. Theoretically speaking, QoE should only depend on strategies and irrelevant of simulator performance.
- Seeder side priority. The usual solution is to schedule chunks according to different requester side priority. However, this decision if fully decentralized. Some higher priority chunks seen by the requester may not be of very high priority compared with other requests seen by the seeder. In the case, differentiated service is essential. In our simulator, this priority information is not transferred and utilized. Architecture upgrade to support this feature is one promising direction.

(Not complete list. See my homepage for further amendments)

—-NEED REVIEW HERE—-

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Appendix

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