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QoS Provisioning for Adaptive Video Streaming over P2P Networks

Mubashar Mushtaq

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UNIVERSITÉ BORDEAUX 1
ÉCOLE DOCTORALE DE MATHÉMATIQUES ET
D'INFORMATIQUE

THÈSE

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Présentée et soutenue publiquement par

Mubashar MUSHTAQ

le : 12/12/2008

TITRE

**QoS Provisioning for Adaptive Video Streaming over P2P
Networks**

**Transport Adaptatif et Contrôle de la Qualité des Services
Vidéo sur les Réseaux Pair-à-Pair**

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Abstract

There is an increasing demand for scalable deployment of real-time multimedia streaming applications over Internet. In this context, Peer-to-Peer (P2P) networks are playing an important role for supporting robust and large-scale transmission of multimedia content to heterogeneous clients. However, the deployment of real-time video streaming applications over P2P networks arises lot of challenges due to heterogeneity of terminals and access networks, dynamicity of peers, and other problems inherited from IP network. Real-time streaming applications are very sensitive to packet loss, jitter / transmission delay, and available end-to-end bandwidth. These elements have key importance in QoS provisioning and need extra consideration for smooth delivery of video streaming applications over P2P networks. Beside the abovementioned issues, P2P applications lack of awareness in constructing their overlay topologies and do not have any explicit interaction with service and network providers. This situation leads to inefficient utilization of network resources and may cause potential violation of peering agreements between providers.

The aim of this thesis is to analyze these issues and to propose an adaptive real-time transport mechanism for QoS provisioning of Scalable Video Coding (SVC) applications over P2P networks. Our contributions in this dissertation are threefold. First, we propose a hybrid overlay organization mechanism allowing intelligent organization of sender peers based on network-awareness, media-awareness, and quality-awareness. This overlay organization is further used for an appropriate selection of best sender peers, and the efficient switching of peers to ensure a smooth video delivery when any of the sender peers is no more reliable. Second, we propose a packet video scheduling mechanism to assign different parts of the video content to specific peers. Third, we present a service provider driven P2P network framework that enables effective interaction between service / network providers and P2P applications to perform QoS provisioning mechanism for the video streaming.

Keywords: P2P, H.264/SVC, Real-time Video Streaming, Adaptive Quality of Service (QoS), SP-Driven P2P Networks, Admission Control

Résumé

Actuellement, nous constatons une augmentation de demande de services vidéo sur les réseaux P2P. Ces réseaux jouent un rôle primordial dans la transmission de contenus multimédia à grande échelle pour des clients hétérogènes. Cependant, le déploiement de services vidéo temps réel sur les réseaux P2P a suscité un grand nombre de défis dû à l'hétérogénéité des terminaux et des réseaux d'accès, aux caractéristiques dynamiques des pairs, et aux autres problèmes hérités des protocoles TCP/IP, à savoir les pertes de paquets, les délais de transfert et la variation de la bande passante de bout-en-bout.

Dans ce contexte, l'objectif de cette thèse est d'analyser les différents problèmes et de proposer un mécanisme de transport vidéo temps réel avec le provisionnement de la qualité de Service (QoS). Ainsi, nous proposons trois contributions majeures.

Premièrement, il s'agit d'un mécanisme de streaming vidéo adaptatif permettant de sélectionner les meilleurs pair émetteurs. Ce mécanisme permet de structurer les pairs dans des réseaux overlay hybrides avec une prise en charge des caractéristiques sémantiques du contenu et des caractéristiques physiques du lien de transport. Nous présentons ensuite un mécanisme d'ordonnancement de paquets vidéo combiné à une commutation de pairs et/ou de flux pour assurer un transport lisse. Finalement, une architecture de collaboration entre les applications P2P et les fournisseurs de services / réseaux est proposée pour supporter un contrôle d'admission de flux.

Mot clés: Pair-à-Pair (P2P), H.264/SVC, transport vidéo temps réel, Qualité de Service Adaptative, Contrôle d'admission, collaboration P2P/Fournisseur

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Publications Arising From This Thesis

Overlay Organization:

- **Mubashar Mushtaq** and Toufik Ahmed, “Hybrid Overlay Networks Management for Real-Time Multimedia Streaming over P2P Networks”, in **Lecture Notes in Computer Science (LNCS)**, “Management of Multimedia Networks and Services”, San José, California, USA, Vol. 4787, pp.1–13, 2007.

Quality Adaptation:

- Toufik Ahmed and **Mubashar Mushtaq** “P2P Object-based adaptive Multimedia Streaming (POEMS)” in **Journal of Network and Systems Management**, Special Issue on Peer-to-Peer Technologies in Network and Service Management, Vol. 15, Number 3, pp. 289-310(22), September 2007.

Service-Provider (SP) driven P2P Framework for IPTV Service Delivery:

- **Mubashar Mushtaq**, Toufik Ahmed, “IPTV Service Delivery over SP-driven P2P Networks - A Framework for Linear to Non-Linear Service Distribution” to appear in IEEE Consumer Communications & Networking Conference (**CCNC 09**), Las Vegas, 10 – 13, January 2009.
- **Mubashar Mushtaq** and Toufik Ahmed, “QoS Provisioning for Video Streaming Over SP-Driven P2P Networks Using Admission Control” to appear in IEEE Consumer Communications & Networking Conference (**CCNC 09**), Las Vegas, 10 – 13, January 2009.

Video Scheduling over P2P Networks:

- **Mubashar Mushtaq** and Toufik Ahmed, “Smooth Video Delivery for SVC based Media Streaming over P2P Networks” in proc. of IEEE Consumer Communications & Networking Conference (**CCNC 2008**) Special session on P2P, pp. 447-451, Jan 2008.

Adaptive Video Streaming over P2P Networks:

- **Mubashar Mushtaq** and Toufik Ahmed “Adaptive Packet Video Streaming Over P2P Networks Using Active Measurements” in proc. of the Eleventh IEEE Symposium on Computers and Communications (**ISCC'06**), IEEE Press, Cagliari, Italy. pp. 423 – 428, June 2006.
- Djamel-Eddine Meddour, **Mubasher Mushtaq**, and Toufik Ahmed, “Open Issues in P2P Multimedia Streaming”, in **IEEE ICC 2006**, workshop on Multimedia Communications Workshop (**MultiCom**), Istanbul, Turkey, 11 – 15 June 2006, URL: <http://www.ieee-icc.org/2006/index/conference-program-w.html>
- **Mubashar Mushtaq**, Toufik Ahmed, and Djamel-Eddine Meddour “Adaptive Packet Video Streaming Over P2P Networks” in 1st international conference on Scalable information systems (**InfoScale 2006**), ACM International Conference Proceeding part of International Workshop on Peer to Peer Information Management (**P2PIM'06**), Vol. 152, Article N°59, Hong Kong, May 2006.

Chapter 1

1 Introduction

1.1 Introduction

The rollout of Internet and the high popularity of the peer-to-peer (P2P) applications have revolutionized the communication society during this last decade. The P2P phenomenon has opened new horizons for future development of human society and in particular, digital video industry by offering mechanisms for collaboration among people in sharing their contents, documents, images, video, music, etc.

P2P networks are distributed Internet systems in which participants called peers cooperate with each others to achieve a desired service. In such networks, there is no centralized entity to control, organize, administer, or maintain the entire system. The promising features of P2P coupled with the widespread adoption of broadband residential access, availability of high bandwidth, portable and handy video capturing devices have attracted both academia and industry to propose new innovative solutions to enable content generation, sharing, and distribution among different social communities. The successful deployment of P2P technologies intend to overcome the shortcoming of client /server model by facilitating content sharing at large scale. In P2P systems, participating peers collaborate with each other in terms of their available network bandwidth, processing power, and shared memory. P2P systems build virtual overlay networks on the top of physical networks to facilitate peers organization for content localization and distribution.

We are now witnessing the emergence of a new class of popular P2P applications, namely, **P2P audio and video streaming**. While traditional P2P file distribution applications are targeted for elastic data transfers, P2P streaming focuses on the efficient delivery of audio and video content under tight timing requirements. Still in its infancy, both live and on-demand P2P streaming present many research challenges. The Internet Protocol (IP) is traditionally based on best effort packet switching technology and it does not guarantee any Quality of Service (QoS) such as transfer delay, jitter, loss, and bandwidth. Real-time video streaming applications require real-time performance guarantee in term of bounded transfer delay and jitter, low packet loss, and bandwidth guarantee. These performance metrics are affected by many parameters, some of them are related to the end systems (e.g. video server load) and others are related to the network (e.g. link capacity, router congestion, etc.). Moreover, P2P networks are constituted of different heterogeneous networks and devices, which may do not have symmetric characteristics to offer the same video quality to end clients. Therefore, a key goal of P2P video streaming design resides in the reliable delivery of high quality video over Internet while coping with unknown and dynamic issues of bandwidth, delay, jitter, packet loss, and peers dynamicity.

In parallel of P2P development, new video coding standards have emerged recently which are enabling the deployment of video content delivery over Internet in real-time with high QoS. There exist a number of coding standards that are used for the video encoding for different applications. MPEG-2 coding standard is considered suitable for major broadcasting application such as DVB-T and DVB-S and data storage in DVD. MPEG-4 AVC (advanced video coding) or H.264 is a coding standard that enables high quality video encoding with error resilience support for the media content suitable for broadcasting and mobile communication (MBMS “Multimedia Broadcast, Multimedia Service”, DVB-H, etc.). Scalable video coding (SVC) is the most promising encoding standard as an extension to H.264 standard is capable to produce video streams with variable bit-rates with different temporal, spatial, and quality scalability. SVC is considered suitable for delivering video content over heterogeneous networks such as Internet, P2P TV, and mobile P2P.

High popularity of P2P networks and the emergence of these coding standards have motivated us to propose an adaptive streaming mechanism over P2P networks. The problem statement and our key contributions in this dissertation are presented in following sub-sections.

1.2 Problem Statement

The QoS provisioning for real-time video streaming applications over P2P networks is considered the major problem. The heterogeneous networks, terminals, and characteristics of P2P networks make the situation more challenging. In this context, finding an optimal set of sender peers ensuring the QoS guarantee is important research topic. It is necessary that all sender peers collaborate in more cooperative and synchronized manner to ensure the better QoS. Thus, we need to address the problems related to (1) the selection of an appropriate set of sender peers to receive the video content with QoS guarantee, (2) dynamicity of the peers, when peers enter and/or leave the P2P system without any notification, and (3) the heterogeneity of peers because P2P networks are composed of different peers having distinct characteristics in terms of bandwidth, processing power, and memory. The overall streaming mechanism needs to ensure the video packets delivery between specific source and destination peers and based on the requirements such as low latency, high reliability, and priority.

P2P applications are today posing serious challenges to Internet infrastructures and there is continues battle between service providers and P2P applications for traffic management. On one hand, service providers do not have any explicit control over P2P traffic passing through their networks, so they are missing opportunities to gain revenues. On the other hand, P2P applications do not have complete knowledge of underlying network structure, consequently video content traverse thousands of miles to reach end-users that eventually degrade the QoS. In this regard, we study and present a framework for the cooperation between both these entities which can be helpful to honor the objectives of all the entities involved in video content delivery.

1.3 Objectives

In this dissertation, we aim to provide QoS provisioning for real-time video streaming over P2P networks. Our target system is P2P video on demand (VoD) streaming where a single peer

intends to receive video content from multiple sender peers. Our objective is to enhance the overall received video throughput with low packet drop ratio, and low packet transmission delay in order to enhance overall QoS. We propose mechanisms to better cooperation among P2P applications and service provider to enable a better utilization of available resources.

1.4 Contributions

In this dissertation, we aim to provide QoS provisioning for the real-time video streaming over P2P networks. Our target system is P2P video on demand (VoD) streaming. In this context, we leverage the characteristics of P2P networks and Scalable Video Coding (SVC) to propose an efficient and adaptive video streaming mechanism to guarantee an acceptable level of QoS. The proposed adaptive streaming mechanism consists of:

- Best peer selection mechanism: we propose a mechanism for best sender peers selection. The selection of an appropriate sub-set of sender peers among available peers is essential to guarantee smooth video delivery. The proposed best peer selection mechanism is based on active end-to-end probing to estimate the offered QoS among receiver peer and sender peers. In this mechanism, the sender peers are organized in different overlay networks on the basis of locality-awareness and semantic-awareness characteristics of the offered video quality. This overlay organization facilitates the selection of sender peers and peer switching.
- Peer/stream switching mechanism to guarantee QoS: P2P networks are exposed to dynamicity issue where participating peers may enter and/or leave the P2P system without any prior notification. Thus, the peers' dynamicity is not acceptable for stable QoS. We present a peer/stream switching mechanism to cope with the dynamicity issues and to guarantee a smooth video streaming. Peer switching is incorporated when the active sender peer is no more available or is unable to contribute to the streaming session. The peer switching mechanism is implemented by selecting the appropriate peer in replacement to the peer which leaves the network. Whereas, stream switching is incorporated when a sender peer fails to deliver the video content properly.
- Scalable video coding adaptation: Scalable video coding (H.264/SVC) is considered as the most promising video encoding technique that offers content adaptation characteristics (spatial, temporal, and quality adaptations) for the heterogeneous networks and terminals. We selected scalable video coding (SVC) for the video coding technique due to its promising characteristics for real-time content adaptation to propose our adaptive streaming mechanism.

- Packet video scheduling mechanism: We present a receiver side scheduler for the packet video scheduling, this scheduler assigns different video parts to sender peers on the basis of their relative schedule and playback deadlines. In this mechanism, different parts of videos (SVC video tiers) are assigned to the sender peers on the basis of their priority. The received video packets from multiple sender peers are collected in receiver-side buffers before their actual decoding for playback.
- Service provider (SP)-driven P2P network framework: this framework presents architecture for explicit interaction between service providers and P2P applications for guiding the applications to achieve more efficient network usage. It intends to address the issues of network efficiency and inefficient interaction between P2P and non-P2P applications. In this context, we propose an admission control mechanism for QoS guarantee.

1.5 Dissertation Overview

This dissertation is organized as follows:

Chapter 2 presents a state of the art of Peer-to-Peer (P2P) networks. This chapter starts with a classification of P2P networks. Then, we present classification for P2P video streaming applications and we identify their QoS requirements and challenges. Furthermore, we present video compression standards to understand the issues related to QoS-based end-to-end content delivery architecture.

Chapter 3 describes the components of our adaptive P2P streaming system which is receiver-centric targeting many-to-one services. The proposed mechanisms consist of best peer selection, peer/stream switching, and quality adaptation. The sender peers are organized in different overlay networks that facilitate the best peer selection. The overlay organization is subject to the locality-awareness and semantic-awareness based on offered video quality at each sender peer.

Chapter 4 presents video packet scheduling mechanism for the efficient collaboration among sender peers contributing in video streaming mechanism. This mechanism implements receiver-side buffer management for ensuring a smooth video quality.

Chapter 5 presents the SP-Driven P2P framework that proposes a mechanism for explicit communication between Service Providers and P2P applications. In this framework, we present an admission control mechanism for end-to-end content delivery over SP-Driven networks.

Chapter 6 concludes this dissertation and addresses some of our future work.

Chapter 2

2 Related Work

2.1 Peer-to-Peer Networks

During the last few years, Peer-to-Peer (P2P) architecture has gained tremendous attention from both industry and academia. This architecture has been widely adapted to propose efficient solutions for a number of applications to serve a large number of communities. The P2P phenomenon has really revolutionized the way how people consume and interact with multimedia content. Currently, P2P applications are utilizing around 70% of the global Internet bandwidth. Moreover, this exponential growth is amplified by the wide adoption of handy and portable media capturing devices and the availability of high speed Internet connections up to the last mile.

P2P networks are virtual networks that are built on top of the physical networks. They consist of heterogeneous interconnected peers that may have different characteristics and capabilities in terms of bandwidth, processing power, memory constraints, shared content, etc. P2P networks are highly dynamic in nature. A participant peer can join the networks to accomplish its own objectives and then leaves it without any prior notification.

Compared to the client-server systems, the P2P systems have no centralized entity to control, organize, administer, or maintain the entire system throughout its lifetime. The participating peers construct a virtual overlay network regardless of their geographical locations.

P2P systems have certain distinct characteristics and design-goals that make this architecture more popular to provider efficient solutions. The characteristics/goals of P2P systems include:

- **Symmetric communication:** The participating nodes (peers) are highly symmetric in P2P networks. These nodes are identical and they collaborate with each others for same objectives. There is no distinction of client and server. All the communication among peers is also symmetric and peers join P2P networks voluntarily.
- **Decentralized control:** Decentralization is an important characteristic of P2P networks. Pure P2P networks do not have any central entity that has global knowledge of the system to monitor and organize the network.
- **Self-organizing:** Peers organize themselves into a network through a discovery process. There is no global directory of peers or resources.
- **Robustness:** P2P networks provide a robust solution in which peers do not suffer from a single point of failure problem as in client server model. Every participant

shares its resources with others and the failure of several peers might have no or little impact on system functionality.

- **Scalability:** P2P networks are highly scalable because identical resources may be available at different peers. The high aggregation of peer's resources leads to high scalability with efficient load-balancing to a large number of peers.
- **Anonymity:** P2P architecture provides anonymity (privacy) to participating peers that is quite difficult in client-server architecture. In P2P systems, it is easy to hide the identification of any peer and the receiver does not know from where he gets the requested services.

P2P systems ensure the efficient utilization of resources with high reliability but on the other hand P2P systems exhibit certain problems. In fact, P2P networks are highly exposed to dynamicity and heterogeneity characteristics. In this regards, QoS guarantee for different multimedia applications becomes an important issue and needs significant. Furthermore, P2P networks are vulnerable to security, copyright infringement and privacy issues that need to be addressed for more efficient solutions. These latter challenges are not addressed in this thesis.

Currently, P2P networks are classified on the basis of certain characteristics that are illustrated in following sections.

2.2 Classification of P2P Networks

Today, Internet is largely based on the client/server model but we have seen that P2P networks are considered as most promising infrastructure for the resources sharing over the Internet. Both industry and academia are carrying their research to investigate further improvements for the P2P systems to make them more suitable for real-time multimedia sharing and to support further new applications. The main rational behind P2P framework is to provide a mechanism to share and utilize the available networking resources among the participating peers in distributed fashion. P2P networks are supposed to be fully decentralized and fully unstructured in the classical design but many existing P2P systems are designed with their own design rationales. These P2P systems can be classified with respect to their underlying architecture that how P2P system organizes the participating peers and how the participating peers collaborate with each other [1] [2]. Hence, a brief description for classification of P2P networks is depicted in Figure 2-1 and described in the following sub-sections.

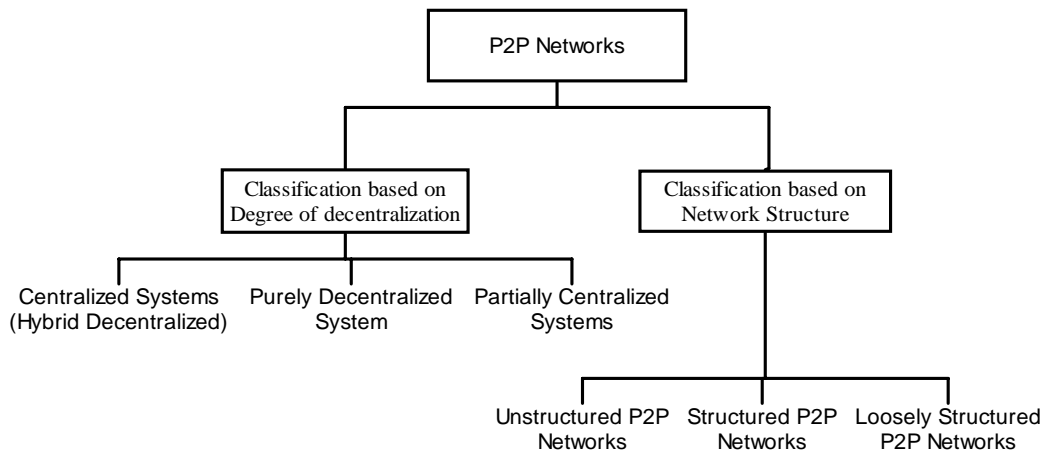


Figure 2-1: Classification of P2P Networks

2.2.1 Degree of Decentralization

As said before, the client-server model results into inefficient utilization of resources and makes the system more sensitive to failure. This creates problem of single point of failure with the increasing number of clients. P2P systems address the problems of the client/server model by providing efficient utilization of shared resources. P2P systems also overcome the problem of single point of failure, bottlenecks, and are highly efficient in terms of scalability. Degree of decentralization is an important characteristic of P2P systems that determines that how often peers rely on some designated servers. Although, the pure P2P architecture design proposes fully decentralized system where all the peers are organized in overlay networks in fully decentralized way but there exist some P2P systems that maintains one or more central servers for providing some functionalities essentially resources location. Thus, the major P2P systems are categorized in different categories on the basis of their degree of decentralization as shown in Figure 2-2 and are briefly described in following sub-sections:

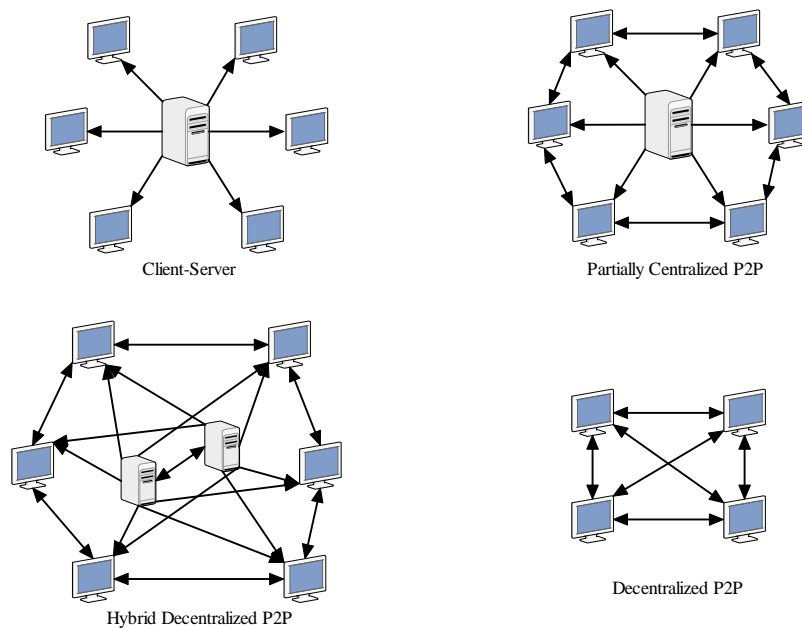


Figure 2-2: Degree of Decentralization

2.2.1.1 Centralized Systems

Centralized P2P systems comprise a designated central server that keeps information of all the participating peers in the P2P network. The central server facilitates the interaction between peers by maintaining directories of the shared files available with the respective peers. In literature, centralized P2P systems are also referred as hybrid decentralized systems [3]. Napster [4] and BitTorrent [5] are the most commonly known hybrid decentralized P2P system. In such systems, the central server keeps information of the shared files and acts as directory server for the peers forming P2P network. All new requests are sent to the central server that in return provides a list of peers having the requested content. Although the searching is done by the central server but the actual content delivery is done directly from the respective peer. The central server facilitates the interaction among the participating peers. These systems are simple to implement and their hierarchical organization enhances the system performance. However these systems are highly exposed to a single node of failure problem, non-scalable, vulnerable to copyright infringements, security, and other technical problem. They are also known as “Brokered Systems”, where new peers connect to the server to discover other peers present in the system but the actual communications are carried among the peers themselves.

Napster is the most famous popular music exchange application that was released in 1999. Although, this application is based on central P2P architecture but it works like a client-server model and it is an example of brokered systems. Napster maintains a central server acting as an index server that keep tracks of metadata of music files available at different locations. In the startup of this framework, when a new user initiates a query to search any file, it contacts the central server which returns a list of peers having the requested media content. After receiving this list, user creates direct connection with some peers and downloads the requested media file directly. US-courts have declared that providing such services contributes to third-party direct infringement of copyrighted materials. Napster was held liable to infringement of copyright by providing this indexing service [6].

Since Napster maintains a central repository for the indexing, so the searching mechanism is very quick and efficient. On the other hand, this centralized management incurs some disadvantages including a single point of failure, copyright issues, malicious attack, etc. Moreover, the organization of Napster does not allow providing a scalable solution for content distribution.

BitTorrent is most popular P2P protocol widely used for the content sharing. BitTorrent allows a large number of users to download quickly huge size content files. BitTorrent is not pure P2P protocol for content sharing but it works as hybrid decentralized P2P system [6][7]. BitTorrent protocol is comprised of different entities including a tracker, peers, and seeds. Each of these entities have distinct role in content sharing. The tracker acts as a central server and clients connect to this tracker for downloading the files. This tracker keeps track of the other peer and guides any peer on how it can reach other peers. Peers are participating nodes who are downloading and uploading different pieces of files. Seeds are those peers who have completed their download and now are participating in upload only. For the huge file, it is required that the seeds remain online until all the peers have successfully downloaded all the pieces of files successfully.

This protocol intends to address the issue of scalability that is most challenging issue for the P2P content distribution. This protocol ensures the fairness in P2P networks by splitting the huge

content into small pieces of same length. Any source peer offering high outbound bandwidth can download its desired file more quickly. The advantage of BitTorrent is that it provides highly efficient and scalable mechanism for the huge size file sharing. The content delivery mechanism suffers in this protocol, if seeds that are actually having the complete file go offline then no peer can get the entire file and the small pieces downloaded till now are of no use.

2.2.1.2 Purely Decentralized Systems

The purely decentralized P2P systems are often called pure P2P systems as well. In the pure decentralized systems there is no central entity to organize the participating peers and all the peers collaborate with each other for the same tasks. Gnutella [8] and Freenet [9] architecture are the purely decentralized P2P systems. In these systems, all the peers act as server and client at the same time. The peers in such systems are commonly known as *Servent* (**server + client**). All the peers facilitate the other peers for searching, routing, and content delivery process. The most promising characteristic of such systems is that they are highly robust due to decentralization. On the other hand, these systems use the flooding based searching mechanism that results into waste of bandwidth and sometimes the requested media content are not guaranteed because of the directory decentralization.

Gnutella is considered as the purely decentralized or real P2P system that evolved in response of Napster in 2000. Gnutella was designed to overcome the dependencies on the central server by implementing a fully distributed mechanism for the peer's discovery. In Gnutella framework peers are organized into virtual overlays without involving any central entity for coordination and all the peers communicate directly with each others. In this framework, communication among peers is carried by four types of messages at application level. These messages include PING, PONG, QUERY, and QUERY HITS. A new peer receives a list of some peers already present in network at start-up while joining Gnutella network. This new peer announces (broadcast) its presence by PING message in its vicinity and the other peers in network respond with the PONG message.

Any peer can QUERY to search some media content. The typical query method is flooding that is propagated (broadcasted) to all neighboring peers. The number of neighboring peers can be restricted by using the controlled TTL mechanism where the query is broadcasted in a certain radius. QUERY HITS is reply to the QUERY for the successful search and the requested file is downloaded from the respective peer.

In the absence of any central coordination entity, Gnutella overcomes the problem of single node of failure. The flooding based search mechanism is the disadvantage of Gnutella system that does not provide a scalable solution. To overcome this problem, a number of alternative methods are investigated [10]. The amount of query messages adds significant overhead that results into waste of bandwidth is another disadvantage of Gnutella architecture.

2.2.1.3 Partially Centralized Systems

Partially decentralized P2P systems present an improved form of the purely decentralized systems. The P2P systems based on FastTrack [11] architecture like KaZaA [12], Morpheus [13], iMesh [14] are partially decentralized systems. The main principle of such systems is exactly as of the purely decentralized systems. However in partially decentralized system some peers are

designated specific roles than the other peers. The specially designated peers are called “superpeer or supernode”. These special peers/nodes act as a central server for certain set of peers. In such system all the requests pass through these supernodes. These systems suffer from the single point of failure problem but in general such systems implements a mechanism to switch to another alternate supernode in case of failure. The intelligent and dynamic switching of supernodes restrains these systems from the single point of failure problem. The selection of an appropriate supernode for certain set of peers is still a challenging issue. The role assigned for the supernode is slightly different and depends on the overall architecture design of underlying P2P architecture. Generally, the incorporation of supernode facilitates the content delivery and improves the searching mechanism.

KaZaA and Morpheus are known as partially centralized and unstructured P2P systems. Both of these systems are quite similar to each other and organize the peer using supernodes. These systems are based on the FastTrack architecture and the supernodes in these systems are dynamically elected on the basis of processing power and available bandwidth. This supernode creates index to the media content available with different peers and they assist other peer in searching process. However the role of these supernodes is dynamically assigned based on the application type. Each supernode knows where the other super nodes are located and form a mesh structure among supernodes. In KaZaA architecture, a new peer joining the system at bootstrap connects to a known supernode and uploads metadata for the shared files. This metadata information is further used for search mechanism facilitated by these supernodes. Any client can download requested media content from other peers using simple HTTP protocol. In this context, KaZaA provides a scalable, efficient, and fault tolerance mechanism for file sharing.

The major advantage of these systems is that they reduce the peer’s discovery time in contrast to purely decentralized systems. These systems do not exhibits a single point of failure problem because there are multiple supernodes present in the system and any peer can become supernode at any instance of time.

The most recent versions of Gnutella 2 [15] have also adapted the notion of “supernodes”. In Gnutella2 the “supernodes” are refered as “superpeers” and it overcomes the problem of single point of failure.

In this section, we have classified the P2P systems with respect to their degree of decentralization. Decentralization is commonly considered an essential characteristic for the P2P system design, however highly decentralization may be good for the media delivery but centralization shows tremendous performance for the searching applications [16]. The intelligent design of Napster shows that decentralization is just a tool for P2P system not a goal [17].

2.2.2 P2P Network Structure

P2P networks are based on heterogeneous peers and possess dynamicity in terms that any peer forming the P2P network enters and/or leave the P2P network without any prior notification. In such scenario the overall structure of P2P network plays an important role for the organization of peers. In general, Peers are organized in overlay networks on the top of the physical networks. These overlay networks are virtual networks that may be totally irrelevant to the physical network.

Such P2P network structure provides a mechanism of indexing, searching, routing, and content delivery from some specific peer. The classification of P2P systems in different networking structures is presented in these sub-sections:

2.2.2.1 Unstructured P2P Networks

Unstructured P2P networks do not have any specific information about the requested media content of the peers. P2P systems like Gnutella, KaZaA, Morpeus, DirectConnect, BitTorrent, etc are unstructured P2P networks and they do not have any information for the organization of the peers with respect to the stored media content with their peers. In such systems, searching is performed on the random basis as no peer has the exact idea about the location of the peer having the requested content. A peer who is seeking for certain media content probes many different peers randomly and this search query is propagated until the requested content is found. This flooding based searching [18] results into duplicate queries and leads to the inefficient utilization of bandwidth. The searching mechanism takes a long time and sometimes the success is not guaranteed. These systems are generally known as first generation of P2P networks and the key advantage of such system is that they are easy to manage and they can accommodate a large number of peers. On the other hand, the major drawback of such system is that they do not guarantee a reliable content location discovery. Moreover, these systems are expensive in terms of their routing mechanism to find the requested content and hence are not considered scalable.

2.2.2.2 Structured P2P Networks

A number of P2P systems have been proposed to mitigate the disadvantages of the unstructured P2P networks. Generally, the structured P2P networks apply Distributed Hash Table (DHT) to support the scalability. These systems are known as second generation of P2P systems and include Chord [19], CAN [20], Pastry [21], Tapestry [22], etc. These structured networks address the problems of the scalability and provide efficient searching mechanism. The unstructured P2P networks are non-scalable because of their random searching mechanism. Whereas, the structured P2P networks organize the peers in overlay networks and the index for media content are placed at specified locations. The overlay organization provides a mapping for each file and the search queries with the help of mapping table. The strict overlay organization with controlled hash table based content locations offers a scalable solution for exact query match because the location of requested media content is primarily known.

The key advantage of the structured P2P networks is the high scalability and the success ratio for searching the media content with minimum time. The organization of peers is major disadvantage of these networks because the organization of peers is very complex and the mapping of routing table is extremely hard where a large number of peers enter and/or leaves the systems.

2.2.2.2.1 Chord

Chord [19] is a decentralized P2P system that provides routing and lookup infrastructure. Chord stores and maps key values for distributed items and peers present in the system. Data locations for certain content locating at different nodes can be implemented on top of Chord by associating a key with each data item and storing the key maps. The keys are assigned to data items and nodes using certain hash function and are organized in ordered form in a ring. Chord

implements a routing map and each node requires only knowing the key of its successor node in the circle. The search query for a certain key is passed along the circle via successor node until the required key is found.

In chord, when a new peer joins the system it is assigned its key and placed in the circle with its successors information. When any peer leaves the system, all keys are updated accordingly. Chord provides the exact location of any content located at any node and guarantee the successful search mechanism. However, performance degrades drastically if the information is not up-to-date due to peers joining and leaving the system. In the worst case, a query traverses all the nodes. To improve the efficiency, chord implements additional information that is called “finger table” to accelerate the locating mechanism [23]. This finger table based routing information provides lookup services for locating the desired content.

2.2.2.2.2 CAN

CAN (Content Addressable Network) [20] is another distributed and lookup service based on hash tables. Each node in the CAN networks stores a part of hash table called Zones and information for the adjacent small zone. The basic operations supported by CAN include: insertion, lookup, and deletion of key information from the hash table. CAN is purely decentralized and scalable system as any node maintains only small information. The provided routing and lookup service is fault tolerant as nodes can follow alternate routes in case of failures. CAN uses a virtual d-dimensional coordinate space to store the key values in hash table. Each peer keeps track of neighboring peers in each dimension. When a new peer joins the network, it randomly selects a point in d-dimensional space and it contacts with the peer who is responsible for this zone. Eventually, the contractor peer splits the zone in two parts and new peer becomes responsible for one part. Later, this peer contacts to all the neighboring peers for the routing table updates. Each node maintains routing table information for about " N " nodes. When a node leaves the system the associated hash table entries are transferred to other neighboring node. In normal conditions all nodes exchange information with each other periodically. This CAN protocol normally guarantees discovery of other nodes (media content) in $\Theta\left(\log N^{1/d}\right)$ steps where “ N ” and “ d ” represent the number of nodes and dimensions respectively.

2.2.2.2.3 Pastry

Pastry [21][24] is another overlay and routing network like Chord that implements a distributed hash table where key values are stored in a redundant peer-to-peer network. In this system, while joining this system, a nodeID is assigned randomly to the peer. Every peer maintains a routing table along neighboring peers set. It updates its routing table and informs other neighboring peers about its presence when a new peer joins the system. The failure of any peer is detected when any intermediate having its nodeID cannot communicate with this peer and is replaced with another peer. The expected routing requires $\Theta(\log n)$ steps when there are " n " number of active Pastry peers. Pastry node routes a message using the nodeID associated with the message to the closest node. The protocol is considered highly decentralized and is capable to provide self organization. Pastry protocol is capable to determine the best route for communication among peers.

2.2.2.3 Loosely/Semi-Structured P2P Networks

Loosely or Semi-structured P2P networks are neither fully structured no fully unstructured, however these systems are classified in between of these P2P networks. These systems exhibit some of the characteristics of both classes. In semi-structured P2P network, files locations are generally affected by routing hints. However, these routing hints are not completely specified. The query responses are not guaranteed in such systems because they do not provide complete routing map to access the respective media content, so the success of each query is not guaranteed. Media file are replicated at different places and they are served from the closest locations.

2.2.2.3.1 Freenet

Freenet [9] is known as the loosely structured and self-organizing P2P network. The overall design goal of Freenet is to provide file storage services rather than file sharing services. Freenet is highly decentralized system that provides anonymous and secure method for storing and retrieving the files. Freenet tends to address certain issues of P2P systems including security, client's anonymity, and issues of data replication for better performance. In Freenet, all the peers are assigned an ID and they know a certain number of other peers present in the network. All the shared content is also assigned an ID that is used for the searching and locating of requested content. The requested content is downloaded from the neighboring peer. The discovery of nearest peers is a repeated process that is based on the peer's and content's id. This protocol uses "chain-mode" content discovery mechanism instead of flooding based as used in the Gnutella systems. In this mechanism, search request is passed from node to node instead of broadcasting to a number of nodes. In this search mode, whenever any peer having the requested file receives the search query, the search process is stopped. In the other case, if any peer does not have the requested file, it forwards the request to some other peers. In Freenet, the actual content delivery is not made directly from the source peer to requester peer. The content delivery follows exactly the same path used while searching the content it follows the path of searching. In this way, the requester receives the requested content without knowing the actual sender of file. In this process the requested file is replicated on the intermediate peers. This mechanism is highly scalable and secure with anonymity of peers.

A brief classification of different P2P systems in terms of decentralization and their network structure is presented in Table 2-1:

	Unstructured P2P Networks	Semi-Structured P2P Networks	Structured P2P Networks
Partially Centralized	KaZaA, Morpeus, Groove, Gnutella , DirectConnect	-	-
Hybrid Decentralized	Napster, BitTorrent	-	-
Decentralized	Gnutella, eMULE, eDonkey2000, Overnet	Freenet	Chord, CAN, Pastry, Tapestry, ...

Table 2-1: Classification of P2P File Sharing Systems

2.3 Taxonomy of P2P Applications

Peer-to-Peer (P2P) has received a lot of attention for facilitating the content distribution across a large number of networks. This architecture has been adopted for the development of a large number of applications instead merely for content distribution. In this section, we have classified major P2P applications on the basis of their application areas:

2.3.1 P2P-based Content Distribution

The main goal for designing the P2P architecture was to support the content distribution across the large scale communities by lowering the load on the central servers. Mostly, the P2P systems fall into this category although we have further divided them in different groups on the basis of their distinct roles and modes of content sharing. The most popular P2P-based content distribution architecture/applications include: FastTrack (KaZaA, Morpheus), Napster, eDonkey/eMule, Akamai, BitTorrent. These systems use peers as distributed repository to distribute and share the requested content with other peers. They provide a controlled mechanism to access the content with the privileged roles of peers. The wide adaptation of such applications has replaced the dedicated content servers and provides more scalable solution to content distribution. These systems focus on dealing with the security problems to restrict the malicious peers accessing the content. Sometime, they are criticized due to the censorship and copyright infringement issues.

Real-time multimedia Streaming is another popular area of P2P-based content sharing. There has been comprehensive research available for proposing solutions for multimedia streaming over P2P networks. A number of architectures have been proposed to address different issues including peers management, robustness, dynamic adaptation, and QoS for the efficient streaming over P2P networks. The most popular architectures include: CoopNet, SpreadIt, ZIGZAG, PALS, and PROMISE. The details for these systems are presented in the section 2.7.2.

P2P-based IPTV service delivery is new emerging area in this category. These applications provide video-on-demand (VoD) service in addition to the real-time streaming. The most popular P2P-based IPTV frameworks include TVUPlayer, Joost, CoolStreaming, Cybersky-TV, TVants, PPLive, LiveStation, GridMedia, and iGridMedia.

2.3.2 P2P-based Communication Application

P2P architecture is widely adopted to offer different communication applications. These services provide the infrastructure for the direct and real-time communication and collaboration among peers. These applications include mainly chat and instant messaging such as: Skype, Internet messaging (AOL, AIM, ICQ, Yahoo, and MSN), NetNews, and Jabber.

2.3.3 P2P-based Distributed Computing & Processing

The idea behind distributed computing and processing applications is to take advantage of the P2P architecture to combine the available processing power (CPU cycles) of each peer. In such system, the original computational task is divided into sub-tasks and individual processing is

assigned to different peers and then the results are compiled. In such computations, a central control is necessary for better coordination and synchronization among the peers. P2P architecture has made it possible to use simple PCs for highly scientific computations for which super computer were required. P2P framework offers the power of supercomputers with very low cost. Seti@Home [25] and Genome@Home [26] are the commonly known applications for distributed processing. They generally use the peers when they are idle and sometimes work like client-server model. SETI @Home (Search for Extraterrestrial Intelligence) assigns tasks to idle computers using a screen-saver like program. In this system, a central server maintains information of all the peers. In such systems, each computational job takes around a week to complete the computation. However, this small program involves very low communications and the transmitted data is not of huge size. SETI@Home is widely used for many scientific computation projects including bioinformatics and weather forecasts, etc. Scalability is the main advantage of this system but this does not provide a general purpose platform for other applications.

2.3.4 P2P-based Collaboration

The core objective of the P2P architecture is to enable collaboration among peers to accomplish some common objective. Groove [27] is provided by Microsoft that is based on P2P architecture and provides an environment for collaborating. The main objective of Groove is to allow users to collaborate with each other without any central server involvement. Groove provides a synchronize mechanism that allows peers to collaborate in secure manner where messages are communicated in XML formats. This product intends to provide communication, content sharing and collaboration (group editing for a document, group drawing, co browsing, etc.) services to participating peers.

2.3.5 P2P-based Platform/Industry Infrastructure

P2P architecture provides infrastructures for the distributed applications. JXTA [28] is well known P2P-based infrastructure that provides a general purpose network programming and industry infrastructure. JXTA creates a P2P system with the basic functionalities and provides basic building blocks for the P2P applications. The main advantage of JXTA infrastructure is the distributed service discovery that is independent to any transport protocol. Scalability issue is major drawback for JXTA because global naming issue is still not resolved. In global naming issue, JXTA does not guarantee unique name for all the peers but for a certain group of peers only.

2.3.6 P2P-based Searching and Database Systems

P2P architecture is also used for certain searching and distributed database systems due to their structured networks organization. Next Page, Open Cola Folders, Pelbio, aKa InfraSearch, WebV2, and Sciencenet are P2P based search engines while Local Relational Model (LRM), PIER, and Piazza systems, AmbientDB, and Xpeer are categorized as relational/distributed database systems based on P2P architecture. These P2P databases overcome the limitations of relational and distributed DBMS that are based on static topologies and require heavy administration work. There is no global schema for the P2P database systems and each peer is a node with in a database.

2.3.7 Other Applications

Beside from the abovementioned application areas, P2P architecture is widely used for a number of other applications. The most popular areas include P2P Gaming (2AM, CenterSpan), medical research (especially for cancer research), educational domains, and many business areas. Writable web is another emerging area of P2P-based applications. It provides an interface to allow users to contribute for the web by adding, editing and sharing information. Blogger, Wiki Web, Manila, and Endeavors Technology, Inc. are the major P2P-based writable web applications.

A brief list of known P2P-based applications and protocols in different areas is described in Table 2-2.

Areas	Architectures & Applications
Content Distribution & File Sharing	FastTrack (KaZaA, Morpheus), Napster, eDonkey/eMule, Akamai, BitTorrent
Multimedia Streaming	Joost, Narada, PPStream, PROMISE, PALS, GNUStream
IPTV Services	TVUPlayer, Joost, CoolStreaming, Tribler, GridMedia, iGridMedia, Cybersky-TV, TVants, PPLive, LiveStation, Pando,
Distributed Computing & Processing	Seti@Home, Genome@Home, GridSystems, Avaki, Jivalti, Entropia, DataSynapse, Distributed.Net, DistributedScience, Popular Power, Ubero, 2AM
Communication	Skype, AOL, AIM, ICQ, NetNews
Collaboration	Groove, Center Span, eZ, Interbind, Engenia Software, Inc. Jabber
Industry Infrastructure	JXTA, Jabber
Distributed Search, Search Engines	Next Page, Open Cola Folders, Pelbio, aKa InfraSearch, WebV2, sciencenet, AmbientDB, and Xpeer
The Writable Web	Blogger, Wiki Web, Manila, Endeavors Technology, Inc.
Gaming	2AM, CenterSpan
...	...

Table 2-2: Examples of P2P-based Applications

2.4 Video Sharing over P2P Networks

Video sharing over P2P networks is one of the most popular domains over Internet. Video sharing is carried out by a set of hardware and software infrastructures. P2P-based video sharing

applications are most popular and are dominating the global Internet traffic. Video streams are shared among users over IP networks in P2P fashion where the entire participants (peers) collaborate with each other. P2P-based video sharing across the communities is not trivial and needs intensive measures for the efficient delivery with high Quality-of-service (QoS). In general, there are two different modes of video sharing over P2P networks as presented in Figure 2-3.

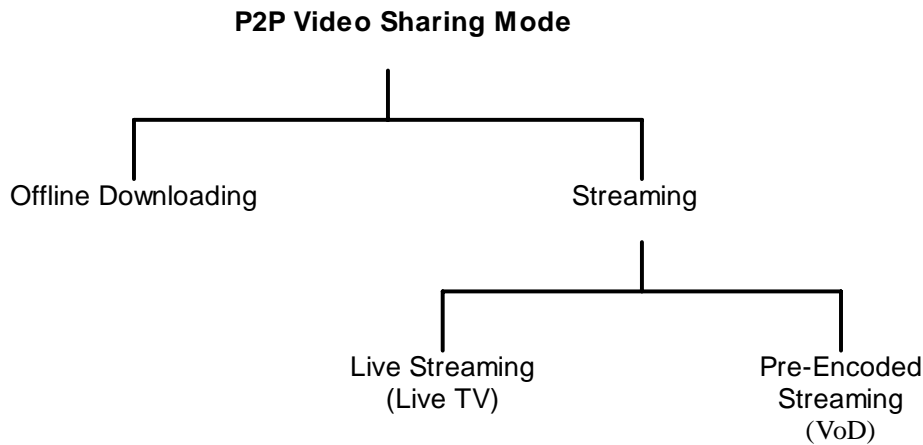


Figure 2-3: Video Sharing Modes over P2P Networks

The offline video downloading is the classic way of video sharing that is also called ‘open-after-downloading’ mode. The offline video downloading mode of video sharing allows delivering video content through a download process. In this mode of sharing the receiver peer download the requested video file from sender peer(s) using common protocols like FTP and HTTP. BitTorrent is most popular P2P application for the offline video sharing. There exist many other P2P systems that support such mode of video sharing among peers and communities, such as KaZaA [12], gnutella [8], iMesh [14], and eDonkey [30]. It takes a long time to download complete video file depending on the video size before actual playing. This technique reduces the flexibility and requires patience from the receiver (viewer) and viewer cannot check the video quality during the downloading.

Video Streaming is the second mode of video sharing over Internet. Live streaming (live TV) and pres-encoded streaming (VoD) are two form of streaming. Video on Demand (VoD) streaming enables the distribution of live and recorded audio and video streams across LANs, WANs, intranets and the Internet upon a client request. There exist a number of VoD applications such as Microsoft Netshow, RealVideo of Real Networks, StreamWorks of Xing Technology, etc. In P2P-based VoD streaming systems the client (content consumer) contacts the peers having the requested video and requests the appropriate stream. These video streams are further transmitted from different peers according to the overall system framework.

Live video streaming (Live TV) is most sensible way of media sharing that is also known as ‘play-while-downloading’ mode. Live streaming technique enables the simultaneous delivery of media content (audio/video) and the receive media content is playback at the same time. This technique overcomes the problems of video delivering via file download. It provides considerable additional features. The principle is as follows. The video is divided into small units which are transmitted over the network. The receiver collects the units, reconstructs the video and begins the

decoding. The playback begins without having to wait for the entire video to be delivered. In general, there are a short delay (5 – 15 seconds) between the start of the delivery and the beginning of the playback at the client. This delay, namely pre-roll delay, determines also the buffer size for storing the received units.

We have identified few of challenges that need to be tackled for efficient video sharing applications deployment.

1. Video services require high bandwidth between sender and receiver peer along the end-to-end content delivery path. A typical uncompressed video movie of 1 minute length, with a temporal and spatial resolution of 25 images per second and 320 x 240 pixels respectively requires 330 Mega Bytes and about 46 Mbps (Mega bit per second) bandwidth for real-time transmission. A compressed video content requires less bandwidth to transfer the video content as compared to uncompressed video content. For example, MPEG-2 TV quality (i.e. MP@ML) needs 4-8 Mbps bandwidth requirement whereas, uncompressed video content requires 200 Mbps. In general, the peers forming a P2P network have different uplink and downlink bandwidths capabilities. As a consequence, it is hard to receive the whole video content from a single sender peer. In this scenario, a single receiver peer intends to receive multimedia content from multiple sender peers. This selection of multiple sender peers leads to new problems that we expose later.
2. In general, video services have real-time traffic characteristics where video content needs to be playback continuously at the rate they are sampled. It is necessary that the whole media content are received before their actual playback deadline. If the data does not arrive at the expected playback deadline then the streaming process will stop. Buffering some parts of this data on the receiver end can reduce this problem for some applications but the latency remains a challenge. Furthermore, high latency is not acceptable for the real-time streaming applications. The latency problem has severe effect on the service provision and affects badly the Quality of Service (QoS) of received content.
3. The original video is encoded using some appropriate video encoding scheme before actual transmission over P2P networks. These encoded video streams are encoded at the sender peer and then decoded at receiver end before actual playing. The selection of the appropriate video encoding scheme plays an important role for the QoS enhancement as this encoding & decoding process adds significant overhead.
4. P2P networks are generally large scale networks that are composed of heterogeneous peers with distinct network, terminal and processing capabilities. In this scenario, the efficient video content delivery requires consideration of the heterogeneity problem to deliver the media content till the last mile.
5. P2P networks are highly dynamic in nature. Any peer node can enter and/or leave the P2P network without any prior notifications. In this scenario, we need to address the issue of peers' dynamicity that plays an important role to ensure the high QoS for the received video streams.

To summarize, Peer-to-Peer (P2P) streaming systems are generally known as self growing systems where all the participants have different role. There is no central media content server in the classical framework for the P2P media streaming system. The “many-to-one” mode of video streaming is most common instead of “one-to-one” [29] in P2P systems. In “many-to-one” mode a single receiver receives requested media content from multiple sender peers, whereas in “one-to-one” mode the requested video content is received directly from a single sender peer. These video services are classified on the basis of the applications transmission model that may be unicast, multicast, broadcast or anycast depending on the application type and targeted users.

The advantages of video streaming are considerable. First, it is not necessary to wait for a long delay before viewing the video. A low storage space is required to store the portions of video to view. These two benefits are determined by the duration of the pre-roll delay.

2.5 Video Streaming over P2P Networks

P2P networks are widely used for different applications as we described in the preceding section. The main focus of this dissertation is to enhance the overall QoS for the real-time streaming over P2P networks. The real-time multimedia content delivery (streaming) with QoS provisioning in P2P fashion is not a trivial task and it needs a careful considerations.

In this section, we classify some of the existing P2P-based multimedia streaming systems. P2P-based multimedia streaming systems can be categorized into two broader categories: in terms of peer organization in overlay networks, and in terms of system architecture. The systems are classified on the basis of their peers organizations and structures and a brief description is presented in Figure 2-4. The detailed discussion for these classifications is presented in the following section 2.6 and section 2.7.

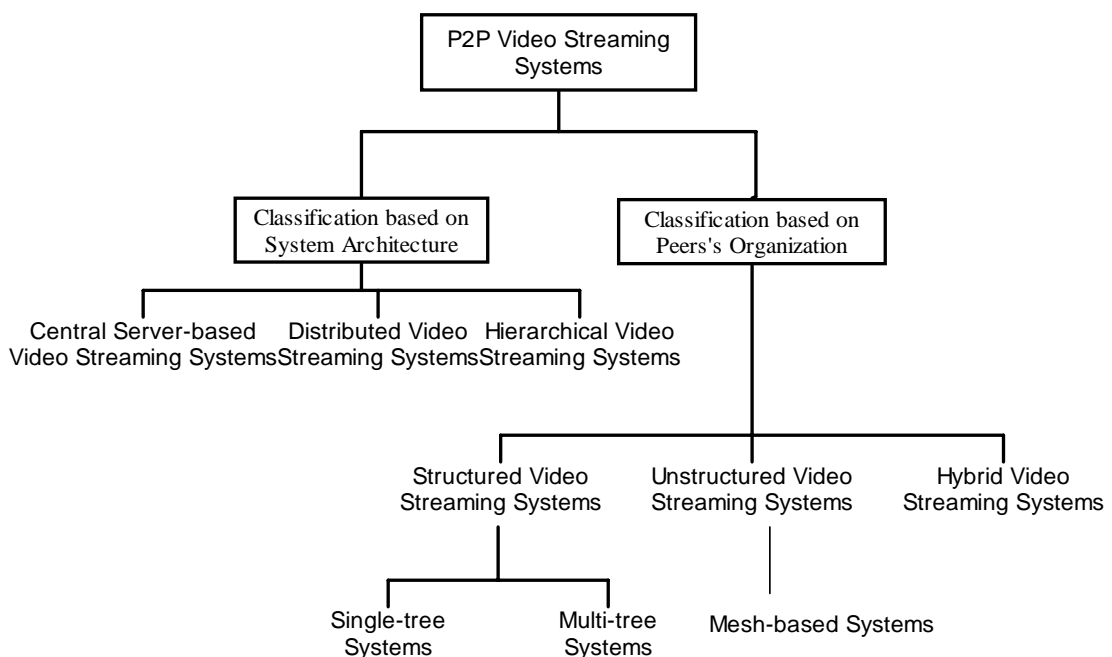


Figure 2-4: Classification of P2P Streaming Systems

2.6 Classification of Video Streaming Systems based on Peer Organization

2.6.1 Structured P2P Streaming Systems

2.6.1.1 Single-tree Streaming Model

Single-tree model is the first and most intuitive model for the streaming applications based on P2P architecture. In such model, peers are organized hierarchically forming a tree. The source peer is located at the root of tree. The media content are shared among peers with a continuous flow from root to leaf of the tree. The main goal of such system is to emulate an IP multicast tree at application level for media streaming. Any peer who enters in the system is organized at certain level of the tree and it receives the media content from its parent peer. The most common P2P Streaming Systems based on Single-tree model include: SpreadIt [31], PeerCast [32], NICE [33], ESM [34], Overcast [35], etc. These systems differ from each others for the creation of peers in single tree, maintenance of the tree, the arrival and departure of any peer from the system.

In single-tree model, generally the number of peers at each level increases exponentially. At the first level there are two peers receiving the media content directly from the source, while at the second level of tree there are four peers receiving media content from their parent peers, and so on as shown in Figure 2-5. The single-tree is not necessarily complete binary tree. At any certain level, there might be different number of leaf peers and again this peers organization in single tree is varying from system to system.

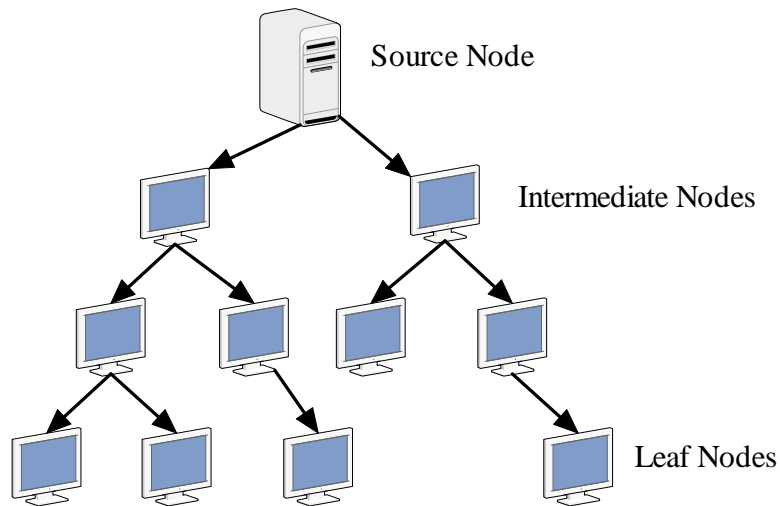


Figure 2-5: Single-tree based Media Streaming

The single-tree model is well suited for the streaming applications as it is simple to implement and has low overhead for organization of peers. Media streaming applications are highly stringent to the delay. In this scenario and for efficient content delivery with minimum delay rate, it is recommended to limit the level of parent peers in such tree-based model. Hence, this model can be used for an optimal delay by limiting the level of parent peer.

Besides from these advantages, single-tree model suffers from certain disadvantages. Peers dynamicity is challenging issue in the single-tree model. The situation becomes worst in the high

churn when a large number of peers enter and/or leave the system. In this scenario, a recovery mechanism is implemented to maintain the peers' organization in tree form. The available peers are attached with the best parenting peer depending on system characteristics. It is very hard to perform quickly the repairing operation that is not viable for the streaming application. This tree construction and maintenance mechanisms leads to extra overhead.

Peers heterogeneity is another issue in such organization. The peers having differences in their uplink and downlink bandwidth capacities require extra measures for efficient content delivery. Leaf peers are highly dependent to the interior peers, if any intermediate peer is unable to provide the required computational or bandwidth resources then the child peers suffer from the high delay and in some cases the content delivery is never assured. Furthermore, the leaf peers that are present at the lowest level of the tree do not have any child peer and thus they do not contribute for their uplink bandwidth. Generally, participating peers want to be only leaf nodes in single tree model due to their greedy behavior. However, a very large number of leaf nodes eventually degrade the peer bandwidth efficient utilization.

Single-tree model do not provide any control over peers, so this model is unable to penalize free-riding problem that is considered as a major threat to the P2P systems. In this issue, peers join some P2P network to download the desire media content or to receive other service without contributing their own resources with other. If only a few peers contribute their resource in P2P systems, these few peers effectively act as centralized servers that is infringement to the objectives of P2P. A study for the Gnutella architecture [134] shows that around 66% of peers do not share even a single byte and joins the networks just for downloading the video content from other peers.

2.6.1.2 Multi-tree Streaming Model

Single-tree based streaming systems are highly exposed to the problems including the load-balancing with leaf peers, fragility arising by the heterogeneity and dynamicity of peers. Multiple-tree model addresses the aforementioned issues by organizing the peers in multiple trees model to facilitate media streaming applications. In multiple-tree model, peers are organized in multiple trees. The original media content are also divided into multiple streams and distributed in multiple trees for their sharing. Every peer in such multiple trees is placed at different level. Every peer acts as an intermediate peer at-least in one tree so that it can contribute its uplink bandwidth capacity by facilitating other peers. The organization of peers leads to efficient bandwidth utilizing where every peer contribute with its capacity and reduce the overall stress over the network. The sudden departure of any sender peer during the media transmission do not effect much as each peer only contributes to a certain part of media content. SplitStream [36] and CoopNet [37][39] are most popular streaming applications based on multi-tree model. Figure 2-6 presents a brief description of multi-tree model with two trees.

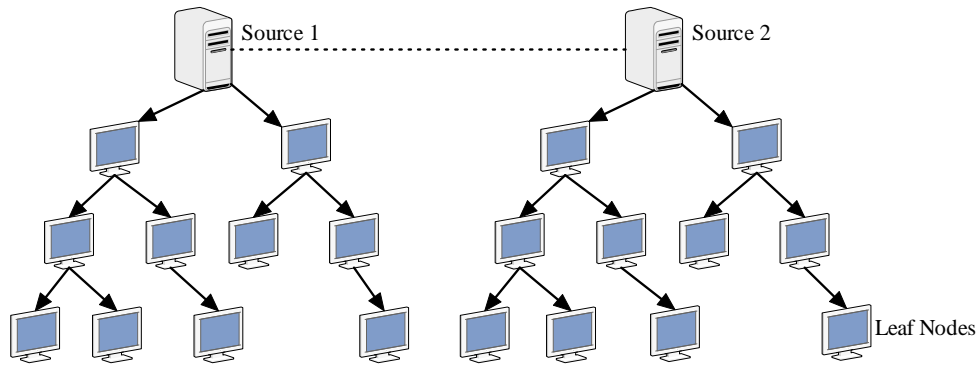


Figure 2-6: Multiple-tree based Media Streaming

In a fully balanced multi-tree model, each peer appears exactly once as an intermediate peer that contributes its uplink bandwidth with other peers. At each level of tree, each video stream is further divided into sub-streams and distributed among child peers. Multi-tree model overcomes the high delay problem of single tree model and provides highly resilience where every peer shares their resources with other. On the other hand, multi-tree model also suffers from some disadvantages. The complexity to organize peers in multiple trees is much higher than that of single-tree models.

CoopNet and SplitStream both are fully decentralized P2P streaming systems and they organize peers into multiple trees. They divide the original video stream into multiple stripes. All the individual stripes are delivered from sender to receiver peer using an independent single multicast tree. In both of these systems, any peer appears as an intermediate node in only one subtree. Therefore, the failure of any peer affects only a single stripe of video quality that eventually can be recovered from other source peers. P2PCast [40][41] is also a framework based on multi-tree streaming model. This system intends to provide streaming to a large number of users with efficient utilization of bandwidth. This framework divides the original media content into small strips as it is based on the multi-tree model where the trees are generated randomly. All the individual strips follow different trees for the streaming. In such system, any new peer joins the network by contacting any other peer already present in the network. This framework addresses the issue of free-riding where each peer contributes exactly the equal amount of uplink bandwidth that it consumes for downlink but leaf nodes do not contribute their uplink bandwidths with other peers.

2.6.2 Unstructured P2P Streaming Systems (Mesh-based Systems)

Besides from these structured and organized streaming systems, there exist a number of streaming systems that do not organize peers in any hierarchical structure for media transmission. In such system, original media content is divided into small parts called “*chunks*”. These systems do not implement any overlay topology. There is no specific route and all the individual video chunks follow different routes/paths from sender peers to receiver peer. These systems provide robustness for the streaming application. If any peer is no more present in the network, the specific chunk is requested from some other active peer while trying to minimize the control overhead. The most famous system based on this approach is CoolStreaming/DONET [42]. Mesh-based streaming systems are commonly known as unstructured P2P systems. In mesh-based systems, peers are connected randomly and there exist many different paths for the actual content delivery.

Tree-based streaming systems are based on either single-tree model or multi-tree model. However, they suffer from the single point of failure problem where any child peer receives media content directly from its single parent node. The departure of any parent peers intentionally or accidentally degrades the received content quality. Mesh-based streaming systems address the single-point of failure problem and provide a robust solution for streaming applications. There is no static topology construction for peers in mesh-based streaming systems. All peers are connected to many different neighboring peers dynamically. Any peer can download media content from multiple peers and can contribute its uplink bandwidth to share media content with other peers simultaneously depending on uplink and downlink capabilities. Many existing P2P systems such as, Chainsaw [43], PRIME [44], Narada [45] and DONet/CoolStreaming [42] implement mesh based peers organization to facilitate streaming applications.

In mesh-based systems original video is divided into many small parts called “chunks” and receiver peers can receive these parts from different source peers. In tree-based systems, video content is transmitted along the tree path. However, there is no specific path for content delivery in mesh-based systems. In mesh-based systems, any peer can receive media content from different source peers and if a source peer is unable to deliver certain parts of video content, it can be requested from other source peer. The actual media transmission is still transmitted by trees but there are multiple paths for content delivery and the virtual connections among peers are monitored constantly. Mesh networks implement a control plane to monitor the currently available active peers to provide a robust solution for streaming. The mesh network is constantly updated on the basis of this information.

Mesh-based systems are very simple to repair in case of peer’s departure. It provides an optimal content delivery with low delay. A low delay is achieved in mesh systems since all the systems implement best sender peer selection for content delivery. On the other hand, these systems are expensive to control the overall infrastructure since the network links and peer states are constantly monitored. These systems implements a pull based data delivery model for the content delivery that is presented in next section.

Narada [45] is an example of mesh-based streaming system that presents the approach of end-system multicast to support audio and video conferencing. Narada is mesh-based distributed P2P streaming system that focuses on multiple senders to multiple receivers streaming application. Narada runs a distance vector algorithm extended with path information on top of the mesh. It proposes a mechanism for the group membership management that facilitates the self organization of mesh in the case when any peer leaves the system. The mesh is maintained after periodically probing of nodes and does not rely on any central node. In Narada architecture, peers are managed in different groups where each peer keeps track of other peers in the group and information is periodically exchanged among the peers present in the same group and forming a mesh. While joining the Narada system, peer gets a list of existing peers in bootstrap process and it contacts those peers to be added as a neighbor peer. When a peer wants to leave the system, it notifies its neighbor peer that updates the information for neighboring peers. This protocol is considered robust due to the mesh based organization. However, Narada emphasizes on a small scaled P2P networks and is not considered suitable for large scaled networks because of intensive control overhead that is employed by intensive interactions between peers.

2.6.3 Pull and Push mode for content delivery

There are mainly two methods for the data exchange among peers for streaming applications. These methods are called push and pull [46]. Tree-based systems implement push method in which source peer pushes the media content to the entire child nodes. In mesh-based systems there is no specific criterion for the organization of peers. Any peer having the requested content blindly pushes it to the neighbor peers. It is possible that any peer receives same chunk of content from multiple peers that results into redundancy problem. To avoid such problem, pull method is implemented in mesh-based streaming systems. In this mechanism, a peer pulls the requested media content from the other peers. More recently a number of new push-pull approaches have been proposed to combine the ideas from both tree-based push approaches and mesh-based pull approaches [47]. All the active peers having the media content maintains different buffer maps representing the information for the available chunks with certain peers. These buffer maps are exchanged among peers. A peer seeking for certain media content determines a schedule to pull different chunks from multiple peers. The number of sender peers depends on the uplink and downlink bandwidth capacity.

2.6.4 Hybrid P2P Streaming Systems

There exist some other streaming applications based on P2P architecture that can neither be classified as structured (Tree-based or Mesh-based) P2P streaming systems nor unstructured streaming systems. These systems exhibit the combined characteristics of both structured and unstructured streaming systems. We can classify these systems as Hybrid Streaming Systems. Bullet [46][47] is one of the well known streaming systems of this category. Bullet system intends to provide mechanism for large file transfer and real-time streaming to a large number of receiver peers. Bullet combines the characteristics of both tree-based and mesh-based streaming systems for their data distribution and streaming. The main functionalities offered by Bullet include: an efficient tree management and maintenance mechanism, efficient membership management, and an efficient peer selection mechanism. Bullet divides the original video content into small chunks. The receiver peers receive these chunks from their parent nodes present in the tree and the missing chunks that are not available with parent nodes are downloaded from other peers. This framework offers a best peer selection mechanism where each node periodically evaluates its sender peers and in case if some new best peer is available it switches to the new sender peer.

2.7 Classification of P2P Streaming Systems based on Systems Architecture

P2P streaming systems can be classified on the basis of their system architecture which specifies how these systems construct the distribution tree to offer real time streaming services. These systems can be classified into three main categories: central-server based, distributed-based and hierarchical-based. We have presented a brief description for the characteristics of some of the P2P systems in Table 2-3. We present the main characteristics of our proposed streaming mechanism in this re-produced table that is originally presented in [37].

P2P System	Architecture	Video Coding	Packet Level-Scheduling
CoolStreaming	Overlay mesh	Possible use of layered coding or MDC	Yes
CoopNet	Multiple Trees	MDC	No
PALS	Receiver-driven tree	Layered coding	Receiver-driven
PROMISE	Receiver-driven tree	Possible use of FEC	Receiver-driven
SplitStream	Multiple trees	Possible use of layered coding or MDC	No
Bullet	Overlay mesh	Possible use of layered coding or MDC	Receiver-driven
Our proposed mechanism	Receiver-driven multiple trees	SVC possible use of layered coding or MDC	Receiver-driven -Yes

Table 2-3: P2P Streaming Systems Characteristics

2.7.1 Central Server based P2P Video Streaming Systems

Central server based P2P video streaming systems rely on a central server for the management of peers and for the creation of the content distribution tree. These systems exhibit the problem of single point of failure. CoopNet is most popular central-server based P2P streaming system.

CoopNet [38][39] is a framework for distributing streaming media content using cooperative networking. This framework is based on the client/server model. The main objective of “CoopNet” is to distribute media contents to a large number of hosts in a scalable way and to overcome the problems arising when the server is overwhelmed. In CoopNet framework, all clients cooperate with each other for the distribution of media content. The main objective of this framework is to support robust and resilient solution for the P2P-based media streaming. CoopNet falls into the category of structured P2P media streaming where a central server plays a major role for content distribution. The participating peers are managed into multiple trees as described in section 2.6.1.2. This central server that is also root of each tree provides a simple and scalable solution to control the arrival and the departure of the peers. Any new peer has to contact the central server which organizes the new peer into tree organization. Any peer while exiting from the network informs the central server which updates the tree structure accordingly. This server-based peer joining and leaving process is very fast but introduces significant overhead on the server. This overhead may cause serious problems in the high churn when a large number of peers suddenly join and/or leave the network. The optimization for the multiple trees is proposed to control the depth of trees with respect to the network conditions. On the other hand, such organization suffers from a single point of failure because the network coordinator and the content provider are hosted on the same server. In [39] “CoopNet” approach is applied for supporting resilient live streaming using application-layer multicast over an unreliable set of peers. The resilience is provided by introducing redundancy both in network paths and data. Furthermore, the server also acts as media server so its failure not only disturbs the tree management but also affects the quality of media quality.

In spite of the above problems, the use of multiple description coding (MDC) [49] for data encoding is a salient feature of “CoopNet” framework. In the case of sender peers failure, receiver peer does not lose the whole video but a proportion of quality is lost. Furthermore, the original video quality can be regenerated even in the case of multiple failures by combing different media streams.

2.7.2 Distributed P2P Video Streaming Systems

In contrast to the central server based streaming systems, distributed system does not have any central server for the peer’s management. All the peers management and tree creation mechanism are fully distributed.

2.7.2.1 SpreadIt

SpreadIt [31] is known as a distributed P2P streaming system. SpreadIt organizes peers in a single multicast distribution tree. This multicast tree is dynamically constructed for peer requesting live media content. Each node presents in the tree, forwards the media content to its descendants. It tends to support real-time content delivery and aims to reduce the data loss during the actual transmission. When a new peer joins the network, it traverses the tree downward to find an appropriate place with unsaturated bandwidth. While leaving the network, if this peer has any child nodes, it sends them a message to connect to its parent node for graceful departure without affecting the system performance otherwise, it leaves the system silently without notifying other peers. In the case of sudden failure of any intermediate peer, the recovery mechanism is difficult and requires periodical probing to detect and adjust the situation.

This mechanism focuses on the dynamic tree construction and does not deal with the bandwidth heterogeneity issues. In contrast to the CoopNet, SpreadIt builds only a single tree where each node relies on parent node for content delivery. In this scenario, this mechanism is more vulnerable in case of peer departure and suffers from the bottleneck at the source.

2.7.2.2 PALS

“PALS” [50][51] is a receiver centric framework for adaptive streaming from multiple senders to a single receiver. In this framework, a receiver peer coordinates the delivery of a layer encoded stream from multiple senders. PALS focuses on using layered video encoding for the robustness. In this framework, a receiver peer periodically requests media content from multiple senders. The requested contents are delivered by each sender in a specific order. “PALS” specifies a peer selection mechanism in order to identify a subset of sender peers that contribute for maximizing the overall throughput. In this framework, initial peers are selected on a random basis because there is no information available in the start of the streaming mechanism about the best sender peers. Peer selection is an iterative process. Each time a new peer is admitted and if it enhances the overall throughput it is kept as sender peer otherwise it is dropped. This framework also implements a sliding window process to keep all the senders synchronized. It uses a diagonal buffer distribution to receive all video blocks from different senders for quality adaptation (QA). The receiver manages its buffer regularly on the basis of packets consumption and sends the buffer state to each sender.

The QA mechanism for “PALS” determines inter-layer bandwidth allocation for a period of time rather than on a per-packet basis.

This framework shows significant performance, however it suffers from few drawbacks as well. The random selection of peers in start of the “PALS” framework may affect the overall QoS in the beginning of the streaming session. Furthermore, it focuses to use layered video coding where all the enhancement layers are decodable with respect to its base layer and is considered as more suitable for heterogeneous networks. “PALS” does not consider the priorities of the video layers. However, P2P network possess highly dynamic characteristics, if a peer contributing for the base layer crashes during content delivery the already received enhancement layers are of no use.

2.7.2.3 PROMISE

In the same context as PALS, “PROMISE” [52] describes a P2P media streaming mechanism using CollectCast where a single receiver peer collects media streams from multiple senders. The proposed idea is based on the collaborating of multiple sender peers for media streaming. This framework uses Pastry [21] as underlying P2P network that returns multiple peers for each lookup request. Authors presented topology-aware best peers selection mechanism after a comparative analysis of mainly “Topology Aware Selection” and “End-to-End Selection” techniques. In the Topology Aware Selection technique all the shared communication paths/links are considered for the best selection of the sending peers, while in the end-to-end selection technique, these shared segments/links are not considered during the selection of the sender peers. To ensure constant streaming, status of all the sender peers are monitored regularly. The monitoring is performed by collecting statistics on the loss rate and streaming rate contributed from each peer. Dynamic switching is performed in the case of peer failure and network fluctuations. In the case of peer failure, another best peer is selected while in the case of network fluctuations new rate is assigned for the active peers or by adding or removing the sender peers from the active set of peers.

2.7.2.4 SplitStream

In contrast to the “CoopNet”, “SplitStream” [53] is fully-distributed P2P system for the live multicast in cooperative environments. The system is based on the tree-based peer organization and it supports application level multicast on the top of Pastry [21] as underlying P2P architecture that provides a structured routing protocol. In SplitStream, the original content are divided into multiple stripes and each stripe is transferred from independent senders in independent tree. This framework organizes peers in multiple trees in such a way that each peer is an intermediate node in only one tree. This organization ensures that each peer do not upload more data than it downloads to provide fairness. Furthermore, this organization leads to robust content sharing, the quality of media content is not affected too much in case of any peer failure because each peer contributes for a small stripe only. In this way, this system disseminates the data across the forest of multicast trees. In the case of sudden peer’s crashes, this framework does not implement any packet loss recovery mechanism. The main objective of such distribution is to balance the load across the trees under the available bandwidth constraints. The forest based multicast does not overload any node beyond its bandwidth.

2.7.3 Hierarchical P2P Video Streaming Systems

The hierarchical based streaming systems organize peer into multiple hierarchical layers/clusters that eventually enhances the system scalability. “ZigZag” [54][55] is known as hierarchical P2P video streaming system, and is based on a single source streaming system. It proposes a method to organize the receiver peers into a cluster hierarchy for their management and organization. The organization of peers is performed in multiple hierarchical layers. The lowest layer consists of all the peers present while the upper layers map the peers into clusters forming the head of the clusters. ZigZag uses a multicast tree construction and maintenance approach based on a hierarchy of bounded-size clusters that helps in reducing the number of processing hops in order to avoid the network bottleneck. An optimizing technique is presented to improve the overall Quality of Service by balancing the load on each peer node.

2.8 P2P-based Video-on-Demand and IPTV Systems

In the previous section, we discussed some of the existing P2P systems offering real-time media streaming service. There are plenty of other systems available offering such services with some additive services like video on demand and live IPTV services. In this section, we describe some of the popular P2P frameworks widely adopted for live IPTV service delivery.

2.8.1 CoolStreaming

CoolStreaming is [56] a P2PTV (peer-to-peer television) mechanism that enables users to share television content with each other over the Internet. It works similar to BitTorrent where participating peers (viewers) upload content at the same time as the programs are downloaded and viewed. This framework is comprised of three layers: network layer, streaming layer, and display layer. “CoolStreaming” is a framework for live media streaming based on data-driven overlay networks where each node periodically exchanges data availability information with each other. The management of nodes that join the network is controlled by an origin node that is persistent during the lifetime of the streaming session. In this mechanism, the nodes in the data-driven overlay network are organized in a Breath-First-Search (BFS) tree, where origin node is located at level ‘0’ while any node at level ‘k’ can be reached in ‘k’ hops from the origin. CoolStreaming is considered as more resilient mesh-based system. The participating nodes in the data-driven overlay construct a mesh. This framework does not maintain any explicit overlay structure but data is forwarded to the requesting peer using the mesh connections. This framework implements a scalable membership and partnership management algorithm with an intelligent scheduling that copes well with the bandwidth differences of uploading clients and thus minimizes skipping during playback.

“CoolStreaming” is known as the first P2PTV system that attracts more than 1 million viewers [57]. Its service was stopped in June 2005, due to copyright issues, but again launched IPTV services in October 2006.

2.8.2 PPLive

“PPLive” is a mesh based P2P live media streaming system offering the most popular P2P-based IPTV services [58]. This framework is a mesh-based P2P live streaming system based on the buffer maps and it implements a gossip based protocol for the management of the peers and for the channels discovery. It also implements a P2P based video distribution protocol. When a new peer joins the system, it sends a query to central channel server and gets the list of available channels. The media streaming maintains buffer maps that provide the information for the availability of the requested video chunk with the respective peers. This mechanism enables the QoS measurement across the P2P networks and it provides the channels with data rate ranging from 250kbps to 450kbps, while few channels are available with 800kbps.

2.8.3 GridMedia & iGridMedia

“GridMedia” is another well known system offering P2P-based IPTV services. It has attracted more than 500,000 users all over the world with more than 15,000 concurrent users [59]. “GridMedia” organizes the peers in unstructured overlay networks and implements a push-pull based approach to fetch the media contents from the neighbor peers [60]. It implements a block scheduling mechanism for efficient contents sharing among the peers. The neighbor peers are selected on the random basis. The push-pull based mechanism offers the benefits of the receiver-centric approach and also reduces the overall latency. This system offers a successful streaming session at 300kbps.

iGridMedia [61] is an extension of the “GridMedia” protocol that is open source solution. This protocol focuses on providing delay-guaranteed services to support real-time applications. iGridMedia provides perfect scalability and it is considered suitable even for very large scale user numbers requesting for same channel. This protocol provides video streaming with 5 seconds guaranteed delay even when the peers churn rate is high.

2.8.4 P2CAST

“P2CAST” [62] is architecture for VoD services that uses P2P approach. It proposes patching techniques and relies on unicast connections only among the peers. The key idea behind this architecture is that each client behaves as server for the other client and provides some patches of data to other clients. This approach is used to overcome the overload on a single media server. The main outcome of this approach is to scale better than traditional client/server architecture by constructing the application overlay and to provide continuous playback.

2.9 Video Coding Standards for Video Applications

Peer-to-Peer networks are expanding quickly as a network of heterogeneous communication networks. The number of users is growing exponentially and they are using services that need high bandwidth requirements such as media streaming. We have stated earlier that these heterogeneous clients have variant uplink and downlink bandwidth capabilities. Furthermore, high popularity of video sharing across the networks arise the issue the networks congestion. There are a number of

problems that affect video streaming. Current Internet infrastructure (IP-Internet Protocol) provides best effort services that do not offer quality of service. These problems result into (1) lower throughput (bandwidth management), (2) packet losses, (3) high transfer delay, and (4) delay variation (jitter). These parameters are unpredictable and never acceptable for real-time applications. Thus, we need to design solutions for efficient video streaming over P2P networks that can address the abovementioned issues.

To cope with the heterogeneity issue, the bandwidth management is crucial for the video streaming applications over P2P networks. The bandwidth among the peers (sender and receiver) is highly dynamic and not known a priori. Furthermore, a single peer is unable to serve the whole video content in real-time to receiver peer due to the difference between uplink and downlink capabilities. In this scenario, if a sender transmits more video packets than the available bandwidth, video packets may be lost during the transmission. If the sender transmits lower video packets than the available bandwidth than the receiver playbacks a poor video quality. One of techniques to deal with bandwidth variation is to use adaptive video streaming. In adaptive video streaming, end-to-end bandwidth among the sender and receiver peer is estimated and then video rate is adapted accordingly. The selection of an appropriate codec is vital and plays an important role to deal with the problem of bandwidth management. The best codec can produce video at certain rate and when the channel conditions change it applies an adaptive behavior. Some of the video codec designed for video streaming are presented in Section 2.9.1.

2.9.1 Video Compression Standards

There exist a number of audio/video coding standards for video content delivery over IP networks. Currently there are different families of video compression standards, performed under the auspices of ITU-T (International Telecommunication Union-Telecommunication), ISO (International Organization of Standardization) and IEC (International Electro-Technical Commission) as explained in Figure 2-7. A summary of the known compression standards is presented in Table 2-4.

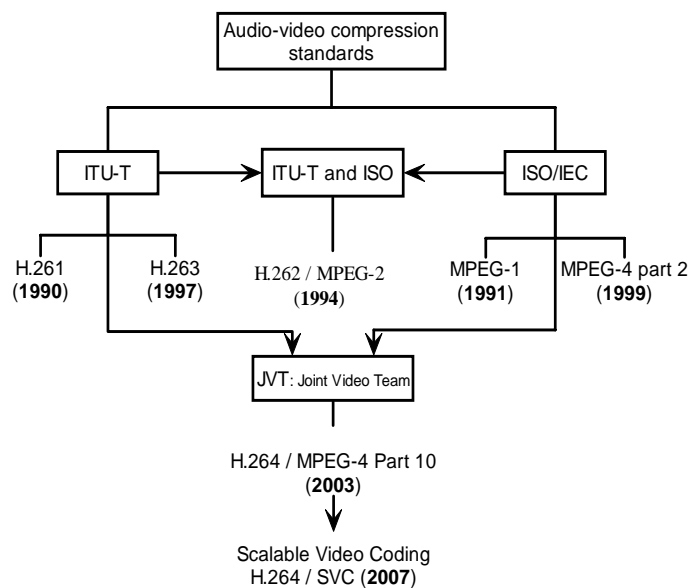


Figure 2-7: Video Compression Standards

Video Coding Standard	Targeted Applications	BitRate	Typical Resolution (Pixels)
H.261	Video Telephony & Teleconferencing over ISDN (low delay)	P * 64 Kb/s 384 Kb/s - 2Mb/s	176x144 or 352x288
H.263	Video Conferencing	28.8-768 Kb/s	352x288
MPEG-1	Video on Digital Storage (CD-ROM)	400 Kb/s – 1.5 Mb/s	352x240 or 352x288
MPEG-2	Broadcast Digital TV, DVD	1.5-15 Mb/s	720x480
MPEG-4 Part 2	Fixed and Mobile Web	Variable	176x144 or 352x288
H.264/ MPEG-4 part 10 (AVC)	Convergence of All video Applications, digital cinema	30 Kb/s – 600Mb/s	352x288 to 1920x1080
H.264/ MPEG-4 part 12 (SVC)	Convergence of All video Applications, digital cinema	30 Kb/s – 600Mb/s	352x288 to 1920x1080

Table 2-4: Summary of Compression Standards

2.9.1.1 H.261

H.261 is video codec for audiovisual services at p * 64 kbit/s (know also as p64 algorithm) [63]. This codec intends to provide a constant bit rate for videophone, videoconference and other audiovisual services over ISDN. The H.261 specification is implemented in several telecommunications devices and is integrated onto custom chips by several manufacturers. H.261 differs from MPEG-1 algorithm in terms of bit-rate and is designed for packet video streaming.

2.9.1.2 H.263 and H.263+

H.263 specifies a coded representation that can be used for compressing moving picture at low bit rates [64]. The basic configuration of ITU-T H.263 is based on ITU-T H.261. However H.263 differs from H.261 recommendation in the flowing. Half pixel precision is used for the motion compensation compared to full pixel precision. In addition to the core H.263 coding algorithm, four negotiable coding options are included to improve performance. All these options can be used together or separately. These options are: Unrestricted Motion Vector mode, Syntax-based Arithmetic Coding mode, and Advanced Prediction mode and PB-frames mode. H.263 supports five resolutions. In addition to CIF (352 x 288) and QCIF (176 x 144) that were supported by H.261, there is sub-QCIF (128 x 96), 4CIF (704 x 576), and 16CIF (1408 x 1152). H.263+ was

formally known as H.263v2 [65]. This codec enhances H.263 codec by adding new types of options which are scalability pictures, improved PB frames, custom source formats, and nine new coding modes.

2.9.1.3 MPEG-1, MPEG-2(H.262) and MPEG-4

Moving Picture Experts Group (MPEG) was created to develop video packing standards. At the moment, the available MPEG-standards include: MPEG-1 (ISO-11172) [66], MPEG-2 (ISO-13818) [67] and MPEG-4 (ISO-14496) [68]. The ISO/ IEC IEC13818 namely MPEG-2 is known also as ITU-T recommendation H.262, This recommendation is designed for high definition video quality [67]. MPEG-1, 2 and 4 are currently the wide well known codec in the word. Thanks to MPEG standards that Digital Television is now possible.

The first standard developed by MPEG committee is MPEG-1 codec. This codec targets a bit storage rate of 0.9 - 1.5 Mbps offering VHS quality at CIF resolution and 30 frames per second. MPEG-1 is not adapted for packet switched networks to the dependencies present in the P (Predicted) and B (bi-directional predicted) frames. MPEG-1 is highly vulnerable to packet losses and used for digital video storage on CD-ROM.

MPEG-2 extends MPEG-1 by including support for higher resolution video and others capabilities. It is designed for Digital Video Broadcasting (DVB) and high quality digital video storage (DVD). The target bit rate for MPEG-2 is 4 -5 Mbps. MPEG-2 supports scalability and it proposes three types: Spatial scalability allows the decoder to treat a subset of streams produced by the coder to rebuild and display video with a reduced space resolution. Temporal scalability allows the decoder to treat a subset of streams produced by the coder to rebuild and display a video with reduced temporal resolution. With SNR Scalability (Signal to Noise Ratio), the coder transmits the difference between the original image and the preceding that allows to get final image with best quality. However, MPEG-2 is not designed for transmission over video over IP due to high bandwidth consumption.

MPEG-4 standard [70][71][72] is an emerging digital multimedia standard with associated protocols for representing, manipulating and transporting natural and synthetic multimedia content (i.e. audio, video and data) over a broad range of communication infrastructures including IP and ATM .

The original characteristic of MPEG-4 (part 2) is to provide an integrated object-oriented representation of multimedia content for the support of new ways of communication, access, and interaction with digital audiovisual data, and offering a common technical solution to various telecommunications, broadcast, and interactive services. MPEG-4 addressed a broad range of existing and emerging multimedia applications such as multimedia broadcasting, content-based audiovisual database access, games, audiovisual home editing, advanced audiovisual communications and video over mobile networks. The MPEG-4 (part 2) standard introduces a new technique of coding multimedia scenes called “object-based video coding”. This technique allows the encoding of different audio-visual objects in the scene independently. An MPEG-4 scene consists of one or more Audio Visual Object (AVO), each of them is characterized by temporal and spatial information.

H.264 MPEG4 – part 10 is new coding standard for the scalable video coding. A brief description for H.264-AVC and H.264 (SVC) is presented in following section.

2.9.1.4 H.264

Since 1997, the IUT-T video experts Group (VCEG) has been working on new coding standard, namely H.26L. In late 2001, MPEG video group and VCEG decided to work together as Joint Video Team (JVT) to create a single technical design for forthcoming ITU-T Recommendation and for new part of ISO/IEC MPEG-4 standard. The final working version carries the denomination H.264 and ISO/IEC 14496-10 (MPEG-4 part 10). It has been adopted in July 2003 as a standard in Berlin Meeting.

The codec specification [73] itself distinguishes conceptually between a video coding layer (VCL), and a network abstraction layer (NAL). The VCL contains the signal processing functionality of the codec, things such as transform, quantization, motion search/compensation, and the loop filter. It follows the general concept of most of today's video codecs, a macroblock-based coder that utilizes inter picture prediction with motion compensation, and transform coding of the residual signal.

The basic configuration of the H.264 codec is similar to H.263 and MPEG-4 (Part 2). The image width and height of the source data are restricted to be multiples of 16. Pictures are divided into macroblocks of 16x16 pixels. A number of consecutive macroblocks in coding order can be organized in slices. Slices represent independent coding units in a way that they can be decoded without referencing other slices of the same frame. The outputs of the VCL are slices. The NAL encapsulates the slices into Network Abstraction Layer Units (NALUs) which are suitable for the transmission over packet networks or the use in packet oriented multiplex environments [74].

Current test demonstrates that H.264 standard can achieve 50% coding gain over MPEG-2, 47% coding gain over H.263 baseline, and 24% coding gain over H.263 high profile encoders within the motion compensation loop in order to reduce visual artifacts and improve prediction [75].

One of the main properties of the H.264 codec is the complete decoupling of the transmission time, the decoding time, and the sampling or presentation time of slices and pictures. The codec itself is unaware of time, and does not carry information such as the number of skipped frames (as common in the form of the Temporal Reference in earlier video compression standards). Also, there are NAL units that are affecting many pictures and are, hence, inherently time-less. IETF has defined RTP payload format for H.264 codec, which defines essentially timing information of NAL units [75].

The H.264/AVC (advanced video coding) also known as MPEG 4–part 10 standard was developed by the ITU-T Video Coding Experts Group (VCEG) together with the ISO/IEC Moving Picture Experts Group (MPEG) [73]. This standard enables higher quality video coding by supporting increased sample bit depth precision and higher-resolution color information, including sampling structures. Furthermore, it also provides valuable error-resilient support for delivery the media content.

H.264 SVC (Scalable video coding) is new scalable extensions for H.264 standard that is considered most promising video format for media streaming over heterogeneous networks [103][104]. A scalable video coding is capable to produce highly compressed bit-streams, in order to create a wide variety of bit-rates. SVC is briefly described in section 2.9.3.

2.9.1.5 Video Management Standards

There are certain other well known video management standards that provide the metadata information for the video. The Moving Expert Group (MPEG) is currently defining designated video management standards for multimedia application that include: MPEG-7 [77], and MPEG-21 [78][79].

2.9.1.5.1 MPEG-7

MPEG-7 [77], intends to provide information about the content (meta-data). MPEG-7 framework emphasizes the standardization of multimedia content description. The main objective for such standardization is to incorporate fast searching mechanism by supporting filtering and content identification methods. It describes the low-level structure of content and provides information on how objects are combined in a scene. MPEG-7 uses the following tools [78] for these objectives: Descriptor, Description Schemes, Description Definition Language (DDL), and System tools.

2.9.1.5.2 MPEG-21

MPEG-21 [79] describes an open framework that allows integration of all components of a delivery chain necessary to generate, use, manipulate, manage, deliver, and adapt multimedia content across a wide range of networks and devices. MPEG-21 elaborates the elements by defining the syntax and semantics of their characteristics, such as interfaces to the elements. It also addresses the necessary framework functionality, such as the protocols associated with the interfaces, and mechanisms to provide a repository, composition, conformance, etc.

MPEG-21 is based on two essential concepts: (1) defining the fundamental unit of distribution and transaction, that is Digital Item, and the concept of users interacting with these digital items. This framework is based on certain key elements that includes: Digital Item Declaration, Digital Item Identification and Description, Content Handling and Usage, Intellectual Property Management and Protection, transparent access to content across terminals and networks, and Content Representation that how media resources are represented.

2.9.1.6 Proprietary Format

There exist a number of proprietary solutions for video compression such as Windows Media Format by Microsoft [80] and Real Video by Real Network [81].

Windows Media 9 Series product was developed and approved as standard in March 2006. It includes many technologies and tools for media creation, delivery, security and playback. However Windows Media is proprietary, OS-based system and details information of this format is not available for public domain. A little work considers streaming using Windows Media Format.

The last release of Real Network has developed recently Real Video 11 and released in 2008. A compression format designed for broadband and high definition television. It promises a good quality with a reasonable bandwidth and suitable to use as streaming media format. Furthermore, real video is a multiplatform solution.

2.9.2 Scalable Video Coding Model

Video Scalability Coding model is also used for the bandwidth management for packet video applications. There are two main techniques used for video coding. Block Discrete Cosing Transform (DCT) and Discrete Wavelet Transform (DWT). DWT has received much attention for its inherent multi-resolution and progressive characteristics. Many works [82][83][84][85] have explored these techniques for adaptive video coding. DCT-based coding also proposes video scalability techniques in many codec such as scalability provided in MPEG-2 and H.263+. In video scalability, the video is coded in multiple representations. The major applications of scalability include Internet video, wireless video, multi-quality video services, video database browsing, etc.

Although a simple solution to scalable video is the simulcast technique that is based on transmission /storage of multiple independently coded reproductions of video. A more efficient alternative is scalable video coding, in which the bandwidth allocated to a given reproduction of video can be partially reutilized in coding of the next reproduction of video. Figure 2-8 depicts the different kinds of video scalability.

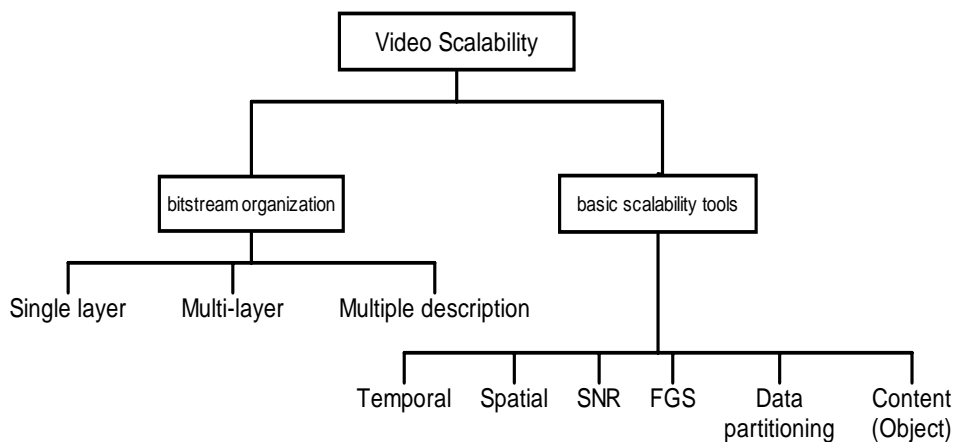


Figure 2-8: Video scalability coding modes

Grouping scalability according to the organization of the bit-stream leads to single layer scalability, multi-layer scalability, and multiple description scalability. In the single layer scalability, the data is coded in a single segmented stream. In this case, a simple truncation can be performed to get lower quality layers which leads to a fine granularity. In multi-layer scalability the data is coded in a base layer (BL) stream and one or more enhancement layers (EL) streams. The base layer is a separately decodable bit-stream. The enhancement layers can be decoded in conjunction with the base layer to increase perceived quality by increasing the picture rate, increasing the picture quality, or increasing the picture size. A third type of scalability is multiple description scalability, in which the data is coded in several (at least two) different streams. Multiple description coding provides two important properties (1) each description can be independently decoded to give a usable reproduction of the original signal, and (2) the multiple descriptions contain complementary

information so that the quality of the decoded signal improves with the number of descriptions that are correctly received. Unfortunately this scalability type is in principle less efficient, because dependencies between the streams cannot be exploited. A number of video coding algorithms have recently been proposed and they offered much functionality among which error resilient [86] [87], [88], [89] and [90].

Grouping scalability for video coding according to basic scalability tools lead to temporal scalability, spatial scalability, SNR scalability, fine grain scalability and object scalability.

Temporal scalability involves partitioning of the video into layers, where the lower layer is coded by itself to provide the basic temporal rate and the enhancement layer is coded with temporal prediction with respect to the lower layer. These layers, when decoded and temporally multiplexed, yield full temporal resolution. There is also support for temporal scalability by the use of B pictures. B pictures allow enhancement layer information to be used to increase perceived quality by increasing the picture rate of the displayed enhanced video sequence. This mode can be useful for heterogeneous networks with varying bandwidth capacity and also in conjunction with error correction schemes.

Spatial scalability refers to enhancement information to increase the picture quality by increasing picture resolution either horizontally, vertically, or both. Spatial scalability involves generating two spatial resolutions video layers from a single video source such that the lower layer is coded by itself to provide the basic spatial resolution and the enhancement layer employs the spatially interpolated lower layer and carries the full spatial resolution of the input video source.

SNR scalability refers to enhancement information to increase the picture quality without increasing picture resolution. SNR scalability and spatial scalability are equivalent except for the use of interpolation. Because compression introduces artifacts and distortions, the difference between a reconstructed picture and its original in the encoder is (nearly always) a nonzero-valued picture, containing what can be called the coding error. Normally, this coding error is lost at the encoder and never recovered. With SNR scalability, these coding error pictures can also be encoded and sent to the decoder, producing an enhancement to the decoded picture. The extra data serves to increase the signal-to-noise ratio of the video picture, and hence the term SNR scalability.

The Fine Grain Scalability (FGS), often known as the Streaming Profile is intended to support applications and environment where the bandwidth and/or computational power cannot be predicted and may vary dynamically. It was developed specially in MPEG-4 in response to the growing need on a video coding standard for streaming video over the Internet [91]. Three proposals were submitted to MPEG-4 for achieving FGS, namely, bit-plan coding of the predicted DCT residue [92][93], wavelet coding of image residue [94][95][96], and matching-pursuit coding of the predicted DCT residue [97][98]. The bit-plan coding of the predicted DCT residue was finally accepted after several experiment as a standard for FGS [99].

FGS and its combination with temporal scalability address a variety of challenging problems in delivering video over the Internet. FGS provides a mechanism that permits a single encoding process, producing a bit stream that can be modified subsequently in two different ways. Prior to transmission, the bit stream may be processed to scale it down to known bandwidth limitations. This can be performed dynamically, for example in response to the requirements of a statistical

multiplexing system. Downstream distribution points may reduce the bit rate if necessary. After the transmission, the receiver terminal can adapt the received stream to its capabilities.

An additional advantage of video scalability is its ability to provide resilience to transmission errors as the more important data of the lower layer can be sent over a channel with better error performance, whereas the less critical enhancement layer data can be sent over a channel with poor error performance.

A detailed description for the video coding standards and scalability video coding model can be found in [100].

2.9.3 Video Coding Schemes for Video Streaming over P2P Networks

Layered encoding (LC) and multiple descriptions coding (MDC) are considered suitable for many applications [101] over heterogeneous and P2P networks. In both these schemes original media content/video stream is truncated into further sub-streams called layers or description. Each layer/description can contribute to one or more characteristics of multimedia contents in terms of spatial, temporal resolution and quality (SNR level). LC and MDC transmit a suitable number of layers/descriptions to cope with the available bandwidth on the network paths.

The difference between LC and MDC lies in terms of dependency among bit-streams. In both schemes different media sub-streams are created from the original media stream. In LC these streams are called “Base Layer” and “Enhancement Layers” while for MDC these streams are known as descriptions. “Base layer” plays a vital role in LC and all the enhancement layers are generated with reference to the base layer. If the base layer is lost, then the enhancement layers are of no use because they are only decodable with reference to its base layer. The base layer provides the basic quality of the media while enhancement layer adds the quality to the base layer. In contrast, in MDC all the descriptions are decodable independently and provide a proportion of the quality. More and more descriptions are used to enhance the overall quality of the media contents. In MDC, if some packets of any description are lost, they can be reconstructed using the other descriptions while LC is sensitive to transmission losses [102]. The error resilience feature of MDC makes it more suitable for the applications operating over heterogeneous networks but extra overhead is added.

The scalable extension of H.264, known as Scalable Video Coding (SVC) is currently considered most promising video format for media streaming over heterogeneous networks [102][103][104]. In SVC encoding scheme, each video stream is encoded in multiple video quality tiers. Each quality tier can be decodable with different characteristics. First tier which provides the basic quality of the video is called “Base Tier” while other tiers, which are used to enhance the overall video quality of the base tier, are called “Enhancement Tiers” [105] [106].

An original SVC stream can be truncated to produce video of different qualities, sizes, and frame rates, i.e. in SNR, spatial and temporal dimensions. This scalability makes SVC bit streams suitable for heterogeneous networks and terminals to meet the QoS requirements for the streaming applications. In SVC stream, the base tier is encoded using a fully standard compatible H264 AVC (advanced video coding), 7.5 frames per second in QCIF resolution. Then enhancement tiers can be added, each providing temporal, spatial, or SNR scalability. SVC has the ability to provide the

decoder different layers with enhancement layers depending on reception of the base layer and lower enhancement layers. Thus a transmission scheme should ensure that these layers are transmitted such that packet loss is kept as low as possible even for high overall packet loss rates. Besides the ability to adapt to different heterogeneous networks and terminals, we can also achieve graceful degradation of video quality in case of packet loss and high end-to-end delay.

A layered stream representation of SVC in terms of spatial, temporal, and SNR resolution is shown in Figure 2-9. In SVC encoding scheme, each video stream is encoded in multiple video quality tiers. Each quality tier is decodable with different characteristics. The first tier (T0, S0, B0) providing the basic quality of the video is called “Base Tier” while other tiers which are used to enhance the overall video quality of the base tier are called “Enhancement Tiers”. It is important to note that the lower tiers (i.e. Base tier) are more important than the higher tiers (Enhancement tier) to decode a particular stream. Thus, for our study, we leverage the characteristics of SVC and P2P networks, to choose different sender peers according to their offered QoS to contribute to different video part (Tiers).

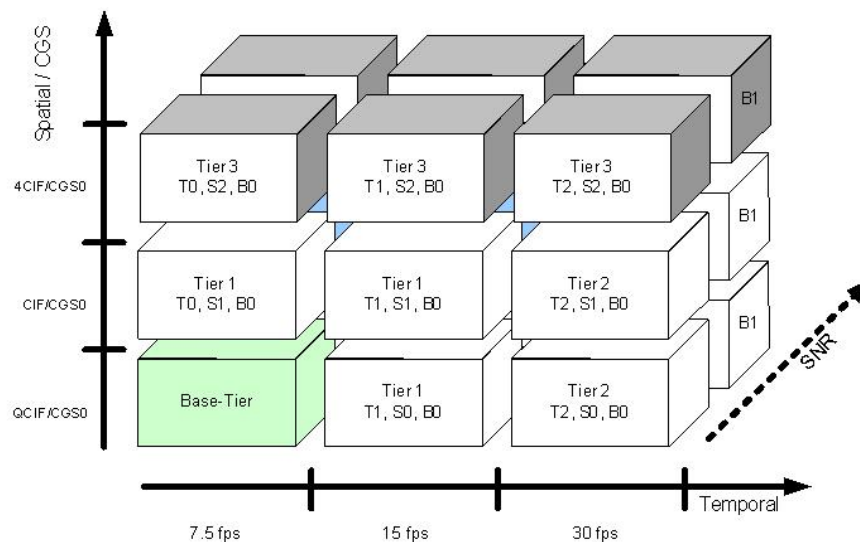


Figure 2-9: Scalable Video Encoding (Layered Stream Representation)

2.10 Conclusion

In this chapter, we have presented a brief state of art for the classification of P2P networks along with some of the popular applications based on this architecture. There are no standards available for the P2P classification until now. This classification represents the key characteristics of these P2P systems. The single-tree model is not suitable for video streaming due to its single point of failure problem, whereas mesh based peers organization suffer from data redundancy problem.

We select scalable video coding (SVC) for the video coding technique due to its promising characteristics and real-time content adaptation support for the heterogeneous networks and terminals. Since, SVC scheme organizes video streams in different tiers, so we adapt multiple trees approach for the organization of sender peer. The peers are organized into multiples trees (overlay networks) on the basis of their respective priority to offer video quality. This organization enables

the best peer selection mechanism where each sender peer is selected for the specific video quality tier from each overlay network.

The proposed adaptive on-demand video streaming mechanism that is composed of best peer selection, peer/streaming switching using the multiple trees approach, and quality adaptation is described in following chapter.

Chapter 3

3 Adaptive Video Streaming over P2P Networks

3.1 Introduction

During the last few years, we have witnessed a rapid growth in the support and the deployment of real time applications and services over the Internet. We live a new phenomenon that has changed the way how users consume and interact with digital content over Internet. In fact, content sharing between communities has revolutionized the Internet thanks to Peer-to-Peer (P2P) networks. P2P file sharing, emerging in 1999 has become increasingly popular to deploy live video streaming and video-on-demand (VoD) services. P2P phenomenon has gained tremendous attention during these years by facilitating the information flow from and back to the end users and contributing for around 70% of global Internet traffic [106]. This exponential growth is amplified by the wide adoption of handy and portable media capturing devices and the availability of high speed Internet connections up to the last mile.

P2P applications are realized by cooperation of peers by sharing their available resources with each others. For the video streaming applications, video contents can be distributed in many ways: (1) “one-to-one” model, (2) “one-to-many” model, and (3) “many-to-one” model. The “one-to-one” is the simplest model where a single peer share its video content with exactly one receive peer at a time. In “one-to-many” model, a single peer shares its video content with many peers at a time. In this scenario, the sender peer acts as a central content server. However, in practice it is very hard for one peer to facilitate many different peers at a time while offering high QoS.

“Many-to-one” model also called “multi-source streaming model” is considered more practical for P2P networks, because a single peer is unable to deliver the whole video content. Internet is currently based on the best effort service model which does not guarantee any QoS. In this scenario, the real-time services delivery over P2P networks becomes more challenging than traditional applications such as web and email. Traditional networks infrastructures are not capable to support the huge demands of real-time traffic in its existing characteristics. As an example, most of the peers (clients) are connected to Internet and the P2P networks using ADSL (Asymmetric Digital Subscriber Line) or cable modem connections. This kind of connection offers three to eight times more bandwidth for downlink (network-to-user) than uplink (user-to-network). The difference of uplink and downlink capacities makes it impossible to retrieve in real-time the whole media content from a single sender peer when it is the only contributor to the stream [108]. In this context, it becomes necessary to have multiple sender peers that contribute to the streaming of the

media content. In many-to-one streaming, a receiver peer reaches the required bandwidth by combining the uplink bandwidth of all respective sender peers. Thus, sender peers selection becomes an important criterion that should be taken into consideration to optimize the performance of the streaming mechanism. Similarly, assignment of the video parts to each peer according to their capability allows the provision of a better QoS and the effective utilization of the available bandwidth.

Figure 3-1 presents many-to-one architecture where a single receiver peer receives video content from multiple sender peers. In this architecture, several peers are connected logically and form the P2P network. Here, we have shown only peers having the requested video content. There should be some others peers that are connected in the P2P network but not sharing or not having the requested media content (these peers are not represented in Figure 3-1). All the peers having the content are **potential** peers for delivering the content. A subset of them is **candidate** for sending the content in the next period of time, among them a subset of peer are **active** and are effectively sending the content.

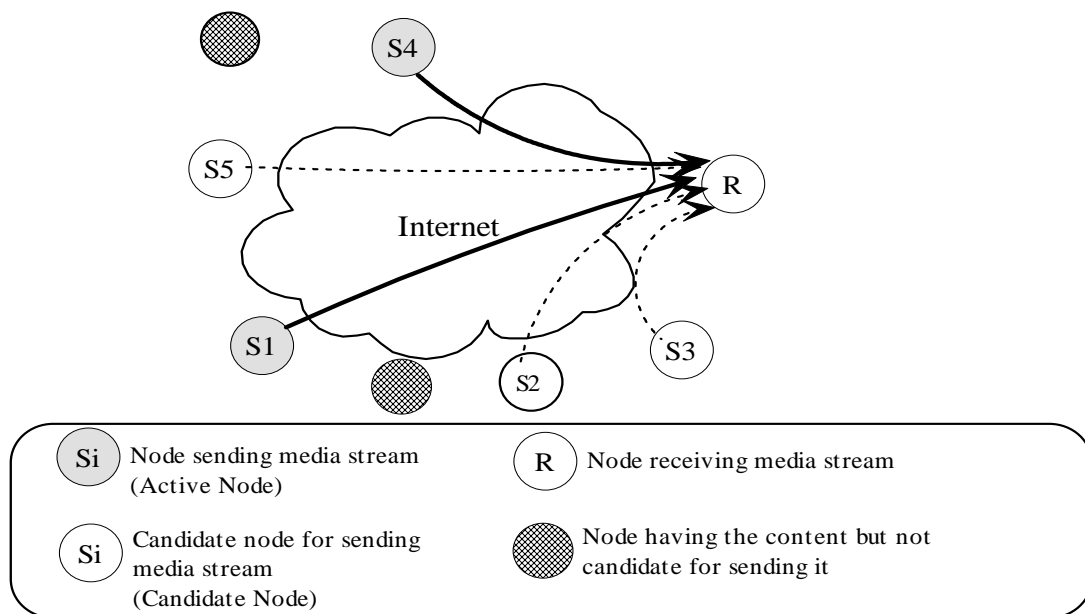


Figure 3-1: Many-to-one (multi-source) P2P Video Streaming

3.2 Design Principles

The design of an adaptive multimedia streaming system based on P2P architecture requires careful investigations to offer the streaming services with improved Quality-of-Service (QoS). We identify few design challenges before presenting our proposed adaptive mechanism for on demand video streaming over P2P networks. The design principles include the selection of appropriate sender peers for actual video content delivery, the selection of an effective video encoding scheme that is well suited for the heterogeneous networks, and video packet distribution among participating peers. These design principles are briefly described in the following sub-sections.

3.2.1 Peering Strategy

P2P networks are composed of a large number of heterogeneous networks and client. In P2P-based video streaming systems, a receiver peer intends to receive requested video from its neighboring peers. In this scenario, an efficient peering strategy is essential to enhance the overall QoS for received video. Peering strategy determines the selection of an appropriate set of sender peers among the peers having the requested contents. In video streaming systems, many clients may request the same sender peer for the video content that might not be capable to serve the entire requests. Thus, peering strategy facilitates the selection of sender peers in accordance to the uplink and downlink capabilities of the respective sender peers.

Such peering strategy is implemented by gathering the information of available video content and current status of any peer. In P2P networks, all the peers exchange their information with their neighbor peers periodically either using some central entity or in distributed way depending on the underlying architecture. In general, the best peer selection is based on the peers' locality. A receiver peer selects sender peers located very close to receive the video very quickly. However, there are many different techniques available to determine end-to-end locality among sender and receiver peers that we will present later in section 3.3.2. The best peer selection is based on the priority of different sender peer towards offered video quality.

3.2.2 Video Encoding

In the video streaming applications, the original video is encoded before actual transmission over the network. An efficient video encoding scheme provides a mechanism to efficiently retrieve the received video packet correctly. P2P networks are highly exposed to packet loss due to the dynamicity of peers and huge noisy traffic. In this scenario, an appropriate selection of video encoding schemes is essential. These video encoding schemes add resilience and help to recover the original video with higher QoS, however they add certain overhead as well.

There are a number of video encoding mechanisms available for the heterogeneous networks and presented in section 2.9.3. The most popular encoding schemes for streaming applications over P2P networks include layered coding (LC), multiple description coding (MDC) and scalable video coding (SVC). These schemes split the original video content into multiple layers/description/tiers that contribute to a proportion of the original video quality.

In our proposed adaptive video streaming mechanism, we focus on the use of SVC video formats as an important component of the proposed P2P system while organizing the sender peers in hybrid overlay networks. In SVC video coding, bit-streams are decodable at different bit-rates, spatial and temporal resolutions are produced. These SVC characteristics make it more suitable for heterogeneous environments especially P2P networks. At present, scalable video coding (SVC) is considered most promising video encoding format for streaming application in heterogeneous networks and terminals [103][104]. A scalable video coding is used to produce highly compressed bit-streams, to create a wide variety of bit-rates. An original SVC stream can be truncated to produce videos of different qualities, resolution, and frame rates using respectively SNR (signal-to-noise ratio), spatial, and temporal scalabilities. We select and assign the SVC video tiers to different

sender peers on the basis of their priorities level. The highest priority-level of sender peer is assigned for contributing to the base video tier.

3.2.3 Overlay Organization

In this section, we describe briefly some of the related works for the peers organization in overlay.

Zhang et al. [114] have proposed a mechanism to construct locality-aware overlay networks. The nodes are organized in the overlay networks on the basis of their distance among the neighbor nodes. Locality-aware overlay networks construction provides an efficient solution for multimedia streaming and reduces the communication costs between the end nodes. In [56], hang et al. developed a framework for live media streaming that is based on data-driven overlay networks where each node periodically exchanges data availability information with other nodes. The management of nodes which join the networks is controlled by an origin node that is persistent during the life time of the streaming session. In the proposed mechanism, a scheduling algorithm is devised for heterogeneous clients. The nodes in the data-driven overlay network are organized in a Breath-First-Search (BFS) tree, where origin node is located at level '0' while any node at level 'k' can be reached in 'k' hops from the origin.

Tian et al. [115] presented a framework for the Hybrid Overlay Structure that is based on Random Walks. The proposed organization is locality-aware where nodes are organized in different clusters. New nodes join any cluster on the basis of its distance from the cluster head. A random walk technique is proposed for the connection of different clusters.

Crespo et al. [116] proposed the "SON" mechanism for the organization of different nodes. The proposed mechanism is based on the semantic based organization of nodes. Nodes having the same type of media files are organized close to each other and therefore clustered together. The semantic based organization results into an efficient improvement of the system performance in terms of the efficiency in query search.

3.2.4 Video Packet Distribution

Video packets distribution among different sender peers is another essential design principle for the streaming systems. The video packets distribution assigns the different parts of the video to different sender peer, whereas a packet refers to a unit of data transmission. In other words, such distributions determine the role for the sender peers that which part of the video they will contribute. We stated earlier that "LC", "MDC" and "SVC" are considered most appropriate encoding schemes for the P2P-based streaming systems. Video distribution strategies assign different video layers/descriptions/tiers to different sender peers and determine the actual schedule of transmission in accordance to the actual play-back deadline. Peers with higher uplink capabilities are assigned the most important part of the video. This video packet distribution plays a major role to cope with the network dynamicity by assigning the different video packets to different sender peers. We implemented a receiver-side scheduler for video packet distribution. This scheduler determines and assigns different video parts to different sender peers on the basis of their priorities for offering the QoS. This mechanism is presented in the next chapter.

3.3 Adaptive Video Streaming

The proposed adaptive video streaming mechanism is capable of supporting QoS using multiple senders streaming towards a single receiver over P2P networks. In such a system, the receiver peer has to receive different parts of the video content from multiple sender peers. The adaptive video streaming is supported by a best peer selection mechanism, a peer/stream switching mechanism to cope with the dynamicity problem, and a quality adaptive video distribution to enhance the overall received video quality. These mechanisms are described in the following sub-sections.

3.3.1 Target Architecture

Our target streaming architecture is receiver-centric in which the receiver peer orchestrates the whole streaming mechanism using the following threefold adaptive mechanism (1) appropriate selection of the sender peers, (2) efficient switching (enabling/disabling) of peers to ensure a smooth video stream when any of the sender peers is no more available in the network, and (3) assigning different parts of the video content to specific peer in order to guarantee a stable level of QoS. Receiver peer is responsible for the selection of best sender peers among the huge set of available sender peers. It is also responsible for deciding which part of the video should be received from a certain sender peer after considering the characteristics of each sender peer and other network constraints. This real-time tracking is done regularly to monitor the peers and to detect any change that needs to be faced by the threefold adaptive mechanism.

The proposed target architecture is described in Figure 3-2. It is receiver-centric architecture composed mainly of a receiver side scheduler which is implemented for the scheduling of received video packets from different sender peers. All the video packets received from the sender peers are combined in different video buffers. To avoid the buffer underflow problem, the scheduler should consider the buffer states regularly. Peer switching and video packets assignment are performed on the basis of the current network conditions and buffer states.

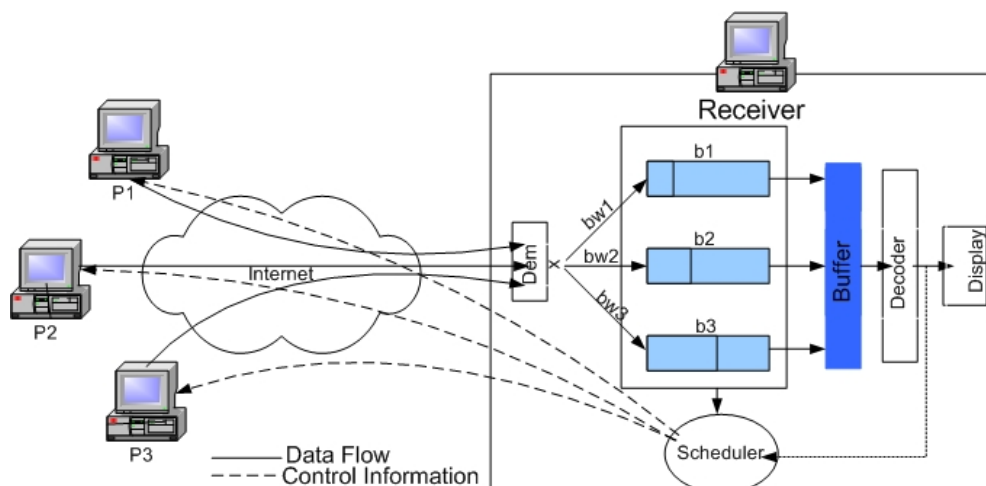


Figure 3-2: Architecture for Receiver Centric Video Streaming

In P2P networks, it is common that a single sender peer is unable to serve the whole video content/quality. Any sender peer cannot provide alone a sufficient bandwidth to provide Base tier quality, and in this case, the receiver peer should select more than one sender peers for a particular

Tier. The video packets received from each peer are stored in an individual buffer and then combined in collective buffer for certain tier. As shown in Figure 3-3, two different sender peers are selected for the Base tier (T_0, S_0, B_0). The received video packets are combined later in a dedicated buffer for Base tier quality video.

In Figure 3-3, we presented three sender peers P1, P2, and P3 which are requested to contribute to different video streams (e.g. QCIF 30 fps) by the receiver peer. This leads to use the Base-Tier (T_0, S_0, B_0), Tier 1 (T_1, S_0, B_0) and Tier 2 (T_2, S_0, B_0). All the sender peers are providing different video qualities. The most important tier is the 'Base tier'. The other video quality tiers are used to enhance the overall video quality (SNR enhancement in term of B_0 and B_1 , Temporal enhancement, and special enhancement). In our proposal, the receiver peer selects a subset of the best available sender peers. To this purpose, we perform an active peer monitoring to observe the offered QoS of each sender peer. Stream/peer switching is performed in case of important degradation of the received video quality.

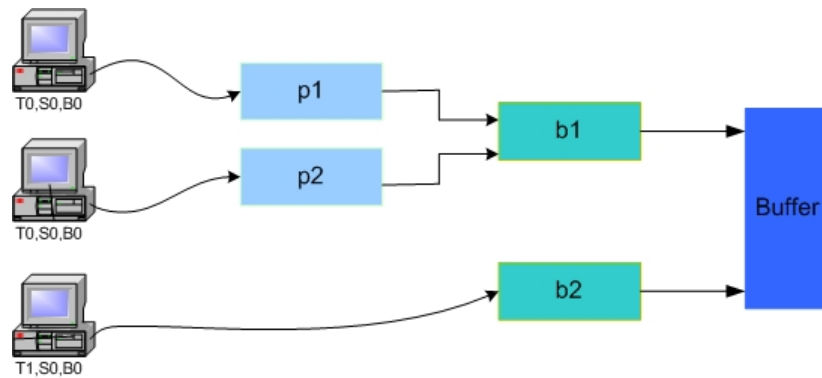


Figure 3-3: Buffering Schedule

3.3.2 Peer Selection Parameters

In the large scaled networks, like P2P, the selection of sender peers becomes more crucial and plays a vital role for the enhancement of the overall quality-of-service (QoS). In this section, we describe some of the mechanisms used for best peer selection.

Generally, in P2P-based streaming systems, a peer is called best peer when it is offering high QoS. A receiver peer intends to receive video content from its neighboring nodes to reduce the transmission delay. There are different known metrics used for the best peer selection. Many of them determine the end-to-end distance (locality-aware) among receiver peer and sender peers. These metrics are briefly described below:

- Round-Trip-Time (RTT): Round trip time (RTT) measurement is a simple metrics to determine the logical distance between receiver and sender peers. RTT measurement gives round trip time for the data transmission and its acknowledgement. RTT estimation can be considered as good candidate metrics for the best peer selection criterion, as it depicts the actual end-to-end transmission time. This metrics can be easily computed using some ping test. However, RTT measurements only provide a rough estimate and do not depict the exact underlying network conditions. RTT

estimates are dynamic and they vary over time. In this regards, an active measurement for this metrics is required throughout the streaming application session.

- Number of Hops: The total number of hops (traversed nodes) between receiver and sender peers can also be considered as another metric for best peer selection. This parameter is similar to the RTT measurement but it is considered as more stable than RTT estimates. Calculating number of hops does not take into consideration the virtual distance between two points and how much time it takes to transmit data between two points. It is possible that one peer offers low hops count but the actual transmission time is quite higher over this respective link. Furthermore, sometimes it is hard to determine the exact number of hops between two peers.
- Geographical Distance: Geographical distance measurement is a simple metrics that determines the end-to-end distance between receiver and sender peers in physical networks. This geographical distance is measured by the exact geographical locations of the peers. This metrics is based on the assumption that peers physically located near to each other can offer best connections. Although it is not true in practice, it is possible that two peers located very close to each other are having no direct connections (attached to different ISPs). There exist different projects such as NetWorldMap [117] and VisualRoute [118] that can be used for the measurement of geographical distance between peers. This metric seems easy to compute but it may leads to network inefficiency problem, if peers have no direct connections and/or require network address translation (NAT).
- End-to-End Bandwidth Estimation: The end-to-end available bandwidth between receiver and sender peers has great impact for providing the higher QoS. If enough bandwidth is available on end-to-end content transmission path, the content can be delivered quickly. However, the bandwidth estimation is not as trivial and it is really hard to determine the exact available bandwidth over P2P networks. Although, there are certain techniques like TFRC (TCP friendly rate control protocol) that provide an estimation of available bandwidth but still they are hard to incorporate for the real-time streaming applications over P2P networks.

P2P networks are constituted of thousands of peers. All the peers join the network for some specific objective. Receiver peer looks up for the list of the peers having the desired contents. Each peer from the list constitutes 'Potential Peers'. As, we have discussed earlier that many peers join the P2P network just to receive some service, so it is possible that there are some peers which have the contents but they don't want to share or contribute in streaming session. The peers having the requested contents and willing to share their contents are named as 'Candidate Peers'. The receiver peer classifies all the candidate peers according to their offered QoS. Only a sub-set of them is selected and activated to contribute to the streaming mechanism. Details of peer selection are given in next subsection.

3.3.2.1 Proposed Mechanism for Peer Selection

Different parameters have been proposed in literature for the best sender peers selection that include end-to-end available bandwidth, end-to-end delay, packet loss ratio, random selection,

offered rate, etc. However, these parameters are very hard to estimate. We have performed some tests to determine the relevant performance metrics that can be used to determine the QoS of a particular peer such as “RTT”, “number of hops” and “packet loss” using “ping” and “traceroute” commands. In some cases, “RTT” can be used as an indicator of the link quality. The number of hops count can also be the best representative metrics for the best peer selection, as it is quite similar to the “RTT” estimates and is more stable over time than “RTT”. In real-time video streaming applications, receiver peer intends to receive whole video content more quickly without taking into consideration that how many hops it has traversed during transmission. The number of hops count does not take into consideration the exact transmission time between two peers.

We have proposed in [119] that “RTT” can be used to improve the overall QoS for the streaming applications in heterogeneous networks like P2P. The motivation behind using RTT is its relationship with the offered bandwidth on the end-to-end path. A simplified version of the equation given the bandwidth (data rate) of a particular TCP-like session is given in Eq. 3-1. The MTU represents the maximum transfer unit and the “Loss” represents the packet loss rate [120].

$$Bandwidth = 1.3 * MTU / (RTT * \sqrt{Loss}) \quad \text{Eq. 3-1}$$

It is commonly agreed that the lower the “RTT”, the more the offered data rate (bandwidth) is higher. Thus, peers offering lower RTTs are considered as having a higher available bandwidth and that may be shared fairly among the services. However, in certain asymmetric networks (such as ADSL networks), the best peer selection based on only RTT measurement may not be suitable and requires certain consideration like the number of Hops between receiver and sender peers.

In this scenario, we select the best sender peer on the basis of certain. These priorities are determined by end-to-end active probing among receiver and sender peers and by taking into account the number of hops between these peers. Thus, we can incorporate different parameters for the best peer selection depending on the network types and other constraints. Eq. 3-2 can be used for assigning different priorities for these parameters by assigning different weights w_1 and w_2 .

$$P_i \approx \frac{(RTT \times w_1) + (\#Hops \times w_2)}{w_1 + w_2} \quad \text{Eq. 3-2}$$

In Eq. 3-2, P_i represents the relative priority of the sender peer. We performed different simulations tests to determine the optimal values for w_1 and w_2 that can be used to determine the priority of each sender peer. In these tests, the end-to-end measurement RTT measurement was performed between each sender peer and receiver peer. We found that ($w_1 = 0.7, w_2 = 0.3$) determine the best priority of sender peers. These values were determined after performing measurement for different set of values. The details of these measurements are given in section 3.4.2.

It is noted that burst traffic may cause a transient congestion that can affect the priority estimates. To deal with this issue, we used a low-pass filter to calculate a smoothed value of priority measurement based on the smooth RTT value. The low-pass filter is an exponential weighted

moving average based on EWMA (Exponentially Weighted Moving Average) [109] [110] Chart. EWMA 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 following equation (Eq. 3-3):

$$X_{New} \leftarrow (1 - \lambda) * RTT + \lambda * X_{Old} \quad \text{Eq. 3-3}$$

In this equation, X_{New} and X_{Old} represent the new and the old estimates while measuring the smooth RTT value. When choosing “ λ ”, it is recommended to use smaller values to detect small shifts and larger values (between 0.2 and 0.4) for large shifts [110]. In this mechanism, we used the small value ($\lambda = 0.2$) that is commonly used to detect small shifts.

This leads us to propose an efficient peer selection mechanism based on relative peers priorities. Any peer is considered as an important peer if it offers the highest priority and will be assigned to contribute to the SVC base video tier. Additional sender peers can be added or removed during the streaming session depending on the available network resources and conditions to maintain at-least a minimum level of video quality during the streaming session.

3.3.2.2 Tracked Peers

We have observed that “RTT” value plays an important role in determining the sender peers priorities along with the number of hops. Receiver peer calculates the “RTT” measurement among each candidate peers and itself. This mechanism works as follows:

Receiver peer sends a query to search for the desire media contents and maintains a list of all the candidate peers with whom receiver can start the streaming. Table 3-1 gives an example of two peers categories considered in the streaming system. We perform a content search for some video content and the top 5 values are shown in the Table 3-1. They are taken from track announces of the Internet on February 2007. Peers categories are:

- Seeds: uploader who has the complete file.
- Leechers: uploader who is also downloader.

Content	Seeds	Leechers	Total
Content 1	22370	22870	45240
Content 2	15959	10927	26886
Content 3	4261	1703	5964
Content 4	1448	738	2186
Content 5	1210	677	1887

Table 3-1: Example of Seeds and Leechers

It is obvious, after performing these measurements, that it is not scalable to track all the available peers and to measure the offered QoS of hundreds of peers. In this regard, our adaptive streaming mechanism takes into consideration only a subset of peers called “Tracked peers” containing a “MaxThreshold” of peers from the candidate subset. This threshold value specifies the maximum number of candidate peers that the receiver can track to measure the offered QoS. The subset of the tracked peers is either constructed randomly by taking MaxThreshold number of peers from the candidate subset or using first coming query/response notification. In fact, the QoS measurement of individual peer is receiver centric. The receiver peer transmits a “Hello” ping packet to all the tracked peers. This “Hello” packet behaves like a ping test to calculate the priorities of the targeted candidate peers.

Once peers are tracked, the receiver peer categorizes all the tracked peers according to their priorities. It selects a subset of these peers with a highest priority. The peers are classified into different overlay networks as described in the following sub-section and the selection of candidate peers belonging to different overlay networks is carried on the basis of relative priority of sender peers and the offered QoS. After forming the groups of tracked peers based on their priorities, receiver peer sends request to start streaming with a subset of “active peers” having high priority. The scheduling mechanism is used for this effect to coordinate and is described in Chapter 5. Figure 3-4 summarizes the hierarchical classification of peers used along the proposed adaptive streaming mechanism.

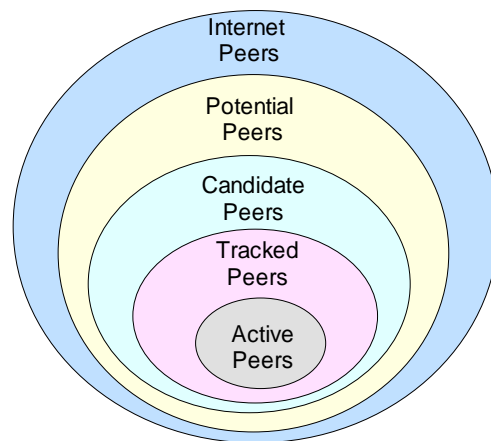


Figure 3-4: Hierarchical Peer Classification

We organize the sender peers in different overlay networks to facilitate the best sender peer selection mechanism. The overall peers organization in these overlay networks along with the best peer selection and peer/streaming mechanisms are described in following sub-section.

3.3.3 Peer Overlay Network Organization

In this section, we describe the proposed overlay network organization for the sender peers. In P2P-based video streaming mechanism a receiver peer maintains the list of entire tracked peers which have the requested media contents and are ready to participate in the streaming process. In this overlay organization, the sender peers are organized in different hybrid overlay networks based on the offered content (i.e. which SVC tier is provided per peer) and based on the QoS offered by

the peer (relative priority). This organization is further used for the selection of the best sender peers and for the efficient P2P management to handle the dynamicity of peers.

The proposed solutions allow the organization of peers in overlay networks on the basis of distances, i.e. locality-awareness. The motivation behind our proposed mechanism in using overlay networks for the best sender peers selection is to support heterogeneous clients and to improve the QoS while efficiently utilizing the available network resources. In our proposed organization, we investigate the organization of these peers to provide an efficient mechanism for the best peer selection. Best peer selection mechanism leads to enhance the overall received throughput and to reduce the overall transmission delay and packet losses that are more crucial for real-time streaming application. In the proposed hybrid overlay networks mechanism, we organize different sender peers in different overlay networks considering both (1) the semantic of the video and (2) the QoS offered by each peer. The classification of these sender peers is based on the offered video contents (semantic-aware) and on the probed end-to-end QoS based on priority of sender peers (quality-aware). We propose to organize the sender peers using the “MinHeap” tree structure. In this organization the best available peer is always present at the root of tree. The hybrid organization of the sender peers make our proposed overlay management mechanism distinguished from prior studies by organizing the sender peers with respect to their offered QoS that makes our mechanism more promising for the multimedia streaming/video on demand services with improved QoS by efficient utilization of the available networks resources.

In this overlay organization, we have slightly different objective for the organization of the sender peers in overlay networks. We aim to facilitate the receiver peer to select the best sender peer to enhance the overall QoS for the offered services. In this chapter, we present a mechanism to organize the sender peers in hybrid overlay networks. These overlay networks are an abstraction and they differ from the classic overlay networks used for the structured P2P networks as described in section 2.2.2.2. The main goal for this hybrid based organization is to bring sender peers closer to the receiver peer and to provide an efficient mechanism for the selection of best peers with the required video quality. The overlay networks can be classified on the basis of peers’ organization criterion. Here, we described some of these classifications along with our proposed hybrid peers organization.

3.3.3.1 Locality-based peer organization

Locality based peer organization is performed on the basis of geographical/physical distances between sender and receiver peers. The locality of peers is determined by different parameters, for example, Euclidean distances, logical distances estimated using number of intermediate peers (e.g. number of Hops, TTL measurement, etc.), and using end-to-end round trip time (RTT) measurements. Locality-aware organization techniques primarily intend to reduce the transmission delays by considering the logical distance between the source peers and the receiving peers. In this scheme, video content are delivered on the shortest logical paths for each service request. Such locality based peers organization results in the minimizing of start-up delay and latency during the streaming mechanism.

3.3.3.2 Semantic-based peer organization

The characteristics and the structure of media contents are useful for the selection and organization of the appropriate peers in overlay networks. In semantic based organization, peers with the same media contents or offering the same video quality are placed together in the same overlay network. Semantic based peers organization influences the delivery of important parts of the audiovisual content to the important peer. Such organization makes it convenient to search media contents rapidly. Semantic based peer organization can be a good choice for scalable video coding and in other coding schemes where media contents are distinguished according to priority. We described earlier that we use SVC scheme for the video encoding that is helpful to adapt to the actual network conditions by adding and/or dropping the video quality tier with respect to the available end-to-end bandwidth and received QoS. However, our proposed mechanism is well suited for all video formats that can provide priority between sub-streams. Thus, we assign the Base tier quality to the most important peer in terms of quality of service, i.e. the best peer must be selected for the most important quality tier to enhance the overall QoS.

3.3.3.3 Community-based peer organization

In community based peers organization, participating peers are organized in different communities/clusters on the basis of their specific characteristics by forming common interest groups (CIG). These community based overlay networks may include educational /training institutes, medical care units, libraries, leisure clubs, stock markets, banks, etc. In this type of organization, all the peers have some knowledge of the other peers sharing the same interests, which is helpful for further cooperation.

3.3.3.4 Hybrid overlay organization

As stated before, we propose hybrid peers overlay networks to organize the sender peers in different overlay networks. This overlay organization is just an abstraction to organize the sender peers to facilitate the best sender peer mechanism. It is influenced by combining locality based (or quality-based) organization and semantic based organization schemes. In this hybrid overlay organization, sender peers are arranged in overlay networks. The streaming mechanism is receiver-centric where the receiver peer creates, organizes, and manages these overlay networks. The sender peers offering the same video quality tiers of the same media types are placed at the same level in overlay networks. Within each level of the overlay networks sender peers are organized according to locality-aware peer's organization. Our mechanism uses the relative priorities for sender peers offering QoS as described in Eq. 4-2. In fact, the best peer is the one offering the best QoS (i.e. having highest priority). This peer should be selected by the receiver to contribute to the base tier of the video.

A hybrid organization of the sender peers in different hierarchies is illustrated in Figure 3-5. Here, sender peers are divided into three different levels of overlay networks. We exploit the "MinHeap" Tree structure for the organization of peers within each overlay network. A "MinHeap" is a minimum complete binary tree in which the value in each node is less or equal to those in its children. In such tree structures 'n' peers can be organized in $\Theta(n \log n)$ time.

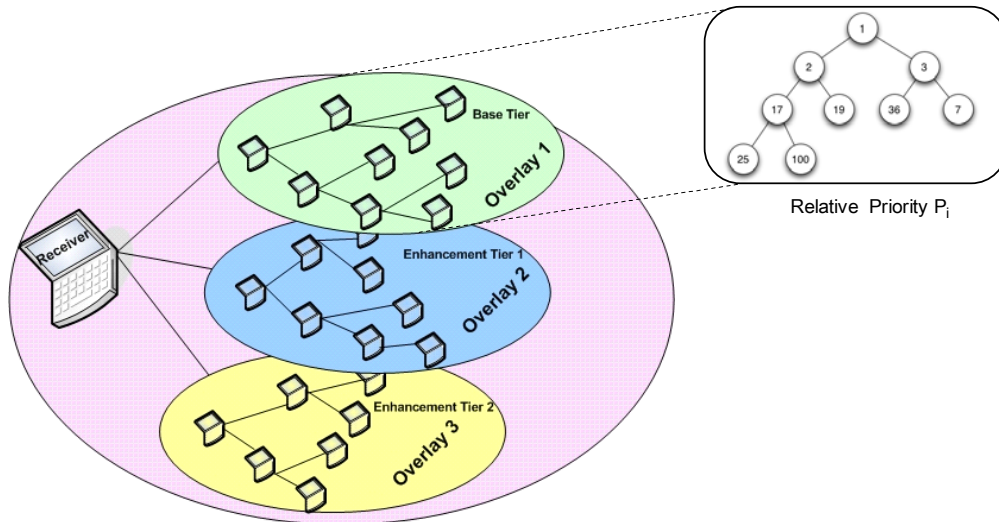


Figure 3-5: Layered Organization for Sender Peers

The sender peers are organized using the “MinHeap” tree structure in each overlay network. “MinHeap” tree structure are considered more suitable when the number of candidate peers are very high, however a sorted array is sufficient to serve the purpose, if the candidate peers are limited to some peers. The organization of peers inside the “MinHeap” tree structure is based on the relative priority of sender peers as shown in Figure 3-5. We have noticed peers offering best priorities (P_i) are always present at the root of the overlay tree. Furthermore, the hybrid organization provides an efficient mechanism to select the best peer offering the best QoS and having the most important SVC tier to deliver.

3.3.4 Proposed Adaptive Streaming Mechanism

The main components of the P2P based real-time video streaming and video on demand services include: indexing of the media contents available to each sender peer, signaling among the peers for their activation, organization of sender peers in virtual overlay networks, media contents retrieval, selection of best sender peer to serve for subsequent request, sender peers switching, selection of quality adaptation path, etc.

In this chapter, we do not discuss the issues of indexing, signaling, and data retrieval mechanisms. In our proposal, we are more concerned with the organization of peers in overlay networks for the efficient sender peers selection to enhance the overall QoS.

We determined to focus on the SVC scheme to propose mechanism for adaptive streaming over P2P networks. In Figure 3-6, we have shown how different video tiers are selected from the hybrid overlay networks where sender peers are present in different multi trees. This overlay organization provides a mechanism to select multiple sender peers from multiple trees that comprises the peers offering different video quality tiers.

In Figure 3-6, we have shown an example of the peer’s selection based on the horizontal pattern. Here, the target video quality is (QCIF, 30fps). A receiver peer selects sender peer to receive all the quality tiers offering (T0, S0, B0), (T1, S0, B0), and (T2, S0, B0) to achieve the perceived video quality. The corresponding video tiers are selected from the multiple trees

organized in the hybrid overlay network management mechanism. However, additional sender peers can be added to receive as necessary video tiers to achieve the desired quality and the mechanism is adaptable for the other patterns including vertical, diagonal and ZigZag as described in section 3.3.4.4.

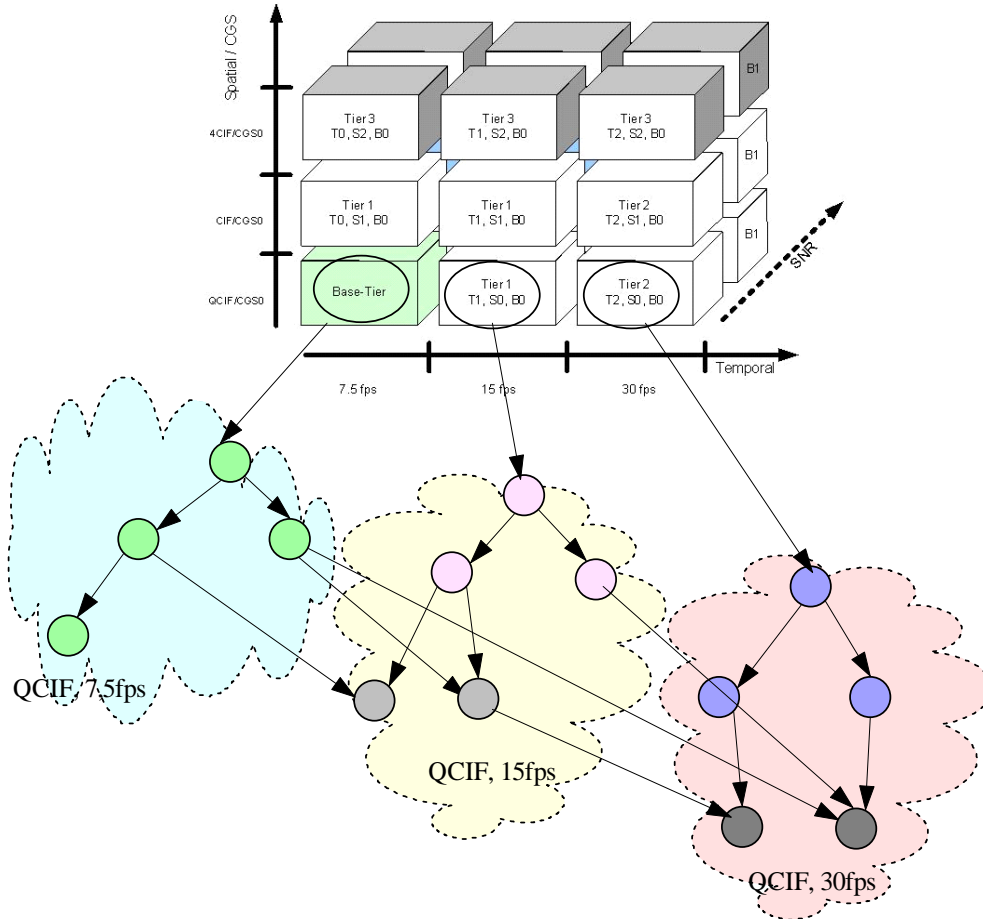


Figure 3-6: Sender Peers organization in Multiple-trees

3.3.4.1 Best Peer Selection Mechanism

We propose to select multiple sender peers from different overlay networks to address the above mentioned issues. The selection of the best peers is performed by assigning the best quality tier (Base tier) to the best sender peer. A peer is considered as the best peer it has highest priority for the offered QoS. Our hybrid overlay networks organization facilitates the management of peer selection. In such organization, a best peer in each overlay is always located at the root of the “MinHeap” tree due to the “MinHeap” tree structure, and is accessible to the receiver peer in $\Theta 1$ time. In this way, receiver peer selects different sender peers from different overlays depending on the perceived video quality, and the received video are decoded to achieve received video with the highest quality.

3.3.4.2 Peer/Stream Switching Mechanism

P2P networks possess highly dynamic characteristics and show unpredictable behavior i.e., any peer node can enter or leave the network without any prior notification. In this situation, P2P

architecture is not considered much reliable for the video streaming and other real time applications which require smooth connection establishment throughout the application lifetime. Video streaming applications require longer and stable end-to-end connection establishment. However, in P2P networks the network conditions may change drastically during the long transmission sessions. High dynamicity and diversity of peers can also vary during the course of the session. This is due to (1) a sender peer crashes/stops contributing the media content, (2) shared bandwidth or network conditions are changed, (3) some new peers enter the network and provide better bandwidth sharing with low RTT value, and (4) heavy traffic can cause more packet loss, higher inter packets delay which ultimately degrades the overall QoS. It is clear that the receiver QoS is totally dependent on the way in which active peer are selected. Likewise, it also depends upon the streaming mechanism's ability to cope with the network degradation and peers failure problem.

In our proposed adaptive streaming mechanism, peer failure is detected in two ways. First, if receiver peer does not receive any response against the 'Hello' message (ping test). After a reliable tentative count and without receiving any response, the sender peer is considered no more available. The frequency of such ping test is discussed in performance evaluation section. Second, we assign different receiver side buffers to store the video packets coming from different sender peers, if the number of received packets present in the buffer are below some certain limit, we conclude that either peer is disconnected or network links are becoming more congested and they are no more capable to support the video delivery. In these cases, currently active peer is dropped from in the active subset of peers and another best peer is selected from the available candidate peers.

Network fluctuations are considered as another problem faced by the video streaming applications and that degrades the received video quality dramatically. Such network fluctuations arise due to congestion over certain active network links. In our mechanism, we can detect these fluctuations while determining the priorities for sender peers. If priority for a respective peer is becoming very low, the receiver peer assumes that such peer is no more suitable for the streaming session to offer high QoS. This issue is handled by the peer switching mechanism in two ways. First, if there are some peers' available offering high priority values in the subset of tracked peers, the effected peer is replaced with this peer to maintain a smooth video packet delivery. In other case, if no other best peer is available, then stream switching is enforced instead of peer switching. In stream switching, receiver peer re-adjusts the received video streams dynamically. While re-adjusting the video streams, the most important part of video is assigned to the currently best available peer that may be chosen already for some other video parts. Thus, the received video quality is adapted to the current network conditions. In the worst scenario, if it is hard to cope with the network condition, we propose to drop the enhancement video tiers of SVC video. We focus to maintain at-least the base video quality throughout the streaming session. The minimum quality assured to the user is the quality provided by the base tier of SVC encoded video.

The incorporation of peer/stream switching mechanism ensures the smooth delivery of video content with adaptive QoS. However, the incorporation of peer/stream switching may lead to another problem of video quality fluctuation. The quality fluctuation problem is encountered when the frequency of the peers switching is high. It is not desirable to switch from one peer to another peer whenever a new peer offering high priority value enters the network. Switching from one

active peer to another peer each time when there is change in priority value results into oscillation effect. The peer/stream switching causes some extra overhead that should be taken into consideration. Such oscillation effects degrade the received video quality instead of ensuring the smooth video delivery. In our proposed mechanism, we incorporate exponential smoothing technique for priority estimates as described in Eq. 3-3 that is used to mitigate these oscillation effects.

3.3.4.3 Is best peer always remains the best?

Our targeted architecture for the proposed solution is based on many to one streaming, i.e. a single receiver selects multiple sender peers to receive different parts of the requested media contents. In this scenario, a natural question arises, what happens when a number of the receiver peers request for the same media contents? Every receiver peer should prefer to select the best sender peer to receive the contents of the highest quality, in this case due to a certain overload on the best peer; can this best peer still remain the best peer throughout the streaming session? Similar scenarios were discussed and applied more often in the “Game Theory” for multi players’ games. Nash presented in his classical “Nash Equilibrium” theory [122] that if every player of the certain game chooses the best policy then no player can end with higher scores. To interpret this theory for streaming application, we can state that if every receiver peer intends to select the same best peer then no receiver can get the entire video with good quality and there might be a system crash due to the heavy load (in the case of large number of receiver peers). The best peer for a particular session will not remain the best for the session.

The proposed overlay management tends to address the above question. The proposed peer organization and their selection provide an efficient way of handling this issue. Firstly, our proposed solution is receiver-centric, and secondly it is based on the probing of sender peers respective priorities as an indicator of the QoS. We noticed that the probed end-to-end priority is not the same for all the receiver peers. The probed priority becomes lower for the later receiver peers trying to select this peer as sender peer. The priority value difference is due to the additive transmission delay with a prior receiver peer. The proposed overlay management mechanism is receiver-centric, thus all the receiver peers maintain their respective overlays of sender peers and candidate peers in their own prospective. In this organization, the overlay network for one receiver peer should not be exactly the same for the other receiver peer. Thus, due to these features of the hybrid overlay networks, the best peer for one receiver might be the worst sender peer for subsequent receiver peer.

3.3.4.4 Quality adaptation

In the real-time streaming applications, an intelligent packet ordering scheme plays a significant role for the perceived quality. It can be beneficial to improve the received video quality by utilizing the available resource efficiently. Packet ordering for the quality adaptation not only ensures the base video quality for streaming session but also adds resilience for the lower video layers/tiers. The packet’s ordering highly depends on the receiver preferences towards the perceived quality. SVC based video coding offers the receiver to select the video either with high SNR quality, temporal resolutions for varying frame rates, or spatial resolution for the different resolutions, etc. All these parameters enhance the video quality while focusing on certain

parameters. The detailed quality adaptation mechanism with different quality patterns is described in details in section 4.3.5.

3.3.4.5 Support for Heterogeneous Networks Terminals

P2P networks are composed of different heterogeneous networks terminals (clients) that connect to the network through heterogeneous network connections. These clients differ from each other in terms of bandwidth, processing power, memory constraints, shared content, etc. An example of the capabilities towards the received video content for streaming and video-on demand services is presented in the Table 3-2.

Network Device	Spatial Resolution	Temporal Resolution (frames/sec)	Resolution (Pixels)
Mobile Phone	QCIF	7.5	176 × 144
PDA	CIF	15	352 × 288
Laptop/Desktop	CIF/4CIF	15/30	704 × 576
TV	9CIF/16CIF	30	1056 × 864 1408 × 1152

Table 3-2: Heterogeneous Device Capabilities

This overlay based peer’s organization supported by the SVC video coding provides an efficient mechanism to select the sender peer in accordance to the end client terminal capability. Furthermore, this approach provides better scalability and dynamicity for peer-to-peer streaming mechanism. The overlay based peers management facilitates the selection of the sender peers for distinct heterogeneous terminals where different video quality tiers are selected from overlay networks according to the terminal capabilities. The quality adaptation to the video packets ordering is adapted accordingly to the respective network terminal.

Scalable video coding is based on the idea of the universal media access. The objective behind such encoding scheme is to encode the original video content only once and afterward the customized video streams/tiers can be used “Anytime” from “Anywhere” (using any access network) and by “Anyone” (using any terminal capability). Thus, the use of the SVC enables the real time adaptation capabilities for the heterogeneous networks and terminals. The end-users are free to select video stream in more customized way.

3.4 Performance Evaluation

This section describes the simulation results of the proposed adaptive packet video streaming mechanism. The simulations for the proposed streaming mechanism were performed using NS-2 simulator [112].

3.4.1 Simulation Scenarios and Evaluation Model

The network model considered for simulations is presented in Figure 3-7. For the test cases, we have generated four different video tiers using MPEG-4 trace files [113] providing different

qualities of the video. We distribute the original video quality tiers equally among different sender peers to see the reaction of the proposed mechanism. In this network topology, we considered only the candidate peers that agree to participate in streaming mechanism. These peers are organized in different overlay networks. We attached a peer “R” which represents the “receiver” for playing-out the received real-time packet video. Receiver peer “R” orchestrates the whole streaming mechanism using the adaptive mechanism (1) selection of sender peers from overlay networks, (2) adaptive peer/stream switching, and (3) quality adaptation. In this simulation, end-to-end RTTs are calculated among all the sender peers and the receiver peer itself. RTT values are only indicative that are used for estimating the relative priorities of sender peers and do not really reflect the current network conditions. However, they give sufficient information to assign certain priority for the sender peer. These measurements are further used for determining the priority of each sender peer. MaxThreshold is set to 100. It represents the number of peers in the tracked subset. It allows enhancing the scalability of the system rather than tracking all candidate peers.

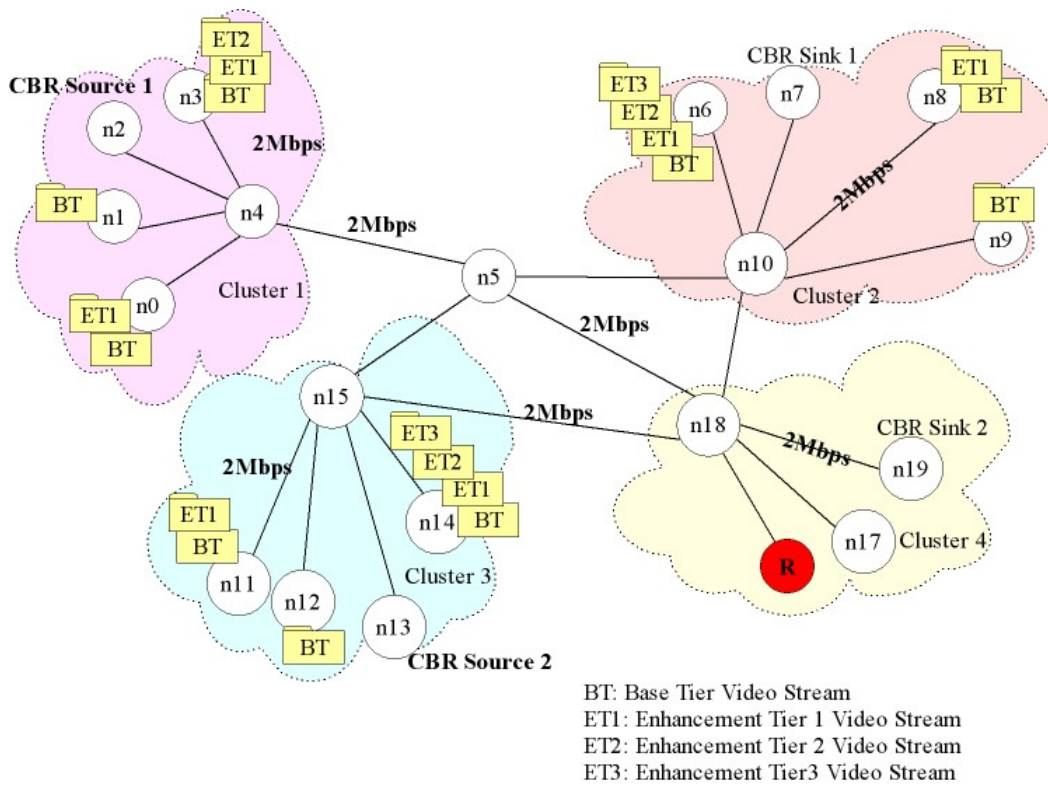


Figure 3-7: Simulation Topology

The sender peers are organized in different overlay networks according to their relative priority and the offered video quality tier. A sender peer having the highest priority appears on the root of each overlay network because of the MinHeap tree structure based organization.

Receiver peer selects and activates best sender peers from different overlay networks (multiple trees model) depending on the respective priority value for the offered video quality tier. Each sending peer sends different tiers of original video file which can be used to reconstruct video file with better quality at receiver node “R”. The video tiers are generated on the basis of respective temporal, spatial, and SNR dimension. For the simulations, Base video tier (I0, S0, B0) offer 40 %

throughput of original file, and the other video tiers (T1, S0, B0), (T0, S1, B0), and (T1, S1, B0) offers 25 %, 20%, and 15% throughput of the original media file respectively as shown in Figure 3-8.

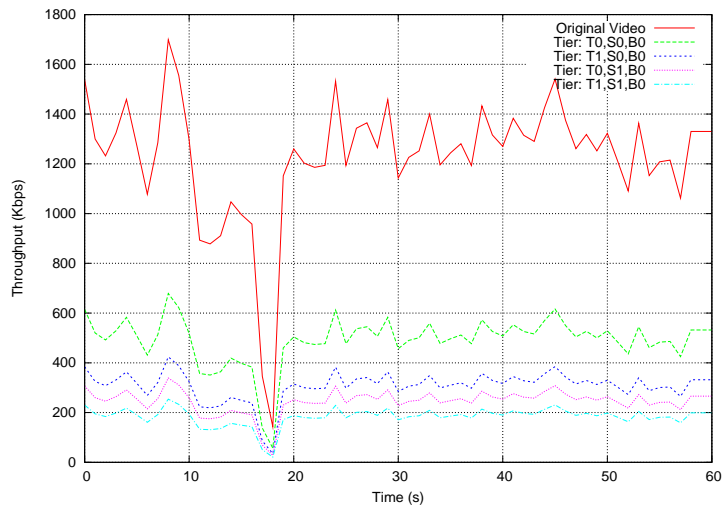


Figure 3-8: Instantaneous throughput of the different Video Quality Tiers

To incorporate the dynamic behavior of peers, we use “on/off” distribution. Peers enter and leave the system according to on/off distribution with “on duration” equal to “off duration” and set to 4s. In real scenarios on/off duration may vary and even peers remain available for quite more than 4sec, but we choose these estimations to incorporate highly dynamic behavior of sender peers. Peer n3 behaves dynamically in this configuration with regular intervals of time. It leaves the system (i.e. it goes down) at time 4s and again it enters the system at time 8s. The overall video throughput of the different video tiers is shown in Figure 3-8. No source node is providing 100% throughput of the original video. However, if “R” receives all the respective tiers it is possible to reconstruct the original file with higher targeted quality with the characteristics of SVC encoding scheme.

The duration of the simulation is 60 seconds. To overload the network, “CBR source 1” is attached to n2. It is started at time 5s and stopped at time 55s. “CBR source 2” is also attached to n13. It is started at time 10s and stopped at time 50s. Both sources have a throughput of 512 Kbps to load the network. We simulate the adaptation mechanism for two scenarios and repeat around 10 times. The presenting results corresponds the average results of these simulations. We simulate our quality adaptive streaming mechanism to evaluate the overall QoS in two scenarios. We conduct the simulations in two steps (1) without integrating the overlay organization for the sender peers and (2) with integration of overlay organization of sender peers. We run the simulations 10 times for each scenario and the result presented in the following sections are the average of the estimates.

Scenario without Quality Adaptation: Simulation without applying any quality adaptation mechanism. In this case, P2P system works like in downloading mode. The sender peers are selected randomly.

Scenario with Quality Adaptation: Simulation with quality adaptation mechanism is performed by sender peers selection and their switching based on active measurements of network links. Receiver peer selects the best sender peer based on relative priority. A receiver side scheduler is implemented to decide that certain video quality tier should be received from a certain sender peer. We applied the peer/stream switching mechanism when an active peer leaves the system or is unable to contribute the offered QoS. First, we perform simulations without considering overlay organization for the sender peers while later we integrate overlay organization in the simulations.

3.4.2 Performance Metrics and Simulation Analysis

3.4.2.1 Finding Optimal Values for W_1 and W_2

We have described earlier that we determine the priority of each sender peer using Eq. 3-2. We conduct simulations for different values of w_1 and w_2 ranges from 0.1 to 0.9 to compute the overall received video throughput and the overall packet drop ratio for each set of values.

A brief description of the results for different set of parameters is preset in Table 3-3. The network topology is based on Figure 3-7 with random nodes attached to nodes n4, n10, n15, and n18. Link capacities are also chosen randomly on uplink and download links (from 0.5Mbps to 2.5 Mbps). The results presented in the table are averages over 10 runs of simulations. We noticed that the receiver peer “R” receives maximum video throughput and low packet drop ratio for the set of values ($w_1 = 0.7, w_2 = 0.3$) for calculating the relative priority value P_i . Thus, we give a weight of 70% to “RTT” measurements and 30% to hops count.

	$w_1=0.1, w_2=0.9$	$w_1=0.2, w_2=0.8$	$w_1=0.3, w_2=0.7$	$w_1=0.4, w_2=0.6$	$w_1=0.5, w_2=0.5$	$w_1=0.6, w_2=0.4$	$w_1=0.7, w_2=0.3$	$w_1=0.8, w_2=0.2$	$w_1=0.9, w_2=0.1$
Average Video Throughput (kbps)	1066.31	1064.00	1014.51	1154.20	1167.34	1061.08	1230.11	1196.25	1217.92
Packet Drop Ratio (%)	13.32	13.74	17.87	6.56	6.28	14.01	0.06	2.91	1.07

Table 3-3: Priority Estimates for Sender Peers

Our proposed quality adaptation mechanism is highly influenced by the priority measurement among receiver peer and each sender peer. The frequency of such priority estimates is really important in this context because such estimates cause extra overhead over the system. For the real-time applications, excessive overhead is not acceptable. So, there should be some optimal value for the frequency of priority measurement. In this regard, we performed different experiments with different frequencies of “RTT” estimates that is further used for the priority estimation. Our experiments varied from frequency of “RTT” measurements from 1 second to 6 seconds. Table 3-4 describes the packets drop ratio (PDR) and average packets delay under different frequency ranges. After analyzing the results, the frequency of the RTT estimates was selected to 3 second for determining the priority of each sender peer, i.e. receiver peer performs active monitoring for all the sender peers and candidate peers after 3 seconds. Moreover, for the peer switching,

exponentially weighted moving Average as described in section 3.3.4.2 is introduced to maintain the stability of the system. For the simulations, we consider the last 10 priority estimates before enforcing the peer/stream switching.

RTT Frequency (s)	Packet Drop Ratio (%)	One Way Delay (ms)
1	8.967011	50.42251
2	13.0797	45.75536
3	10.22612	45.09204
4	13.14482	56.76277
5	15.41529	107.3248
6	13.03923	43.71222

Table 3-4: Frequency of RTT measurements

3.4.2.2 Simulation Analysis without Overlay Organization

We perform simulations for the both scenarios (With Quality Adaptation and Without Quality Adaptation) for the two cases as we described before. We did not take into consideration the overlay organization for the sender peers and perform simulations for the best peer selection mechanism with incorporating peer/stream switching.

Figure 3-9 shows the received video throughput at receiver node “R” in both scenarios. We notice that the quality adaptive mechanism enhances the received throughput as compared to the scenario without quality adaptation. The overall packet drop ratio for both scenarios is presented in Figure 3-10 whereas, the packet drop ratio for the individual video tiers is presented in Figure 3-11(a) and Figure 3-11(b). The received video throughput is less than the expected throughput even in the scenario with quality adaptation and the overall packet drop ratio is not acceptable for the video streaming applications. To overcome this issue, we incorporate the overlay organization for the sender peer. The results of simulations with overlay organization are shown in following sub-section that enhances the overall received packets throughput and lowers packets drops.

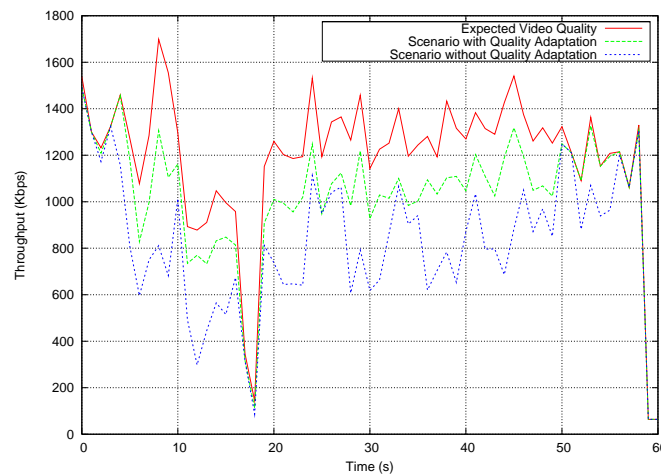


Figure 3-9: Received Video Throughput without Overlay Organization

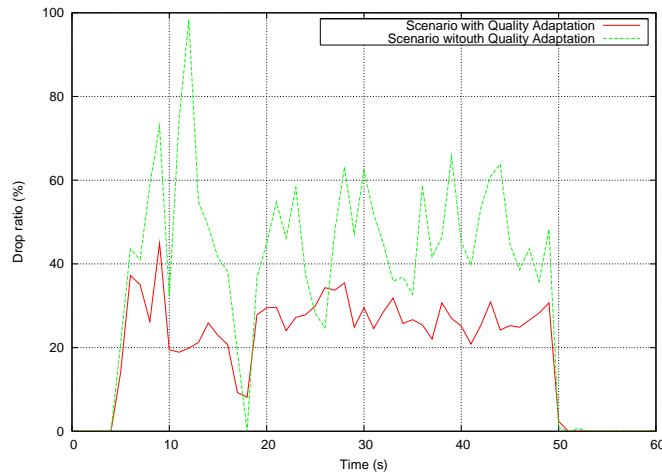


Figure 3-10: Packets Drop Ratio without Overlay Organization

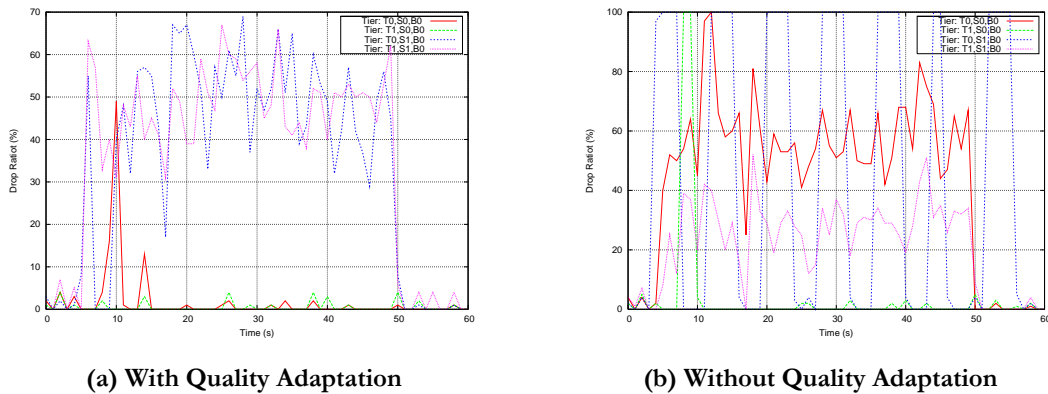


Figure 3-11: Packet Drop Ratio for individual Video Quality Tiers without Overlay Organization

3.4.2.3 Simulation Analysis with Overlay Organization

We integrate the overlay organization for the sender peers and again perform the simulations for both scenarios. Figure 3-12 shows the received video throughput at receiver node “R” in each scenario i.e. with our proposed quality adaptation streaming mechanism (Scenario 2) and without applying quality adaptation mechanism (Scenario 1) along with the expected video quality while using the three different quality tiers. We observe that the adaptation mechanism maximizes the received throughput as compared to the scenario without quality adaptation. The received video throughput is less than the expected throughput even in the scenario with quality adaptation. The reason is that the network is heavily stressed by CBR/UDP traffic. The background CBR traffic results into a lot of packets drop which are presented in Figure 3-13. We observed that packet drop ratio is much less in the scenario with quality adaptation as compared to scenario without adaptation. The proposed mechanism enhances the overall throughput and lowers packets drops which results into an overall improvement of QoS.

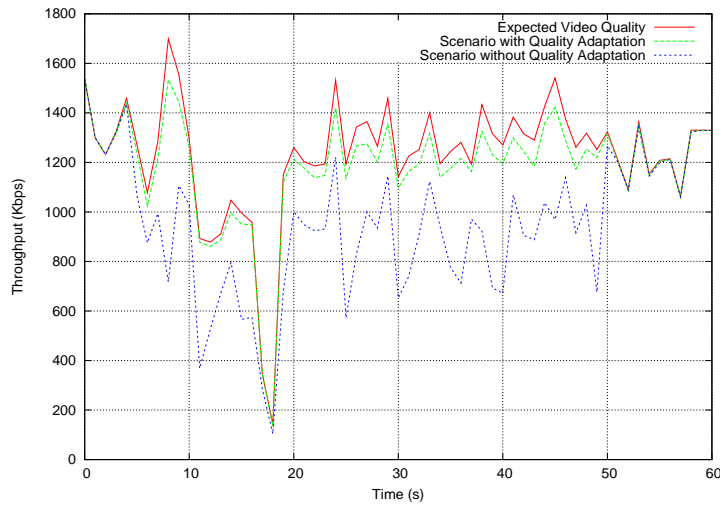


Figure 3-12: Received Video Throughput with Overlay Organization

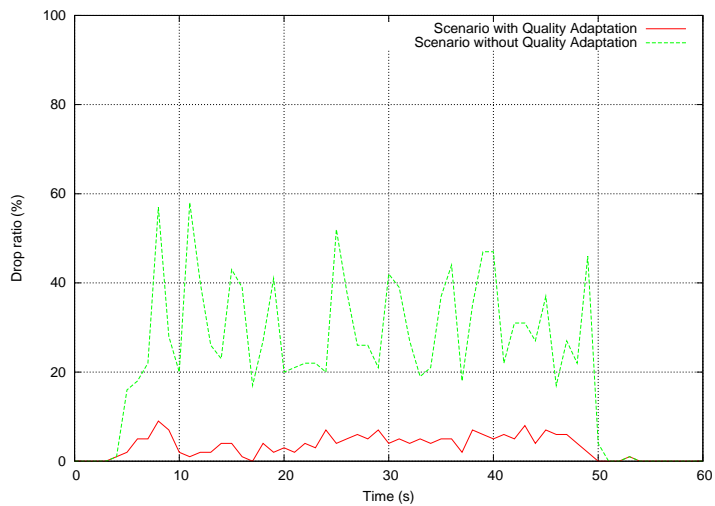


Figure 3-13: Packets Drop Ratio with Overlay Organization

The receiver peer monitors all the active network links constantly and selects best sender peers. The stream switching is done by selecting the best sender peer offering better QoS (high bandwidth share). Dynamicity of peers was performed by on/off distribution. The selection of only one peer from each overlay network is proposed to avoid congestion on certain network links because sender peers present in same overlay may share common link to connect to the superpeer, however that can be exact scenario in real. It is not necessary that all the peer forming same overlay network share the same bottleneck link. Figure 3-14 (a), Figure 3-14 (b), Figure 3-14 (c), and Figure 3-14 (d) describe the comparison between received and original video quality of base tier and other enhancement tiers. As shown in these figures and since peers selections are based on active measurement, most of drops are located at tier 3 (T0, S1, B0) and tier 4 (T1, S1, B0). This does not affect considerably the received quality at node “R” that results in a smooth quality when base tier is maintained. The received base tier and tier 1 ensure smooth video delivery at receiver side. Packet drop ratio of the base tier, tier1, tier2, and tier3 is represented in Figure 3-15 (a) and Figure 3-15 (b) for the scenario with quality adaptation and without any quality adaptation respectively.

We have noticed that the integration of the overlay organization of sender peers have improved the overall received video throughput. Figure 3-9 and Figure 3-12 shows the significant difference in the received video throughput in both cases when we perform simulations with peers overlay organization and without peer overlay organization.

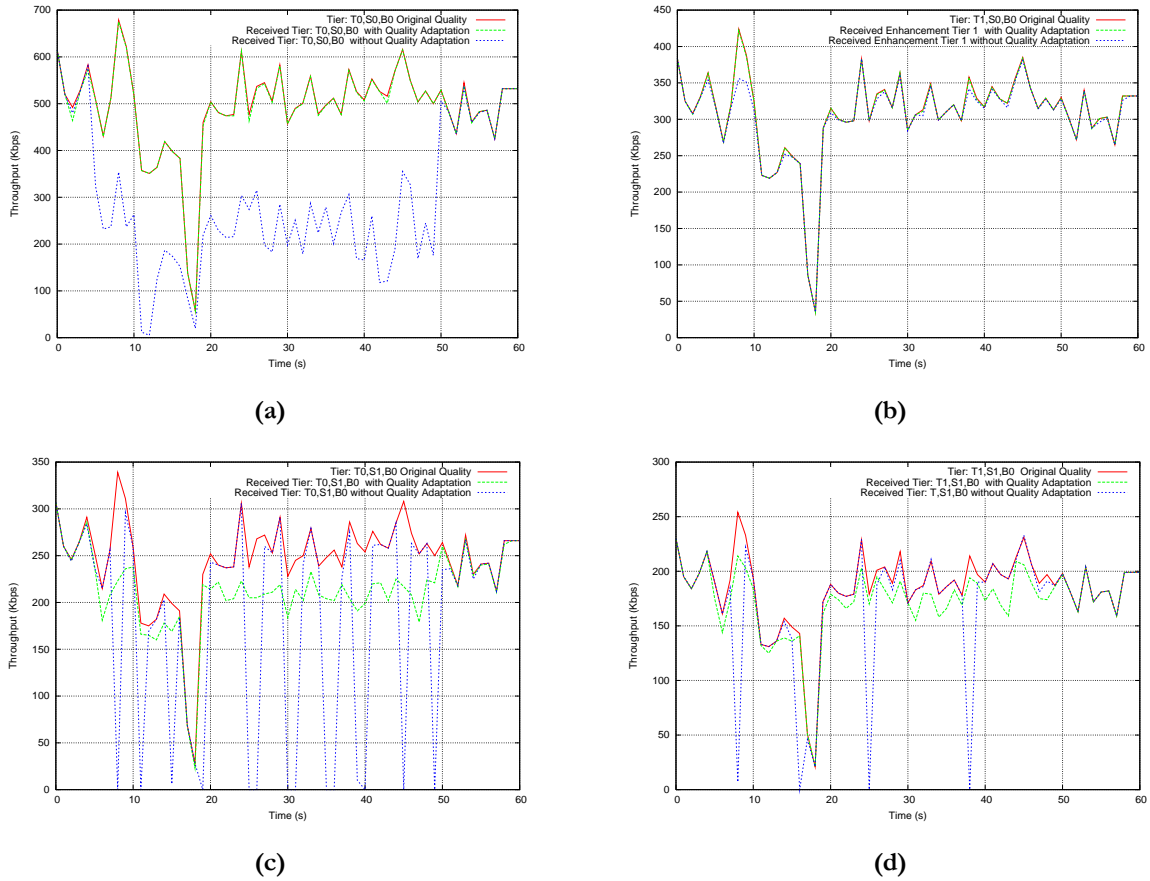


Figure 3-14: Received Video Quality for individual Video Quality Tiers with Overlay Organization

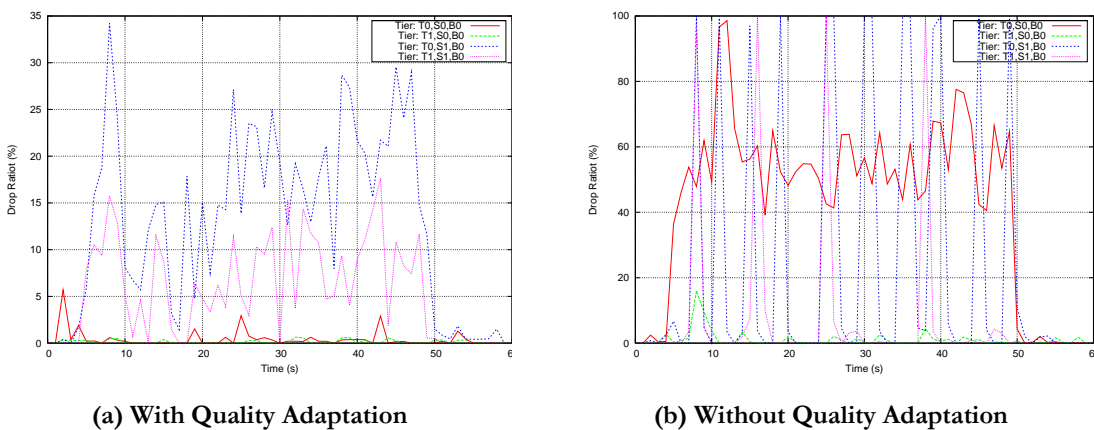


Figure 3-15: Packet Drop Ratio for individual Video Quality Tiers with Overlay Organization

Table 3-5 describes a brief summary of simulation results for the received video throughput of original and individual video tiers in both scenarios. The simulation results show a remarkable

improvement for the received video quality when our proposed adaptation technique was implemented.

	Received Video Quality (%)	Average Received Base Tier (%)	Average Received Enh. Tier 1 (%)	Average Received Enh. Tier 2 (%)	Average Received Enh. Tier 3 (%)
With quality Adaptation	96.2	99.6	99.9	94	86.8
Without Quality Adaptation	77.1	59	98.7	92	75.5

Table 3-5: Summary of Simulation Analysis

We combine all the video tiers to construct the original video scene based on the video packet received during the simulation time for both scenarios. These videos are used to assess the user-perceived QoS. We used two relevant metrics to measure the objective video quality: Peak Signal to Noise Ratio (PSNR) and Structural Similarity Index (SSIM) [123]. Peak signal-to-noise ratio (PSNR) is commonly used for measuring picture quality degradation for the received image quality compared to the original image. The Structural Similarity (SSIM) index is a new method that measures the structural similarity between the original and the received image. We used MSU quality measurement tool for these measurements [124].

The comparison results for PSNR and SSIM of the video generated in both scenarios to the original video are shown in Figure 3-16 and Figure 3-17 respectively. PSNR and SSIM measurements show a noticeable improvement in the received video quality when our proposed quality adaptive streaming mechanism was applied.

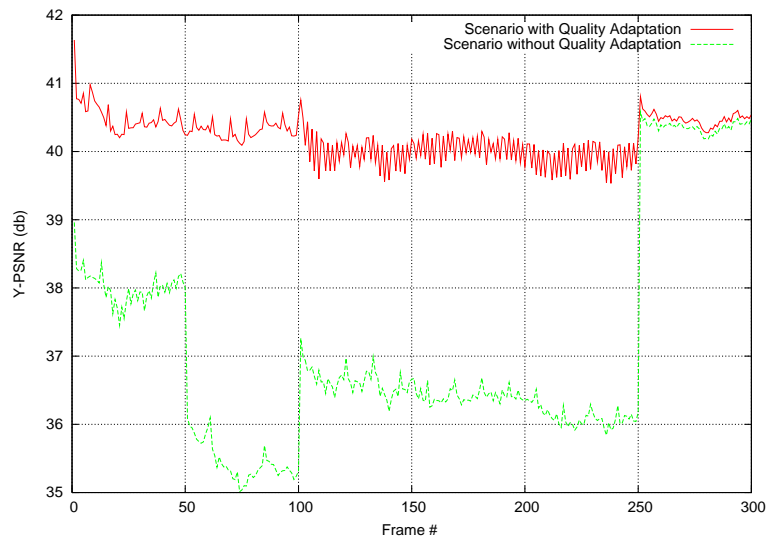


Figure 3-16: PSNR measurements for Received Video Quality

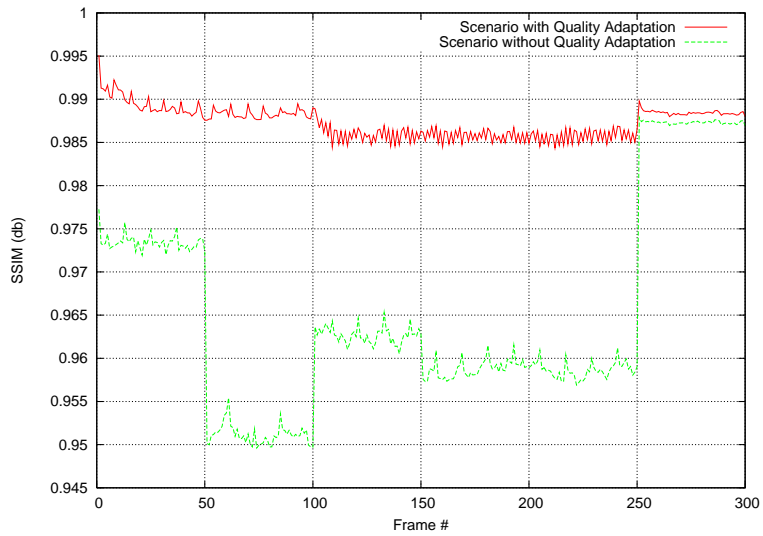


Figure 3-17: SSIM measurements for Received Video Quality

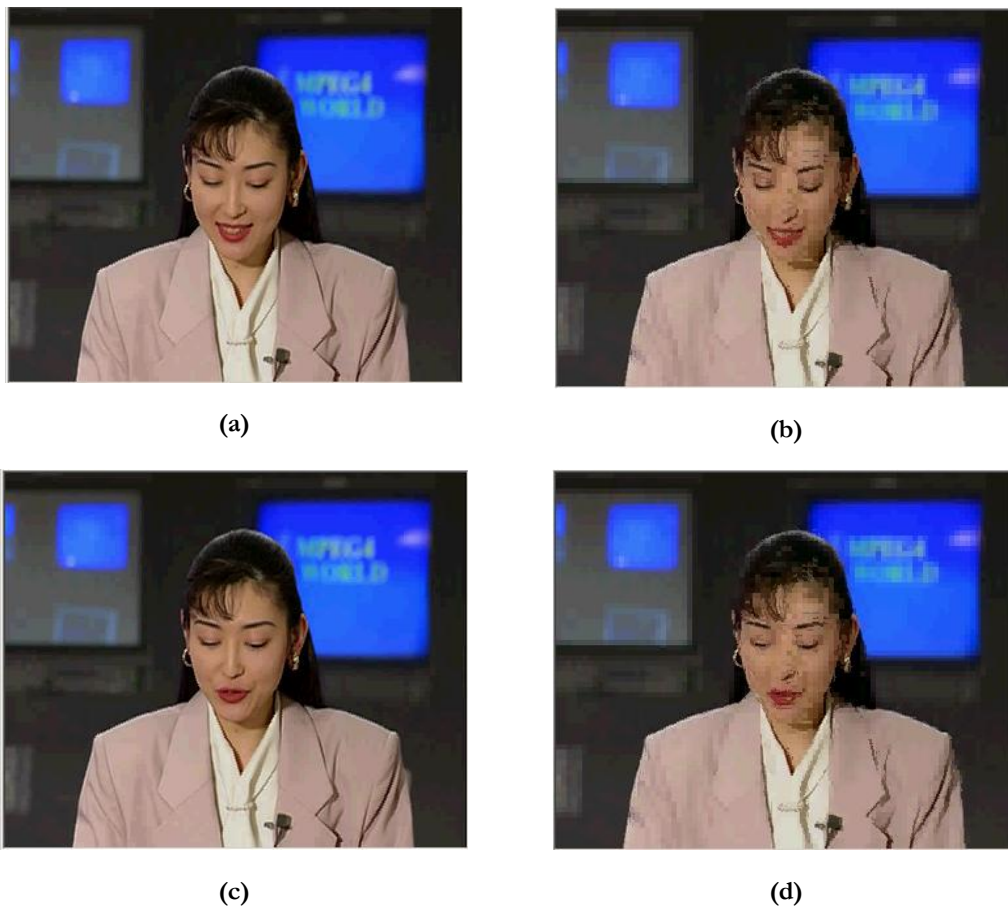


Figure 3-18: Snapshots of Received Video

We have presented some snapshots of the received video quality at receiver end. The received video was regenerated by using “AKIYO” video sequence. Figure 3-18(a) and Figure 3-18(c) represent the received video quality with our quality adaptive streaming mechanism, while Figure 3-18(b) and Figure 3-18(d) represent the received video quality for the same frames without applying

any quality adaptive mechanism. These video snapshots are merely based on the percentage of the received video throughput for each video quality tiers, so they provide just an indicative of the received video quality.

3.5 Conclusion

In this chapter, we presented a quality adaptive video streaming mechanism for P2P networks. The proposed receiver-centric streaming mechanism leverages the characteristics of the Scalable Video Coding (SVC) and P2P networks. The key components of the streaming mechanism include adaptive mechanism (1) organization of sender peers in hybrid overlay networks, (2) efficient sender peers selection and switching mechanism and (3) assignment of video parts to different sender peers. The hybrid overlay networks are constructed on the basis of respective priority of each sender peer towards offered video quality. The overlay organization provides a mechanism to select the best sender peers in $\Theta(1)$ time. The use of the scalable video coding makes the proposed mechanism more promising due to its offered characteristics. The main objective of this mechanism is to ensure smooth video quality delivery with the guarantee of the stable QoS. The best sender peers selection is carried on the basis of QoS measurement of the candidate peers. The peer switching is performed to cope with the dynamic behavior of the sender peer, after active monitoring of the active network links to ensure the smooth streaming quality.

We evaluated the performance of the proposed streaming system by simulations. The obtained results show remarkable improvements in the received video quality using our quality adaptation mechanism. Furthermore, the organization of the sender peers in different overlay networks provides means to utilize the available network resources that reduce the network traffic load on the main servers, and the smooth service delivery can be assured in a more scalable and cost effective manner.

In future, we aim to actually implement this mechanism to observe the real performance especially for the large scale P2P networks. Furthermore, we intend to extend this mechanism for many-to-many streaming scenario and for the mobile P2P streaming.

Chapter 4

4 Packet Video Scheduling Over P2P Networks

4.1 Introduction

We have presented in previous chapters that P2P networks are widely used for the deployment of the on-demand video streaming applications. In many-to-one (multi-source) streaming model, the receiver peer selects many sender peers to receive the video content with QoS guarantee. P2P network is composed of many different heterogeneous terminals having different capabilities as shown in Figure 4-1. In P2P networks, a single sender peer is unable to deliver to whole video content to the receiver peer. We have presented a complete architecture of our adaptive streaming mechanism in Figure 3-2. In this mechanism, a receiver side scheduler plays an important role for the efficient collaboration among sender and receiver peers for the content delivery.

A receiver side scheduler is implemented for the scheduling of received video packets from different sender peers that determines and assigns different video parts to different sender peer. The video packets assignment is based on the priority of the sender peer towards offered QoS and following the traffic specification (TSPEC) model for the video tiers. In multimedia applications, TSPEC parameters are used for the QoS negotiations for the reservation of the resources. All the video packets received from the sender peers are combined in different video buffers. To avoid the buffer underflow problem, the scheduler should consider the buffer states regularly. Peer/stream switching mechanism is implemented to maintain the smooth video delivery to guarantee a stable level of QoS on the basis of the current network conditions and buffer states. The proposed packet scheduling mechanism for the transmitted video streams enables adaptation for the live-video streaming over P2P networks.

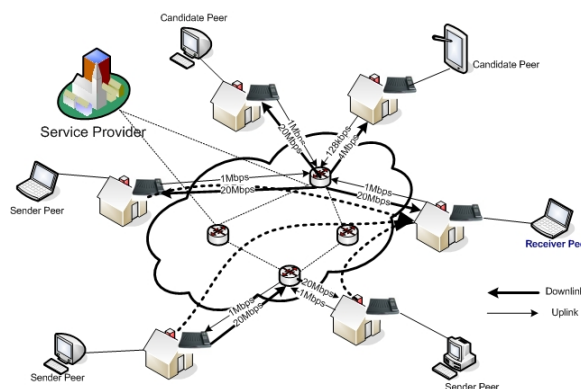


Figure 4-1: Heterogeneous Network Terminals constituting P2P Network

The rest of this chapter is organized as follows. A brief motivation and some of the related work for the packet scheduling is presented in the section 4.2. The proposed video tiers scheduling mechanism along with buffer management are presented in section 4.3. The proposed mechanism is evaluated using network simulations and these results are presented in section 4.4. Finally, section 4.5 concludes the chapter by highlighting the key contribution and by identifying some future perspectives.

4.2 Related Works

There has been an increasing demand to propose solutions for multimedia content delivery over P2P and error prone networks. In this section, we have described some of the related works for the packet scheduling to guarantee QoS.

In [125], the concept of the congestion-distortion optimized (CoDiO) packet scheduling for the pre-encoded video streams have been introduced. This CoDiO algorithm determines the efficient schedule that determines which video packets should be sent, and at which time. The main objective for this scheduling is to achieve the highest decoded video quality while avoiding unnecessary queuing delay over the bottleneck links of the network. The CoDiO scheduler attempts to minimize end-to-end packet delay and to minimize video distortion that plays important role in QoS for the received video streams.

In [126], distortion-optimized packet scheduling techniques are extended to the case of peer-to-peer multicast. The proposed mechanism improves the retransmission scheme that is an extension of the CoDiO mechanism and known as CoDiO P2P. This mechanism gives priority to the most important missing packets in terms of video quality from the parents of the peer. This prioritization algorithm is combined with a distortion optimized retransmission scheduler which operates at the receivers and adapts the video content to request the most important missing packets first. This mechanism implements a scheduler that runs by the receivers which do not know the properties of received video frames in advance.

A recent study [127], have proposed packet scheduling mechanism for the video streaming over QoS-enabled networks. The proposed packet scheduling mechanism is based on the TSPEC parameters and intends to address the problem of static negotiations of the QoS parameters. The mechanism introduced the viable scheduling using a receiver-side buffer management. The packet scheduling strategy is determined on the basis of the priority of the video layers. Authors proposed their solution with a new multi-track hinting method that is an extension of the MP4 hint-tracks in order to provide real-time adaptation for multimedia streams to multiple quality levels.

We present video scheduling mechanism over P2P networks where received video tiers are assigned to different sender peers on the basis of their offered QoS. A receiver side scheduler is implemented to assign different video parts to sender peers on the basis of their relative schedule and playback deadlines. The most important part of video (base tier of SVC) is assigned to the sender peer offering highest priority. The received video packets from multiple sender peers are collected in receiver-side buffers. The buffers are divided into two parts, (1) reliance part and (2) active part. The reliance part of the buffer keeps the received video packets that can be used for decoding and ensures that all the corresponding video streams are available for respective video

tier. This part of the buffer can be seen as a guard part. If it starts to decrease, remedial actions have to be taken such as peer or stream switching. The active part is designated to receive the video packets continuously for the video streams decoding phase.

4.3 Packet Scheduling & Buffer Management

The efficient scheduling of video packets becomes more important under the unpredictable and time-varying nature of the best-effort Internet. A receiver side scheduler is the key element of our proposed system that takes the input information related to the playback time of each video packet in each tier and the current receiver buffer state of each tier. On the basis of this received information, scheduler assigns different video tiers to different sender peers and uses the Traffic Specification (TSPEC) model [111] which is generally applied for QoS negotiation. These TSPEC parameters determine the peak and average requirements for the streaming applications. The signaling protocol for TSPEC negotiations is out of scope of this chapter. Under this model, the TSPEC parameters are negotiated among the sender and receiver peers before starting the streaming mechanism. TSPEC includes parameters such as peak data rate (PDR), mean data rate, maximum burst size, worst case delay, average packet size, maximum packet size, maximum packet error rate, and maximum worst-case delay as described in Table 4-1. The key elements of the proposed scheduling mechanism are presented in the following sub-sections.

4.3.1 Data Flow Model

The data flow model is an important aspect of the video streaming mechanism. A scheduler is implemented on the receiver side for the relative scheduling of video quality tiers according to their priorities, transmission time and actual playback time. The scheduler determines two sets of parameters for each video packet; (1) the transmission of video packets from sender peers on the basis of relative schedule and priority, (2) the admission of the received packet at the receiver side with respect to respective playback deadlines. These set of parameters are used to determine the viability of video packets before putting them in the receiver side buffer, i.e. if certain video packets are received after the respective playback deadline, these packets will be discarded. We represent these set of parameters by $\beta_k^{schedule}$ and $\beta_k^{receive}$, where $\beta_k^{schedule} = \{s_k, P_k^{playback}, P_k^{transmit}\}$ and $\beta_k^{receive} = \{s_k, P_k^{playback}, P_k^{receive}\}$. Here, s_k represents the packet size, $P_k^{playback}$, $P_k^{transmit}$, and $P_k^{receive}$ represents relative playback time, transmission time, and receiving time of packet 'k' at the receiver end. The scheduler ensures that each video packet arrives at the receiver side buffer on time to avoid buffers under flow problem. Toward this end, a token bucket model is applied to ease packet scheduling of incoming traffic (video streams) from different sender peers using TSPEC.

Acronym	TSPEC Parameter
PDR	Peak Data Rate of video layer
ADR	Average Data Rate of video layer
CDR	Committed Data Rate for allocated bandwidth
PBS	Peak Burst Size of video packets
X_{avg}	Average Packet Size of video packets
X_{max}	Maximum Packet Size of video packets

Table 4-1: TSPEC Parameters for Video Streaming

All the received video packets from multiple sender peers are kept into different receiver-side designated buffers before their actual decoding. A brief illustration for the buffer management is present in next section.

4.3.2 Scheduling Mechanism

Here, we described the scheduling mechanism that show, how video tiers and their respected packets are assigned to different sender peers. All the sender peers are organized on the basis of their offered QoS (highest priority) according to the overlay organization that we presented in chapter 3. The best one is assigned high quality layer “Base tier”, and the enhancements tiers are assigned to the sender peers in descending orders of offered bandwidth. The algorithm’s steps for the video packets selection from sender peers and their decoding phases are described below:

To provide a smooth video delivery, a receiver side scheduler determines the relative schedules for each video description according to their priorities that include transmission time and actual playback time. We consider two sets of parameters to apply scheduling mechanism for each video tier. First, the transmission of video packet from the sender peer is done according to relative scheduling time and priority. Second, the video packet should arrive at receiver-end before playback deadline (i.e., maximum delay bound).

These set of parameters present in section 4.3.1 determine the viability of the video packets before putting them in the receiver side buffer. If certain video packets are received after the respective playback deadline, these packets along-with the corresponding video packets of upper layers are discarded. This information is used for the transmission schedule of video packets and to determine the viability of received video packets at receiver side before putting these video packets into respective buffers.

The incorporation of the video scheduling mechanism allows controlling the admission of the new traffic flows over network links such that each packet sent by a particular peer must not exceed the peak data rate. Hence each packet sent must obey to this rule: $\forall i: \frac{S_i}{t_i} \leq PDR$, where t_i represents a time interval between two packets.

We can derive the first constraint (Eq. 4-1) that presents the traffic test conformance (i.e. maximum throughput constraint):

$$\sum_{k=1}^m s_k \leq (\sum_{k=1}^m P_k^{transmit}) PDR \quad \text{Eq. 4-1}$$

Here, we are sure that the amount of traffic sent (i.e. $\sum s_k$) does not exceed the PDR.

The second constraint (Eq. 4-2) is related to the draining of buffer at the receiver side, since we should be sure that those packets still arrive at the destination.

$$0 \leq \sum_{k=1}^m s_k - ADR \sum_{k=1}^m P_k^{transmit} \leq PBS \quad \text{Eq. 4-2}$$

This constraint shows that the buffer should not be overflowed or underflowed.

The third constraint is related to the end-to-end delay. Eq. 4-3 represents E2E delay of the arrival video packets from sender to receiver. This constraint is more important as the real-time applications require minimum delay and play a major role in the overall QoS. Packet loss occurs when the deadline delay is passed.

$$0 \leq \sum_{k=1}^m P_k^{receive} - \sum_{k=1}^m P_k^{transmit} \leq D_{\max} \quad \text{Eq. 4-3}$$

D_{\max} can be determined on the basis of estimated maximum RTT (the priority of sender peers) that can be tolerated between the sender and the receiver node. However, in streaming applications, we may tolerate some delay for the draining buffer (10s-15s). In our proposal, D_{\max} is inspired from the retransmission timer technique used in TCP.

$$D_{\max} = \min[U_{bound}, \max[L_{bound}, (\alpha * SRTT)]] \quad \text{Eq. 4-4}$$

In Eq. 4-4, U_{bound} represents the maximum size the receiver side buffer, and L_{bound} represents the minimum bound for buffer that can be 50% of the U_{bound} , α is factor of variance of delay, and SRTT corresponds to the smooth RTT estimates by an exponential average that gives more weight to the recent RTT estimates compared to the older estimates.

4.3.3 Buffer Management

On the receiver side, separate buffers are assigned to receive different video packets from each sender peers. All the received video tiers are combined before actual decoding. The reception of more and more quality tiers enhances the overall video quality.

Another receiver side global buffer is allocated to receive the packets of base quality tier and is represented by three parameters (1) Total Buffer Size, (2) Active part, which is continuously receiving the base quality video packets from different sender peers, and (3) Reliance Part, which holds the base quality video packets ready to decode. The reliance part of the buffer allows us to receive a reasonable amount of video packets that can ensure smooth video streaming. We name this part as reliance part due to the fact that before starting the decoding of the video packets, we

are sure that at-least minimum level of video quality (i.e. Base tier) can be assured after decoding the received packets. This classification of receiver side buffer is depicted in Figure 4-2.

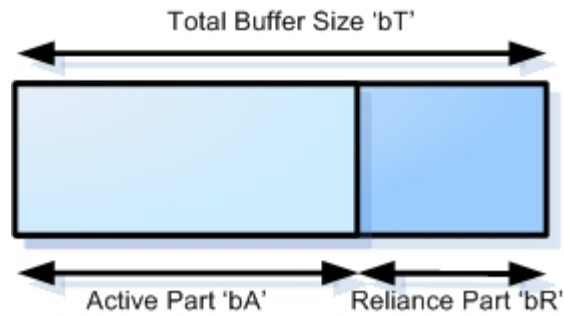


Figure 4-2: Buffer Classification

Each packet in the “Reliance Part” of the buffer is distinguished according to its playback deadline. At the time of decoding, packets from each tier are combined to construct the original video stream with maximum quality. A lost packet can be re-requested using ARQ (Automatic Repeat Request), if there is reasonable playback deadline as received packets are recognized according to their respective video quality layer and playback deadlines. On the other hand, if it is not possible to re-request the lost packet, all the subsequent upper layers packets will also be dropped because they cannot be decoded without its reference base tier.

We represent the receiver side buffer using the parameters ‘bT’, ‘bA’ and ‘bR’, where ‘bT’, ‘bA’ and ‘bR’ represent the total buffer size, the active buffer size and the reliance buffer size respectively.

The received packets are extracted from the buffers according to their respective playback deadlines. Receiver peer continuously receives video packets from different sender peer. The new received packets are kept in active part of the buffer; on the other hand, decoding is performed for the video packets present in the reliance part of buffer. To ensure the provision of at-least the base tier quality, it is necessary that: $bR > 0$ (i.e. there should be some video packets in the reliance part of the buffer to avoid the worst case). Furthermore, we ensure that at-least video packets for base tier must be received in ‘Reliance Part’ of buffer before decoding process.

Our aim in this study is to ensure that at-least base tier of media content remains available before starting the decoding phase. A brief description for the selection and the decoding phases of the received video packets on the basis of packet scheduling mechanism and buffer management are described as follow:

Selection Phase:

- Select sender peers from set of candidate peers using hybrid overlay organization.
- Scheduler along with sender peer negotiate TSPEC (Traffic Specification) parameters for each sender peer based on the estimated measurement and on the basis of relative $\beta_k^{schedule}$ and $\beta_k^{receive}$.
- Receive video packets and put them in respective buffers.
- If packet in the receiver buffer < the threshold: perform stream or peer switching

Decoding Phase:

- Fetch packets of all the corresponding video tiers from the respective buffers,
 - If any packet of base tier is not present in reliance part, fetch it from the active part,
 - If the corresponding video packet is not present in active part, drop all the video packet corresponding to this base tier,
 - If reliance part decreases prepare peer or stream switching mechanism.
 - continue.

These tiers and the packet assignments work in a round robin fashion under the respective priority of tiers. Whenever there is provision of admitting new peer, lower level tier offering high quality is assigned first.

4.3.4 Peer & Stream Switching

P2P networks are dynamic in nature and network condition may change with the course of time during the streaming mechanism. In this scenario, peer & streaming switching mechanism is incorporated to ensure the smooth video delivery from all the sender peers. Peer switching mechanism is incorporated when a sender peer leaves the P2P network, whereas stream switching mechanism is applied when a sender peer is unable to offer the QoS for the received video content as the expected arrival of the data violates the TSPEC model and leads to the draining the reliance part. In the case when a sender peer leaves the system, the scheduler assigns the corresponding video stream to some alternate sender peer present in the overlay network. The scheduler keeps track of all receiver-side buffers for the received video packet, if any of the senders peer is unable to contribute for the video content, scheduler re-assigns the video streams to these peers. The detailed peer & stream switching mechanism was described in section 3.3.4.2 for Quality adaptation.

4.3.5 Packet Ordering for Quality Adaptation

In this subsection, we describe packet's ordering that determine the manner in which video packets are filled in and drained out from the buffers during the streaming session [51]. In the real-time streaming applications, an intelligent packet ordering scheme plays a significant role which can be beneficial to improve the received video quality by utilizing the available resource efficiently. The packet ordering highly depends on the receiver preferences. It depends on either the receiver wants the video with high SNR quality, temporal resolutions for varying frame rates, or spatial resolution for the different resolutions, etc. Video packets can be ordered depending on the application type and end-user preference by adopting horizontally, vertically, or diagonally packet ordering as shown in Figure 4-3.

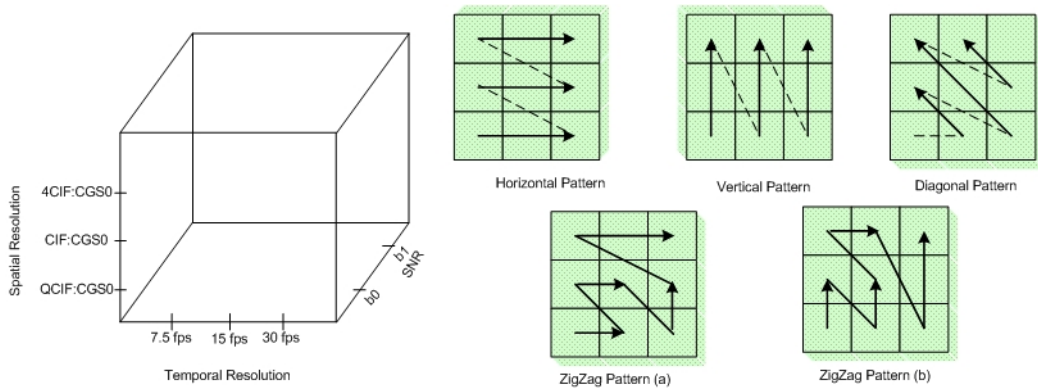


Figure 4-3: Video Tier Assignment Patterns

Horizontal and Vertical ordering pattern is more suitable for the cases when receiver intends to receive video quality based on a single parameter, i.e. temporal, spatial, or SNR is given more weight for received video. The selection of tiers (T0, S0, B0), (T1, S0, B0), and (T2, S0, B0) provides good temporal quality, while tiers (T0, S0, B0), (T0, S1, B0), and (T0, S2, B0) can be selected for Spatial resolution. In the same way, peers can be selected for SNR quality.

ZigZag and Diagonal packets ordering provides mechanism to select different sender peers to receive video quality based on multiple parameters (temporal, spatial, SNR) offering different characteristic to the video sequence. These packet ordering patterns receive the video streams with respect to the base stream according to SVC decoding model. In general, video content are adapted with respect to three parameters, temporal, spatial and SNR resolutions. The selection of these parameters depends upon the network, application, and user's terminal characteristics and preferences. In our study, we propose diagonal ordering to fill-in and drain the video packets out of the receiver side buffer in order to enhance the video quality in terms of SNR, spatial and temporal scalability. In ZigZag packet ordering, receiver peer receives the base video tier (T0, S0, B0) in the beginning, whereas the enhancements video tiers are received with respect to the base tier. All the enhancement tiers are decodable with respect to their lower layers (base tier, etc), so extra care is required to receive the base quality tier.

The ZigZag and Diagonal packet ordering implies that packets of lower layers are requested well before actual playback. Such strategies are helpful to assure at-least base tier quality throughout the streaming session. In contrast to ZigZag and Diagonal packet ordering, horizontal and vertical packet ordering schemes are more effective when receive intends to receive smooth quality in one dimension whereas, in ZigZag and diagonal packet ordering pattern, the received video quality improves incrementally in multiple dimensions.

In general, we select an independent sender peer for contributing different part of video. However, it is possible that a single sender peer is unable to contribute for the corresponding video tier. In this case, multiple sender peers are selected for the same video tier. Here, we have shown the video distribution at packet level and different packets are assigned to two sender peers. Furthermore, additional sender peers can be added to receive the video tiers to achieve the desired quality depending on the network conditions. In Figure 4-4, two different sender peers are selected to contribute for a single video. The target video quality is (T1, S1, B0) focusing for the temporal and spatial resolution. A receiver peer must receive all the corresponding lower video tiers to

achieve the required video quality, in this case receiver peer must receive the quality tiers (T0, S0, B0), (T1, S0, B0), (T0, S1, B0), and (T1, S1, B0).

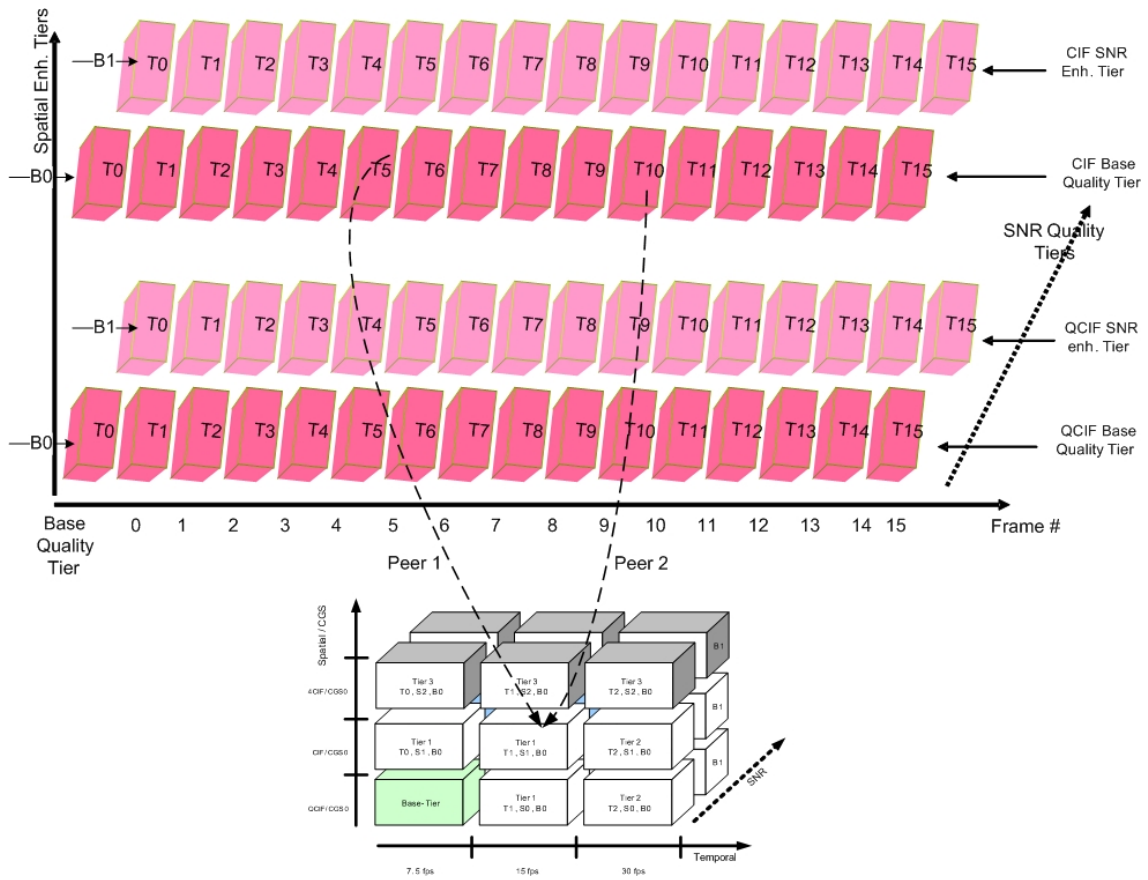


Figure 4-4: Scalable Video Coding: Bit-Stream Dependency

4.4 Performance Evaluation

This section describes the simulation results of the proposed video scheduling and buffer management for the live video streaming mechanism. The simulations for the proposed streaming mechanism were performed using the ns-2 simulator [112].

4.4.1 Performance Metrics and Evaluation Model

We consider new Gnutella like P2P topology for the simulations, where end nodes (peers) are attached to a super node and we organize different groups of peers on the basis of proposed hybrid overlay scheme. We implement a receiver-side scheduler that determines that video tiers assigned to different sender peers. We integrate this scheduling mechanism along with adaptive streaming mechanism based on the selection of sender peers from multiple overlay networks and peers/stream switching as described in chapter 3. The underlying video encoding scheme is SVC and the most important video tier (base tier) is assigned to the best sender peer offering highest QoS. For the test cases, we generated 4 different quality tiers using MPEG-4 trace files [113]. The video sequence is divided into one base tier and three enhancement tiers contributing for the temporal, spatial and SNR characteristics.

Receiver peer intends to receive the video quality of CIF with 15 fps. The packet ordering for the received video is based on ZigZag pattern. Each sending peer contributes by different quality tiers of the original video file, so that it can be used to reconstruct a video file with the best quality at the receiver node. The scheduling mechanism is incorporated on the basis of TSPEC for the SVC video streams following the different constraints as presented in Eq. 4-1 to Eq. 4-4.

The overall video throughput of the different video quality tiers is shown in Figure 4-5. We noticed that no source peer is providing 100% throughput of the original video but if a receiver peer receives all the 4 tiers from different sender peers, it is possible to reconstruct the original media file with 100% quality with the selected characteristics of scalable video coding scheme. The simulation time was 60 seconds and the presented results are the average results of ten runs of these simulations. We attached two “CBR sources” to overcharge the network. “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. Both sources inject constant throughput of 512 Kbps with 512 Bytes UDP packet lengths. The proposed mechanism was simulated for the two scenarios to observe the performance.

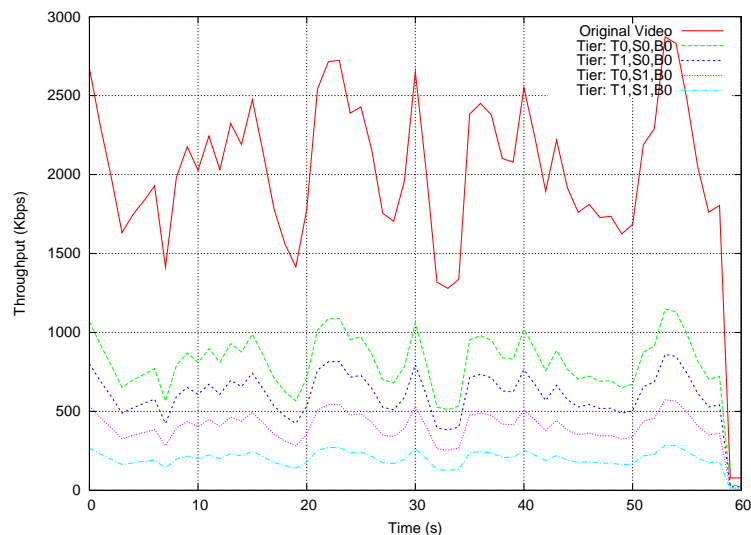


Figure 4-5: Video Quality Tiers Throughput

- **Scenario without Quality Adaptation:** Simulation without applying any packet scheduling mechanism. In this case, P2P system works as in downloading mode and the different video streams are assigned to different sender peers randomly.

- **Scenario with the proposed Quality Adaptation:** Simulations with quality adaptation mechanism is performed where a receiver-side scheduler assigns different parts of video streams to the different sender peers. The peer/stream switching was incorporated on the current buffer states for the received video streams.

4.4.2 Simulation Analysis

Figure 4-6 shows the received video throughput at receiver peer in both scenarios i.e. with our proposed quality adaptation streaming mechanism and without applying quality adaptation mechanism along with the expected video quality when using the four different quality tiers. We observed that the quality adaptation mechanism improves the received throughput compared to the scenario without quality adaptation. In quality-adaptive scenario the sender peers were assigned different video stream with respect to TSPEC. We have noticed few packets drop even with quality adaptation mechanism. These packets drops are caused due to the heavy stress on the network created by CBR/UDP traffic which is presented in Figure 4-7. We noticed that packet drop ratio is much lesser in the scenario with quality adaptation compared to scenario without adaptation. The enhanced throughput and lower packets drops results into the overall improvement in QoS.

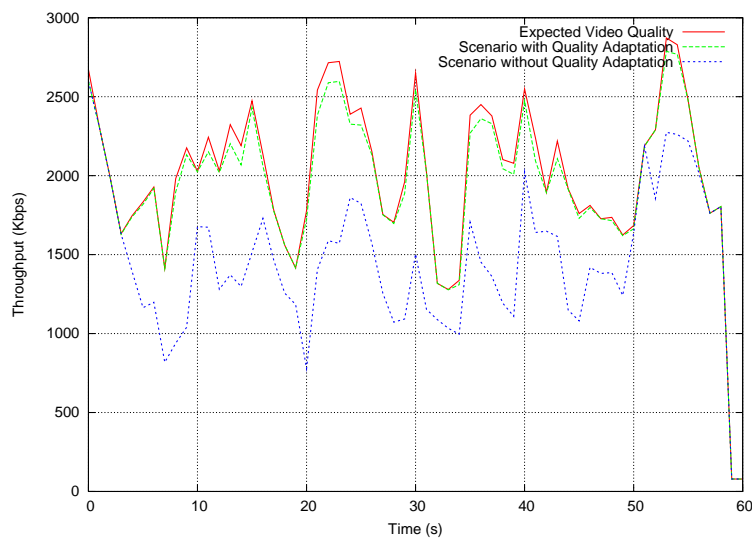


Figure 4-6: Received Video Throughput

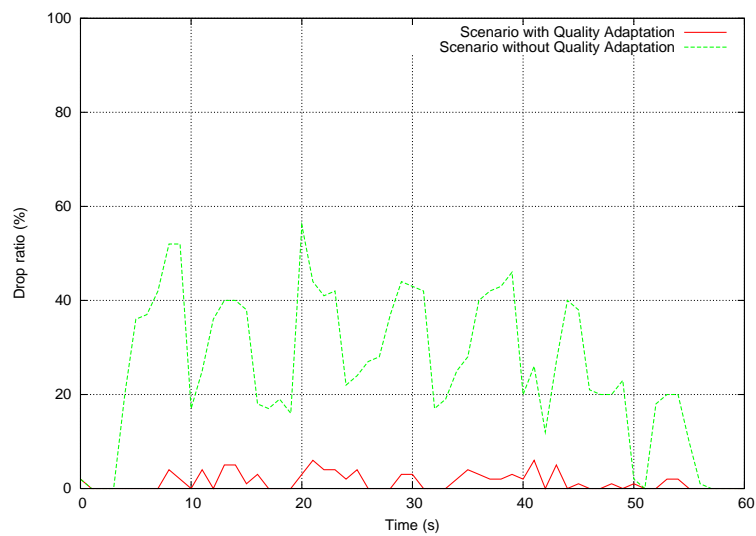
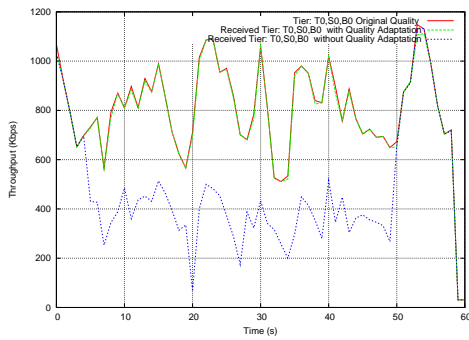
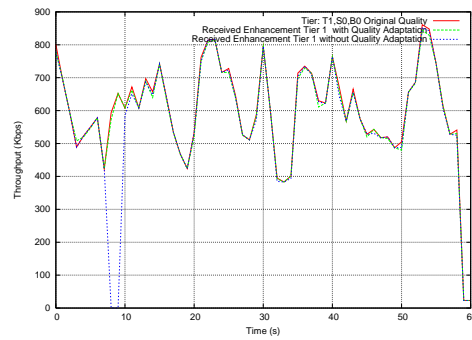


Figure 4-7: Packets Drop Ratio

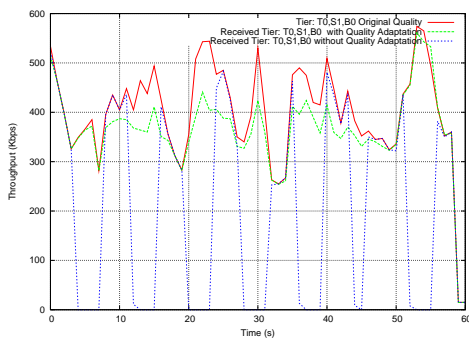
Figure 4-8(a), Figure 4-8(b), Figure 4-8(c), and Figure 4-8(d) describe the comparison between received and original base tiers, enhancement tier 1, enhanced tier 2, and enhanced tier 3 respectively at different interval of times during the simulations.



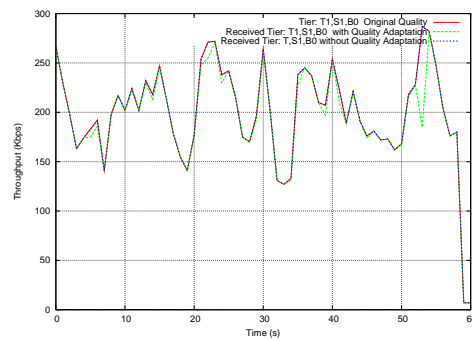
(a) Base Tier



(b) Enhancement Tier 1



(b) Enhancement Tier 3



(b) Enhancement Tier 4

Figure 4-8: Received Video Throughputs

The overall received 1-way packets delay (from sender peer to receiver peer) for both cases is presented in Figure 4-9. The packet transmission delay has been significantly improved when quality adaptation mechanism is applied that is very promising for the real-time streaming applications where a receiver intends to receive whole video content more quickly. This improvement in the transmission delay is due to the intelligent selection of sender peers that is facilitated by overlay organization and the effective video packets ordering by the scheduler. This lower delay plays a more role in the smooth video delivery and highly desirable for the real-time applications.

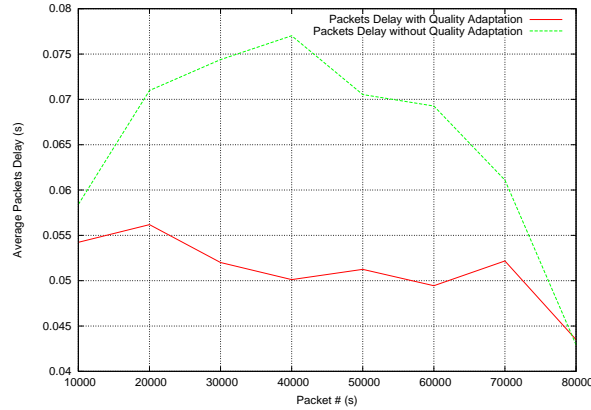


Figure 4-9: One-way Packet Delay

By analyzing the obtained results, we can see clearly that our proposed mechanism performs a smooth video delivery with higher quality, lower loss and delay. A summary of these results is given in Table 4-2. We have noticed that receiver peer receives almost 100% of the original enhancement tier 2. However, this video tier is decodable with respect to its base tier, while the received base tier for the corresponding enhancement tier is only 60%. In this case, this enhancement tier cannot contribute to enhance the overall QoS and the video packets for enhancement tier are discarded.

	Received Video Quality (%)	Average Received Base Tier (%)	Average Received Enh. Tier 1 (%)	Average Received Enh. Tier 2 (%)	Average Received Enh. Tier 3 (%)	Average Packet Delay (ms)
With quality Adaptation	98	99	99	99	93	45
Without Quality Adaptation	76	60	96	100	57	60

Table 4-2: A Summary of Simulation Analysis

We reconstructed the expected video file using the “AKIYO” video sequence on the basis of the received video quality at the receiver end. The generated “AKIYO” videos are used to evaluate the PSNR and SSIM objective measurements. We used MSU quality measurement tool for these measurements [124]. Peak signal-to-noise ratio (PSNR) is commonly used for measuring picture quality degradation. The Structural SIMilarity (SSIM) index [123] is a new method for measuring the similarity between two images. The comparison results for PSNR and SSIM of the videos generated in both scenarios to the original video are shown in Figure 4-10 and Figure 4-11 respectively. PSNR and SSIM measurements shows a noticeable improvement in the received video quality when our proposed quality adaptive streaming mechanism is applied.

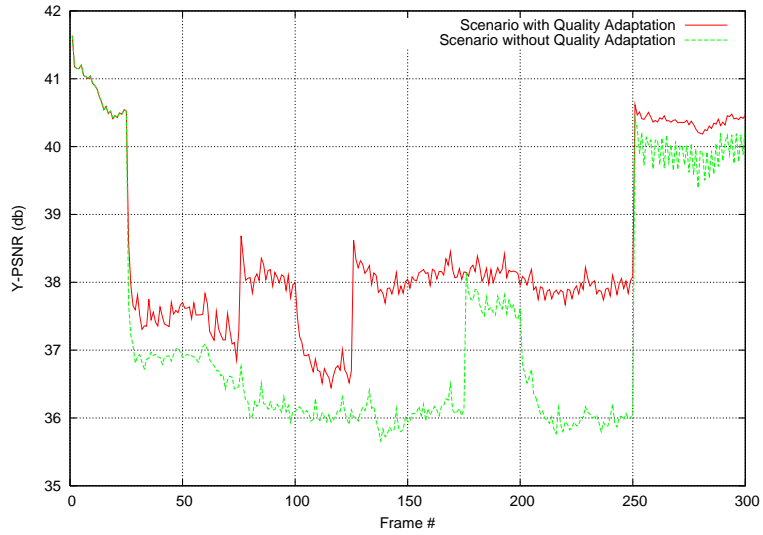


Figure 4-10: PSNR Measurements

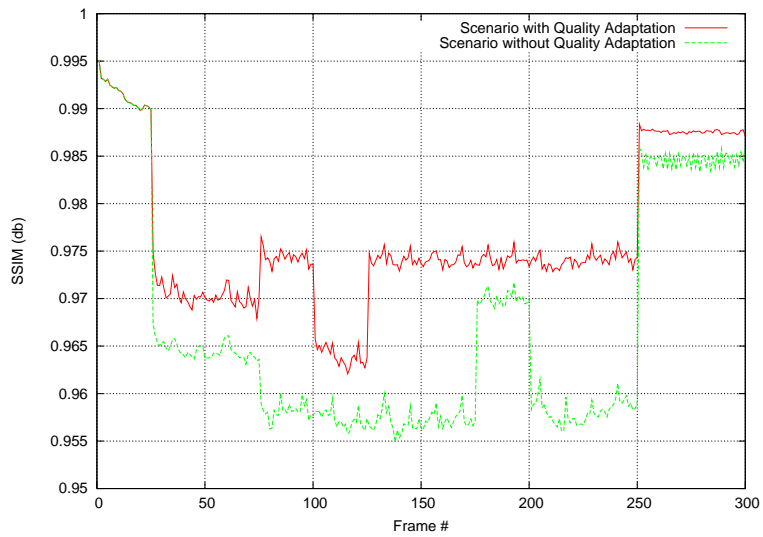


Figure 4-11: SSIM Measurements

We regenerated received video by using “AKIYO” video sequence on the basis of the received video through in both simulation scenarios. Figure 4-12(a) represent the received video quality with our quality adaptive streaming mechanism integrated with packet scheduling mechanism and Figure 4-12(b) snapshots for the received video quality for the same frames without applying packet scheduling mechanism.

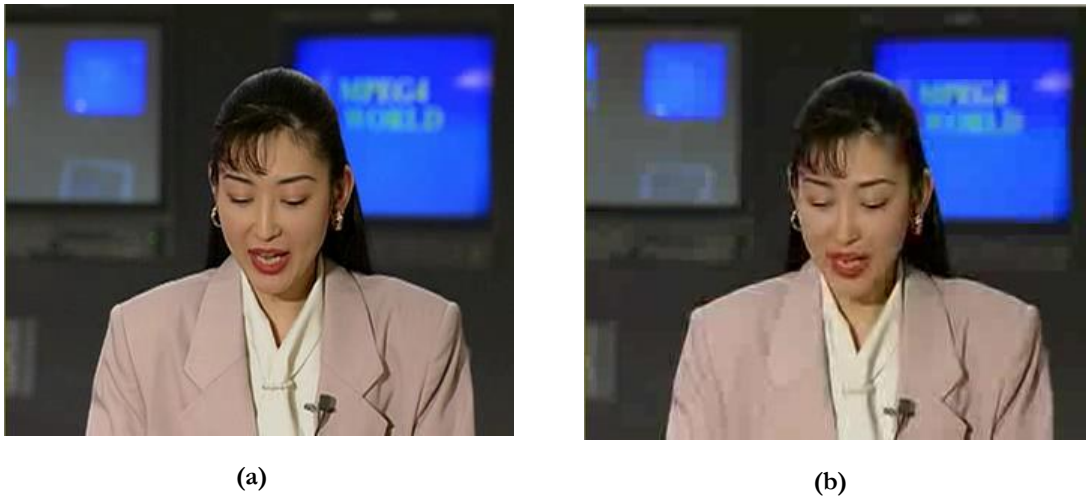


Figure 4-12: Snapshots of Received Video

A noticeable improvement in the received video throughput, the reduced packet drop ratio, and lowered packets delay in the scenario with quality adaptation based on the scheduling mechanism incorporated with organization of sender in overlay networks enhance the overall QoS for multimedia streaming application over P2P networks.

4.5 Conclusion

In this chapter, we presented a receiver-side scheduling mechanism that is used to assign different video streams to different sender peers. The scheduler determines the video schedules on the basis of priority of sender peer to offer the QoS in streaming mechanism and is facilitated by different receiver side buffers. These buffers are designated to receive the video streams before their actual decoding. The proposed mechanism is evaluated using simulations and a noticeable improvement in the received video quality is observed under the proposed mechanism. In fact, a significant improvement of the received throughput especially for the important video quality tiers, lower packets drop ratio for the important quality tiers, and a considerable improvement in the received packets delay are observed.

For the future perspectives, we aim to implement this mechanism to see the exact behavior in real application. Furthermore, we also plan to evaluate the complexity added by the scheduling mechanism and their impact for the live and on demand video streaming over P2P networks

Chapter 5

5 SP-Driven P2P Networks: A Framework for IPTV Service Delivery with QoS Provisioning

5.1 Introduction

Since the last few years, Internet has witnessed a massive growth in the deployment of real-time applications and multimedia services. Recent measurement studies [128] have shown that traffic generated by P2P applications has started to dominate the bandwidth consumption over Internet access links and a significant part of the current global Internet traffic today is generated by P2P applications. These P2P applications often deal with large amounts of content delivery using direct P2P connections between sender and receiver peers. P2P applications usually have little knowledge of the underlying physical network topology, and choose the content delivery paths based on self estimated measurements and statistics [129]. As a result, this network-oblivious peering strategy may cause traffic to scatter for many P2P applications [130]. Hence, the actual media content unnecessarily traverse multiple links within the network to reach the targeted client. This delivery process imposes higher load on network backbone and such traffic pattern may cause imbalance between peers, leading to potential violation of peering agreements between network providers.

Today's, Internet is undergoing through a massive shift from being a simple monolithic data service network to a ubiquitous multi-service network where different stakeholders co-operate for offering value-added services and applications to content consumers (CCs). Content providers (CPs), service providers (SPs), and network providers (NPs) have different roles and requirements for providing services and applications which may dynamically be created and managed. In this massive growth of P2P, the abovementioned stakeholders (i.e., SP, CP, NP) involved in the content distribution are missing opportunities for future revenue growth. The prevailing use of P2P applications is a major driving force for the adoption of broadband access, which is a significant source of revenue. A successful strategy to manage P2P traffic must address both the stakeholder perspective of costs and the content consumer perspective of quality of service (QoS). The problem is that SPs have no rules over the structure of the P2P networks and therefore they can neither control nor manage the associated traffic flows. So far, P2P users just want to retrieve their desired content quickly, with good performance. Nevertheless, P2P system has no knowledge on the underlying network structure and it either has to measure the performance itself or to build its

overlay topology more or less agnostic of the underlay. This situation is frustrating to both SPs as well as P2P users. The cooperation between SPs and P2P users not only improves the network utilization but also allows the SPs to provide better services to its customer's [129]. Moreover, such cooperation can be helpful to address the issue of application-layer traffic optimization problem [130]. We believe that service provider (SP)-driven P2P system will constitute the next generation P2P system that is willing to provide better QoS performance to CCs whilst enhancing revenue growth for different actors involved in the delivery chain. Explicit communication for cooperative control between P2P and providers has been recently the subject of a working group called P4P [131]. P4P designs a set of business practices and integrated network topology awareness models to optimize network resources and to enable P2P-based content delivery.

P2P systems organize the participating peers in virtual overlay networks in order to support content/service delivery effectively. The overlay networks are constructed virtually on top of physical networks to improve the current Internet architecture regardless of the access and core networks topologies. It provides a scalable solution in the content sharing/distribution to support increasing number of requests/users in more transparent way. Furthermore, it does not require many investments for deploying the infrastructure. Initially, P2P systems were designed for content distribution but nowadays P2P systems are widely used for many more applications due to its promising features. Today, the most popular applications based on P2P architecture include: video conferencing, signaling for IP telephony, P2P gaming, video streaming/video on demand, P2P-IPTV, etc. IPTV is expected to be the fast growing industry in coming years and according to some industry estimates there will be around 80 millions IPTV worldwide subscribers by 2011 [132].

The promising characteristics of P2P network make it more suitable for the deployment of the IPTV service. In the service delivery process, different stakeholders cooperate to offer value-added services to end-customer/content consumer (CCs). Content providers (CPs), service providers (SPs), and network providers (NPs) are the main stakeholders that have different roles, requirements and interests for providing such services. Regardless to the high popularity of P2P for wide adoption of service delivery it poses certain problems and eventually not acceptable for all the stakeholders participating in the service delivery process.

In this P2P-based service delivery process, network inefficiency is considered as one of the major problems caused by the P2P systems. P2P applications show greedy behavior and tend to utilize the maximum available bandwidth and other resources that ultimately affect the performance of other applications. This problem leads to extra network resource usage (e.g., using bandwidth of more links) and hence increases the network operational cost that is not acceptable for the NPs and SPs. On one hand, such inefficient utilization of bandwidth and inefficient interactions with other applications degrade the overall Quality-of-Service (QoS) for different applications. On the other hand, free riding is another problem with P2P system where many participating peers intend to join P2P network to utilize the available resources without sharing their resources with others. P2P applications are popular and considered good to content owners, however from the network's perspective, P2P is a problem. Free-riding is prevalent in file sharing networks [133]. A study for the Gnutella architecture [134] shows that around 66% peers do not share even a single byte and joins the Gnutella networks just for downloading the video content from other peers. Whereas, 87% video content/files were shared by only 10% peers and around 20% peers share 98% files.

These statistics show a serious trend and greedy behavior of peers joining the P2P network without sharing their content with each others.

SPs guide applications to efficiently utilize the network resources but P2P applications implement their own policies and tend to utilize maximum available resources. Most of these problems arise due to lack of any central controlled entity that can manage P2P architecture by controlling the utilization of available resources. P2P users are determined to retrieve their desired media content more quickly although they do not have any knowledge on the underlying networking structure. SPs also have no control over the structure of the P2P networks and therefore they can neither control nor manage the associated traffic flows. These problems can be resolved by incorporating effective communication among network providers, service providers and P2Ps applications.

Thus, in this chapter, we present a brief description of SP-driven P2P framework for the IPTV service delivery with the explicit communication among SPs and P2P applications. We also present a QoS provisioning mechanism for real-time media streaming over SP-driven P2P networks using admission control. The SPs have a global view of the available bandwidth and maintain a traffic forecast matrix that provides the information regarding allocated and available bandwidth over certain network links. SPs further use this information to establish QoS-enabled paths between peers participating in a session for actual content delivery. The admission control mechanism results into efficient utilizing of resources by preventing the unauthorized peers from accessing the SP-driven P2P overlays.

5.2 SP-Driven P2P Networks

Our SP-driven P2P mechanism allows better cooperation between P2P and service provider (SP) through explicit communications. SP updates and shares the current network conditions with P2P applications and allocates end-to-end network resources for efficient content delivery. SP organizes all the participating peers in virtual overlay networks. All the peers constituting P2P networks have multiple roles and may form coordinated groups/communities. The participating peers in overlay networks are organized on physical characteristics like media awareness (offered media content), locality-awareness (organization of peers having same characteristics closer to each other) and QoS-awareness (offered video quality). Initially, in SP-driven mechanism, we organize peers using the hybrid overlay organization that is composed of locality-awareness and context-awareness. All the peers offering same video quality are organized closer to each other and with respect to their virtual distances from receiver peer. Generally, overlay networks are intended to determine the routing over the service delivery path; however in our SP-driven P2P network such overlay organization is used to assist the receiver peer for choosing best sender peers. SP is responsible for the creation, maintenance, and destruction of these overlay networks. The overall mechanism is SP centric though the streaming mechanism for the IPTV service delivery is receiver centric, where each receiver peer selects multiple sender peers from the overlay networks. The overlay networks maintenance is essential in the dynamic behavior of P2P networks where peers enter and/or leave the networks more often and without any prior notification. In SP-driven P2P network, SP performs continuous network probing to maintain the overlay networks and thus to

ensure the smooth content delivery. The overall architecture of SP-driven P2P framework is present in Figure 5-1. Service provider (SP) plays a vital role in this framework to support the service delivery. SP has to manage different entities to support the service delivery mechanism.

The main objective behind designing the SP-Driven P2P framework is to enable better SPs and P2P coordination where SPs and P2P jointly decide the peering strategy that eventually enhances the overall QoS for the P2P applications.

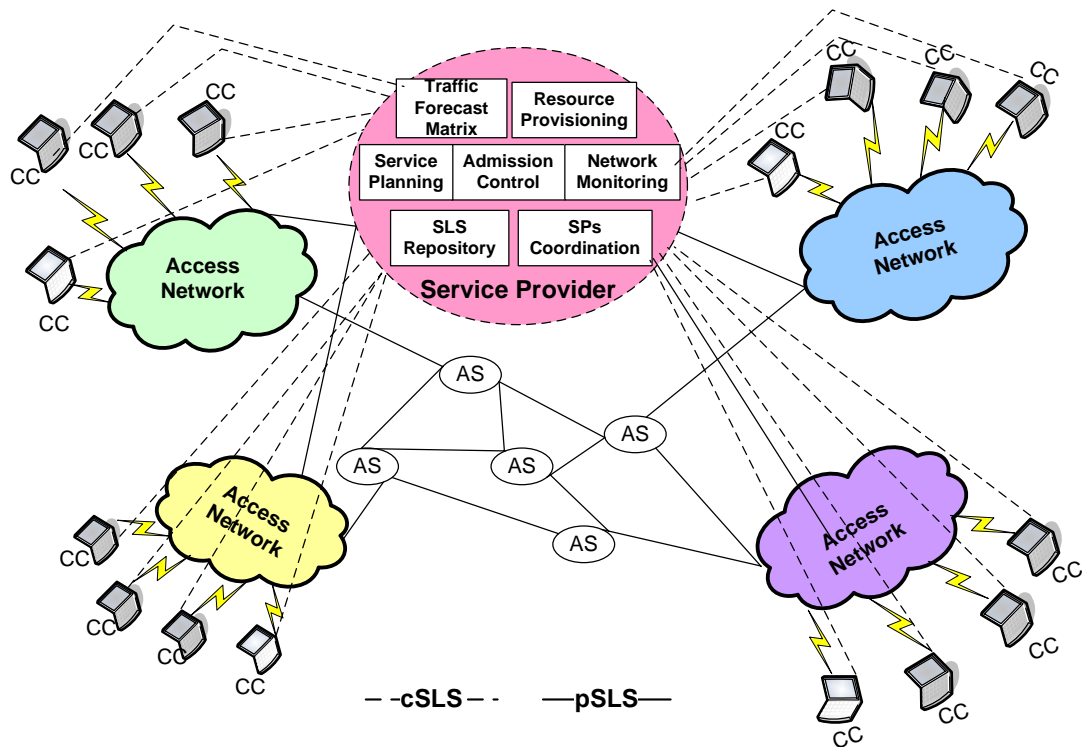


Figure 5-1: SP-driven P2P Network

SP has a global view of all the peers that are participating in video streaming and IPTV service delivery mechanism. The entities present within SP domain have their distinguished role in the overall service planning and efficient IPTV service delivery. SP is responsible for managing different entities and performs major functionalities that include:

- **Traffic Forecast Matrix:** Traffic forecast matrix is an important entity for the proposed SP-Driven P2P network. This entity provides a global view of currently available and under utilized network resources. The information is further used for the selection of an appropriate end-to-end path for actual content delivery. Traffic forecast matrix is continuously updated to provide the exact information of the network resources.
- **Network Monitoring & Resource Provisioning:** Network monitoring plays an important role to measure the actual network conditions. Network monitoring provides basic information for the traffic forecast matrix. Such network monitoring is further used for the resource provisioning.

- **Service Planning:** Service planning module defines the set of rules for the offered services.
- **SLS Repository:** The SLS repository keeps information of all the service-level negotiations among different entities.
- **Admission Control:** The admission control mechanism is implemented to prevent unauthorized access of network by admitting or rejecting traffic flows over certain network links.
- **SPs Coordination:** SPs coordination block is responsible for the negotiations among different SPs through explicit communication for efficient utilization of shared resources.

In SP-driven P2P framework, traffic forecast matrix has a distinguished role beside other entities controlled by SP. This traffic forecast matrix is continuously updated for the current available and under utilized network links. The resource allocations are performed on the basis of requested TSPEC (traffic specification) by each receiver peers intending to receive IPTV services. In the current practice, the Service Provider (SP) is responsible for aggregating content from multiple content providers and offering/delivering the content to consumers via best effort or QoS-enabled infrastructure. SPs may not necessarily own or administer any networking infrastructure, but they should interact with network providers through SLA (Service Level Agreement). For the resource provisioning SLA (service level agreements) and SLS (service level specifications) are negotiated between the NPs and SPs for determining requirements for the service delivery. SLAs describe the characteristics of the service offering and the responsibilities of the parties involved for using/providing the offered service. The SLS (Service Level Specification) is the technical characteristics of the service offered in the context of a SLA. The service technical characteristics refer to the provisioning aspects of the service e.g. request, activation and delivery aspects from network perspectives. There is a distinction between customer to provider SLSs and provider-to-provider SLSs. The cSLA/cSLS is established between end-customers and service/network providers. The pSLS is established between the service and network providers or between network providers.

The SLS template is considered as the nucleus of IP QoS-based services that includes the following attributes: SLS Identification, Scope of SLS, Flow Identification, Traffic Conformance & Characteristics, Traffic Excess Treatment, Performance Guarantees to be given to the traffic, etc. This information plays an important role for overall QoS provisioning using admission control.

The traffic forecast matrix updates are also subject to these SLAs and SLSs negotiations. In SP-driven framework, a Super node tracker (SN-tracker) is another important entity that is part of the peers in P2P network and it may be owned by SP or chosen par SP. SP administers this Super node tracker for updating the traffic forecast matrix. It provides three kinds of services to the P2P applications including capability, policy and descriptor.

- **Capability:** Super node tracker specifies the capabilities of the network provider such as different classes of offered services.

- **Policy:** Super node tracker specifies how a network provider would like its network to be utilized by all the applications to avoid the inefficient interaction between P2P and non-P2P applications.
- **Descriptor:** Super node tracker specifies a descriptor for network status itself where it provides the autonomous system's (AS) id, the geographic peer location that is further used for overlay organization etc.

We believe that such mechanism can address the issues of inefficient utilization of network resources and inefficient interaction among the applications. Moreover, such mechanism can be helpful to gain additional benefits for SPs and NPs due to some additional control and reduce the infrastructure costs. For the P2P applications perspective, SP-driven P2P framework enhances the overall QoS for the received services. This framework guides P2P applications to achieve more efficient network usage by avoiding expensive/congested network links to more desirable links that eventually ensure the smooth content delivery with the less start-up delay and latency.

5.3 Related Work & Motivation

The advances and popularity of P2P networks have drawn attentions of both academia and industry to propose solutions for the real-time streaming applications. P2P-based IPTV service delivery is hot area of study nowadays and a number of solutions are available for the subject. In this section, we briefly describe some of the existing work for IPTV service delivery along with our motivation to propose SP-driven P2P framework.

Regarding admission control mechanisms for P2P, different mechanisms have been proposed in this context. In [136], a distributed rate control mechanism is provided for P2P networks. This mechanism is based on the congestion awareness which exploits an optimization model. Each service provider allocates its rate capacity on the basis of the offered price by service consumer. The mechanism ensures fair share of available resources for participating peers. For the multi-source downloading, the rate control algorithm needs to run at both service provider and receiver end to determine the upload and download capacities. This model cannot be adapted to real business model since SLA needs to be established before service invocation.

In [137], an admission control mechanism for media streaming over P2P networks is presented. The admission control algorithm intends to maximize the load allocation problem for the number of concurrent requests for a given set of peers. The proposed algorithm provides resilience for the smooth streaming that is assured by the reliability of peers. In the case of degradation of peer's reliability and/or network congestion, the admission control mechanism adapts to ensue smooth streaming. The proposed architecture is hybrid since it combines both pure peer-to-peer streaming model and centralized server model in order to take advantages of them.

In [138], an architecture for QoS-aware P2P multimedia network is presented. The QoS is ensured specially by admission control manager that is responsible for assigning the network resources. The main focus of this architecture is on the utilization of a distributed RSVP, which we believe is not scalable. In [139] and [140], rate allocation mechanisms are presented for the video streaming services over wireless networks. Paper [139] introduces a rate allocation and network

selection optimization framework for clients accessing multiple applications over parallel networks. The optimization process takes into account the available network resources and the connection parameters of clients, along with the specific quality requirements of each application. In [140], a congestion-distortion optimized framework is investigated for streaming multiple videos in a common wireless ad-hoc network. The allocated rate at each video stream is chosen to achieve a common trade-off between network congestion and reduction of video distortion.

P4P [131] is a framework that provides an extension of the classical P2P framework to address the issues arising with the absence of any control over P2P. P4P framework provides a mechanism that allows effective control over network traffic among network providers and certain applications. Such cooperation can be helpful in improving the QoS for application as well as for efficient service provision by the service providers. However, this framework is relatively young and no real mechanism has been proposed by the working group.

Application-Layer Traffic Optimization (ALTO) [130] is IETF work in progress group which started on April 2008, to deal with the problems related to traffic optimization at application layer. The goal of ALTO is to provide information which can help peer-to-peer (P2P) applications to make better decisions with respect to peer selection. These solutions can be adaptable for non P2P applications as well. Peer selection strategy is considered as the challenge in P2P networks and is being tackled by ALTO WG.

ALTO describes that it would be difficult or even impossible for application entities to acquire this information by other mechanisms (e.g., using measurements between the peers of a P2P overlay), because of complexity or because it is based on network topology information, network operational costs, or network policies, which the respective network provider does not want to disclose in detail to third parties. The commonly used parameters for peer's selection are causing the problem of network inefficiency. In this scenario, ALTO intends to provide topology information services regarding the underlying network that can facilitate P2P applications to choose the best endpoints.

In the same context, the Transport for Advanced Networking Applications (TANA) [144] is another IETF working group working on the networking traffic. TANA is focusing on two issues, (1) to standardize end-to-end congestion control that enables advanced applications to minimize the delay that is introduced in the bottleneck network links and (2) multiple connections used by the P2P applications. A standardize congestion control mechanism will allow the P2P and other applications to send large data over networks while minimizing the latency across the congested links. The existing P2P applications open many connections for the data transfer. Many of these connections are only used for the small metadata pieces that complicate the overall network structure. TANA proposes to limit these numbers of connections.

In contrast to the other studies, we focus on the SP-driven P2P networking architecture, where SPs have explicit communications with other entities to efficiently utilize available resources. The SP negotiates with the network provider to maintain its traffic forecast matrix that is further used to allocate the available resources for the streaming services. The potential use of admission control mechanism with P2P networks for service delivery allows the reduction of the network traffic load on the central servers.

In P2P networks, the participating peers possess heterogeneous network capabilities for their uplink and downlink. So, in this chapter, we focus on a scenario where a single peer receives media content from multiple sender peers. We identify certain network constraints for the admission control by using the traffic specification (TSPEC) model [112]. In our target architecture, we focus on delivering scalable video coding (SVC) composed of different layers or tiers [135]. This video encoding scheme helps us to evaluate the delivery of different SVC layers coming from different senders.

There exist a number of solutions for IPTV service delivery over the P2P networks. Although, P2P systems have addressed the issue of scalability and the available solutions have shown noticeable performance, however it leads to certain deficiencies that include inefficient utilization of network resources and inefficient interaction with other applications. Therefore, the motivation behind our SP-driven P2P framework is to provide a mechanism for IPTV service delivery that exploits the functionality of P2P networks while addressing the aforementioned problems. Furthermore, the proposed mechanism prevents the unauthorized peers accessing the available network resources. The detailed mechanism is presented in the following sections.

5.4 SP-Driven IPTV Framework for Linear to Non-Linear Service Distribution

IPTV is considered the next killer application over the Internet due to its promising features. It facilitates the consumers to have the opportunity to use interactive TV functionalities either in the real-time streaming mode or in video-on-demand (VoD) mode. Currently, interactivity is considered as an integral feature of IPTV services. In the SP-driven P2P framework, we intend to support the transmission from linear to non-linear IPTV service delivery along with the other interactive functionalities. Such non-linear service distribution is deemed to be most fascinating feature of the IPTV service that allows users to use the service according to their preferences. The aim for the non-linear distribution is to allow the consumption of content shifts from a linear pre-packaged approach, to a personalized mode (“what the user wants, when he wants, on the terminal he wants”). In the non-linear distribution any user can access the pre-recorded content in VoD mode, either directly from the main content servers or with participating peers.

SP-driven P2P networks architecture intend to address the issues of network efficiency and inefficiency interaction among the P2P and non-P2P applications. The main objective of this mechanism is to enhance the overall QoS for the IPTV service delivery while providing incentives and effective coordination among all the stakeholders. An abstract view of the SP-driven P2P network for the IPTV service delivery is present in Figure 5-2. The participating peers are organized in different overlay networks. The overlay network organization is assisted by the SP to allow users to consume contents in their personalize way. This SP-driven peers organization serves the requirements of the entire heterogeneous client forming the P2P network. The content searching is performed in the distributed P2P fashion.

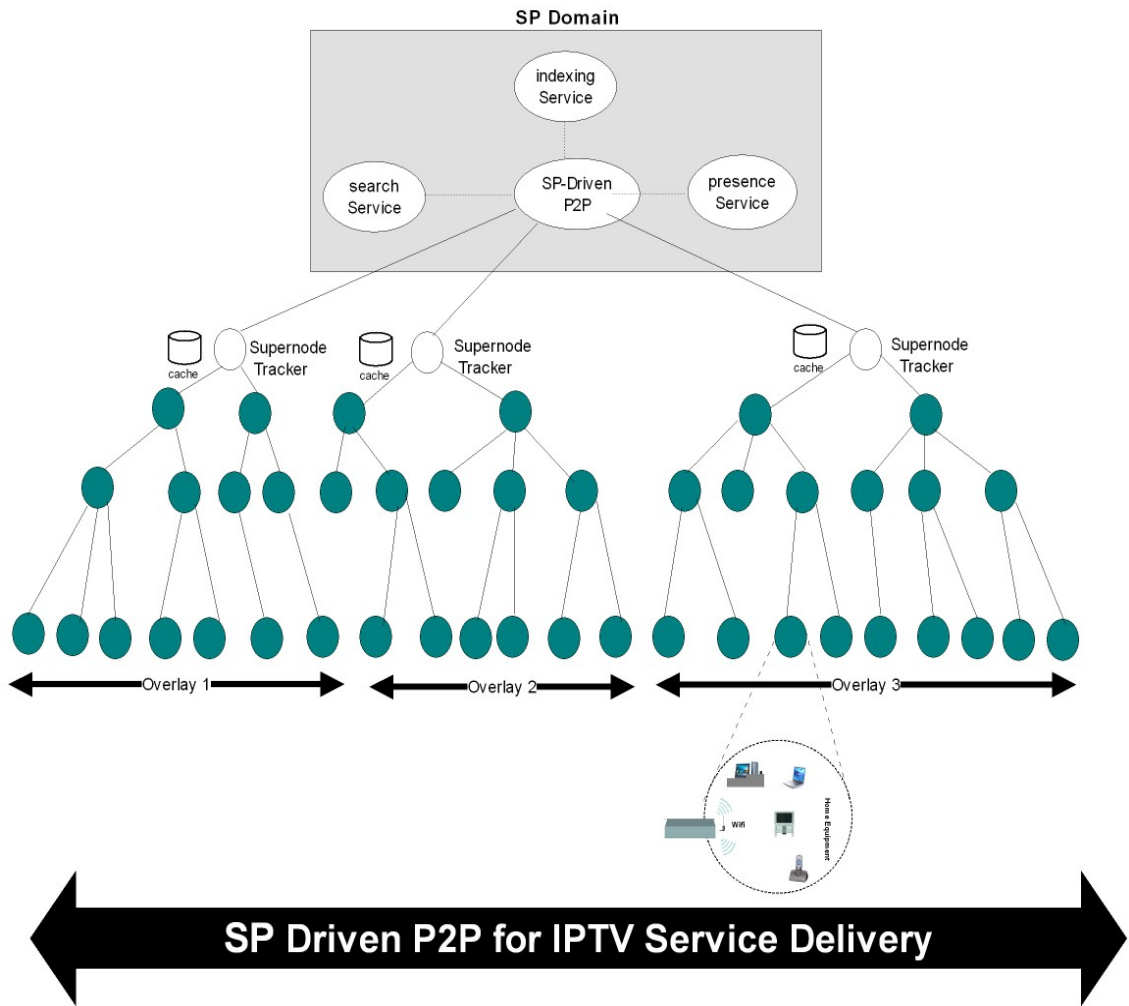


Figure 5-2: A big picture for SP-driven IPTV Service delivery

SP-driven P2P framework also ensures QoS provisioning by preventing the unauthorized peers from accessing the SP-driven P2P overlays. This QoS provisioning is performed by an admission control mechanism that is implemented to prevent unauthorized access of network by admitting or rejecting traffic flows over certain network links. This admission control mechanism resolves the issue of free riding in some extent. The QoS provisioning mechanism through admission control is presented in following section. A brief summary of main phases for the IPTV service delivery is described in the Table 5-1.

Phase	Description
Bootstrapping	While joining the P2P network peers execute a bootstrapping function. In this phase new peers discover other on-line peers and establish a neighboring connection with these peers. No peer can participate in the content sharing process without this bootstrapping. In our mechanism, this bootstrapping process is assisted by the SP.

Searching	After the bootstrapping, peers search for the desire media contents. The searching is strongly influenced by the choice of neighbor peers resulting from bootstrapping step. This search process is facilitated by the SP that keeps tracks of the seeds (peers sharing their content) and leechers (peers also downloading content) in the P2P networks.
Resource Allocation	Receiver peer contacts the SP with a list of best sender peer with whom it aspires to start streaming. SP allocates the end-to-end aggregate pipes for the actual content delivery on the basis of available network resources. The resource allocation is influenced by the SLA and SLS among the SP and NP domains for the QoS provisioning.
Media Streaming	In this phase, actual streaming of audio/video content is performed. In SP-driven P2P framework, a single receiver selects multiple sender peers to receive media content. The selection of the multiple senders leads to the smooth content delivery but requires a careful scheduling among the sender peers.
Active Monitoring	Although the actual content delivery in the SP-driven framework is carried out using the designated QoS enabled pipes but it still requires active monitoring of the active network links to cope with any abrupt changes. We apply peer switching by selecting another best peer for the smooth content delivery
QoS Adaptation	QoS adaptation plays a vital role while assigning the different SVC video tiers to different sender peers. The received data throughput is compared against the expected throughput from each sender peer to maintain an acceptable level of QoS throughout the streaming session. In this phase, we incorporate stream switching or video adaptation to maintain the smooth content delivery.

Table 5-1: IPTV Service Delivery over SP-driven P2P Framework

5.5 End-to-End Content Delivery over SP-driven P2P Networks & QoS Provisioning through Admission Control Mechanism

The different entities involved in the content distribution chain that provide end-to-end (e2e) service delivery with QoS provision may include: Network Providers (NPs), Service Providers (SPs), Access Network Providers (ANPs), Content Providers (CPs), and Service/Content Consumers (CCs). Each entity has a significant role for delivering the requested service along the e2e QoS-enabled path. In general, and in case of unicast service, CCs request the service directly from the SP. This later establishes a QoS-enabled e2e path over aggregate pipes for real-time media content delivery [141]. However, the e2e QoS provision across P2P networks is more challenging not only because of the presence of many paths between CC and potential content senders but also due to the heterogeneity of consumers and P2P architecture, where peers may enter and leave the P2P system without prior notification. On the other hand, P2P system provides better scalability for enabling QoS service provision. P2P provides an efficient infrastructure to use available networking resources in a more transparent, scalable and cost-effective way. To ensure smooth service delivery in SP-driven P2P system, the service provider has to monitor the network resources regularly (service level monitoring) and/or to perform admission control before service invocation.

Our proposed mechanism is based on admission control that provides QoS for the video streaming applications across P2P networks. This mechanism is based on the traffic descriptor admission control (TDAC) [142] where traffic descriptions of the video are provided by the sender peer prior to the connection establishments. The service provider establishes the e2e aggregated pipes on the basis of traffic descriptor (or traffic specification) and on its traffic forecast matrix to check for resources availability on certain network links.

Real-time applications such as video streaming and VoD are highly stringent against fluctuation in network conditions and require certain QoS mechanism to ensure smooth content delivery over P2P networks. So, the incorporation of an appropriate admission control strategy can be helpful to guarantee an acceptable level of QoS for real-time content delivery over P2P networks where a single peer seeks to receive video content from multiple sender peers having distinct characteristics. This admission control mechanism is used to control the amount of video traffic injected in each intra-domain and inter-domain links while accepting or rejecting the service requests to ensure QoS. Once transmitted, the video content are received in a particular buffer before the decoding phase. The streaming process is controlled by using a receiver side scheduler which ensures the conformance of the data arrival to the TSPEC model. If the conformance test fails during the content delivery phase, remedial action should be taken either by performing peer/stream switching or by quality adaptation [143]. Toward this end, NP applies a token bucket model for performing admission control of incoming traffic (video streams) from different CPs using TSPEC. TSPEC model is generally applied for QoS negotiations that are based on the pre-defined parameters. SP maintains and updates its traffic matrix to keep track of all used/available resources over the contracted NP domains.

The requested video stream is composed of multiple SVC descriptions/tiers and each sender peer contributes different video quality tiers. Receiver peer selects multiple sender peers for different video quality tiers and SP implements SP-AC mechanism for QoS provisioning. The

overall admission control mechanism is presented in Figure 5-3, where L_1 , L_2 , L_3 , and L_4 correspond to the Base tier and enhancement tiers of SVC video. The targeted peers are activated by SP on the basis of respective video description. In this figure, we have shown that a content consumer being part of the P2P network searches for some video content (e.g. Titanic movie). After having the list of potential peers having the request video content along with the description of offered video quality, the SP-Driven P2P can be invoked for best peer selection. In this case, Super node tracker performs its roles as defined in section 5.2. It provides information related to capability, policy and descriptor. This information can be offered as service that can be invoked. Service level agreements are established between SP and NP to authorize the requested services. Content consumer selects the sender peer to receive the video content by acquiring the status of all sender peers from the SP-Driven P2P network. The actual video delivery is assisted by SP-Driven P2P by activating the requested peer and admitting the traffic flow by NP.

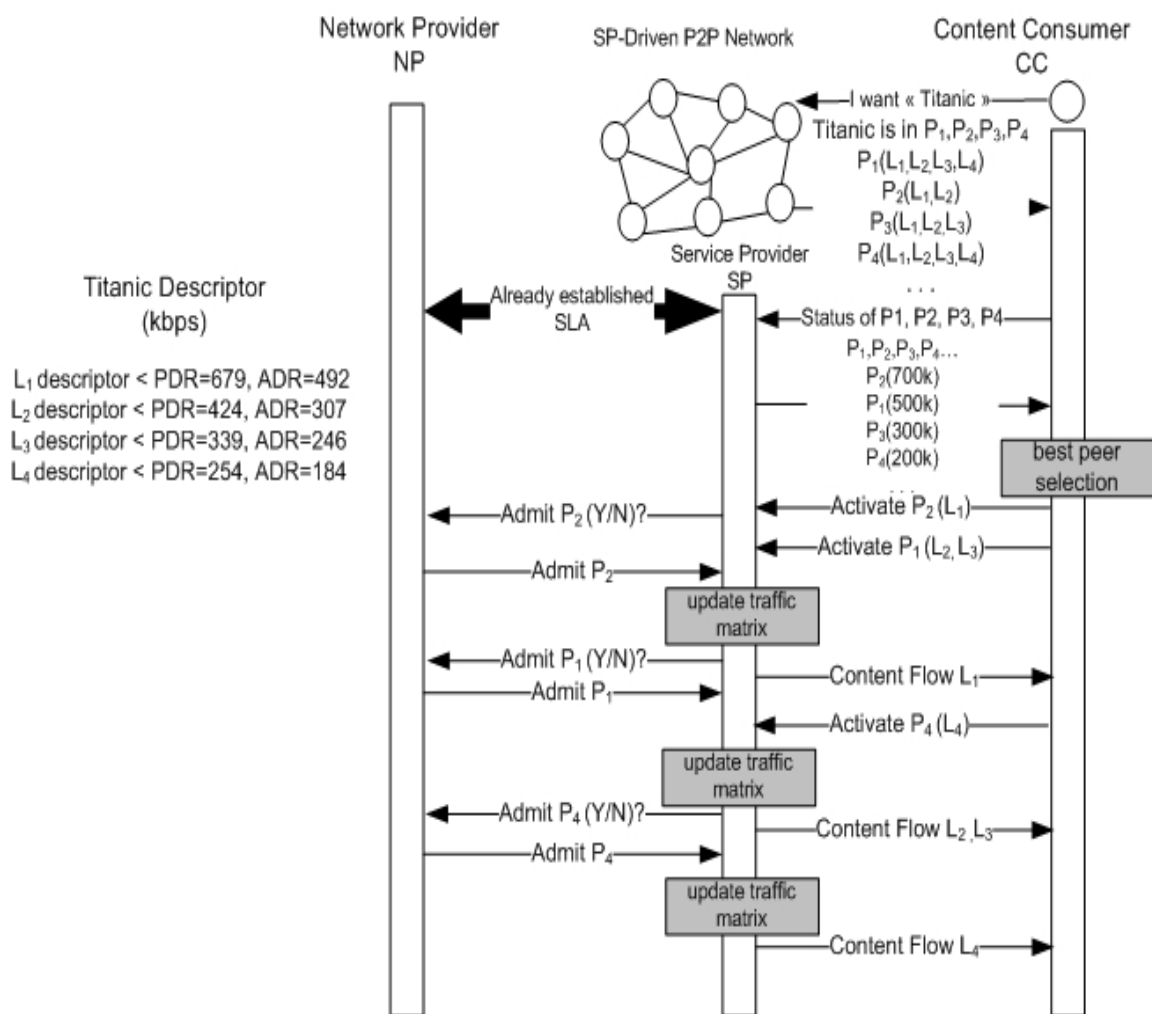


Figure 5-3: P2P based SP-AC Mechanism

Since the underlying network topology is based on the P2P architecture, there exist several intermediate nodes involved in the content delivery along the e2e path between sender and receiver peers. Thus, we propose to use a cascade model [141] for SLA/SLS negotiations between SPs and NPs to ensure the overall QoS along the E2E path. The signaling protocol for the TSPEC negotiations is out of scope for this work.

5.6 Performance Evaluation

We evaluated the proposed mechanism using ns-2 simulator [112]. We use the SVC based video encoding for the MPEG-4 video trace file ‘‘Simpsons’’ [113] into four different video tiers, i.e. base tier offering 40% of original video quality, enhancement tier 1, 2, and 3 offering 25%, 20% and 15% of the video quality respectively. We distribute SVC based video tiers among different sender peers to examine the performance of the proposed admission control based QoS mechanism. The video traces used for simulations provide VBR (variable bit rate) video stream and SP allocates the network resources on the basis of requested video descriptions. We perform simulations for two scenarios as follows:

- **Scenario without Admission Control based Quality Adaptation:** In this case, P2P system works like simple streaming mode without any admission control mechanism.
- **Scenario with Quality Adaptation:** In this case, QoS provision mechanism is implemented by incorporating admission control where SP allocates the end-to-end bandwidth for the content delivery among sender and receiver peers as described in section 3.

We focus on many-to-one (multisource) streaming scenario where a single receiver (CC) intends to receive the real-time video packets from multiple sender peers (CPs). To address the dynamicity of P2P networks, we implement a peer/stream switching mechanism as presented in [143]. The CC requests the video file directly from the service provider. SP has a global view of network condition with explicit communication with network providers and maintains a traffic forecast matrix for the on-going communications and available bandwidth over certain paths. The communication among SPs and NPs are carried by SLA/SLS negotiations which are out of the scope of this chapter. For the scenario with admission control based quality adaptation, the actual content delivery is carried out by allocation of e2e aggregate pipes on the basis of TSPEC (traffic specification) and no other traffic is allowed to pass through these dedicated pipes. These TSPEC parameters include the peak data rate, average data rates and inter packet arrival delay for the each video quality tier while satisfying the constraints as presented in chapter 4 (Eq. 4-1, Eq. 4-2, Eq 4-3, and Eq. 4-4). These constraints allow the admission control mechanism to admit new traffic flow over certain links. Table 5-2 presents the peak and average data rates of video tiers used for the simulations.

	Original Video	Base Tier	Enh. Tier 1	Enh. Tier 2	Enh. Tier 3
Peak Data Rate	1699	679	424	339	254
Average Data Rate	1231	492	307	246	184

Table 5-2: Peak and Average Data Rates of Video (kbps)

Figure 5-4 presents the received video throughput in both scenarios along with the expected video quality. Figure 5-5 presents the packet drops ratio for the received video at receiver-end in both scenarios. We notice huge packet loss ratio in the scenario without applying any admission control based QoS mechanism. These results give a brief picture of the SP-driven traffic forecast matrix that is used for the installation of the QoS enabled aggregate pipes over the e2e links. The traffic descriptor-based admission control mechanism enables the QoS provision for the streaming

application but still it requires more analysis to determine its performance for other parameters (SLA/SLS parameters).

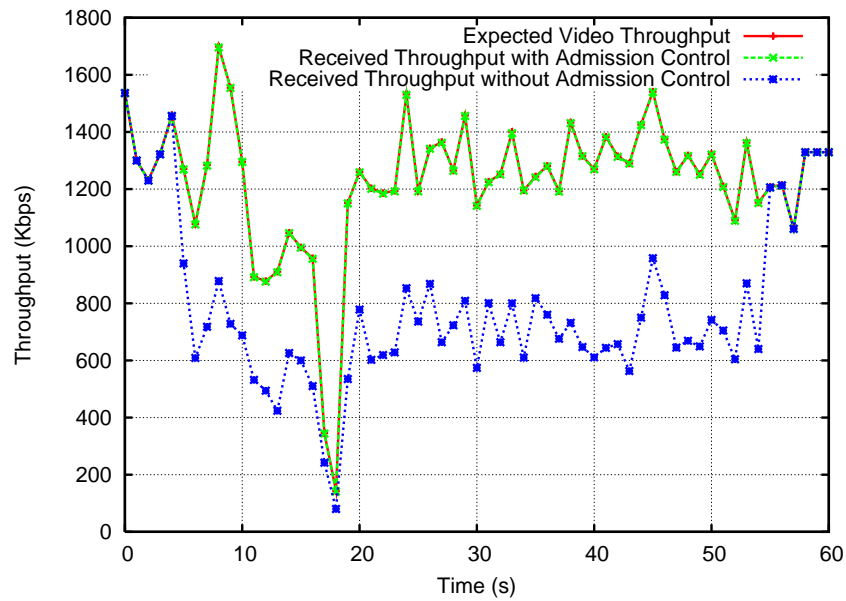


Figure 5-4: Received Video Throughput

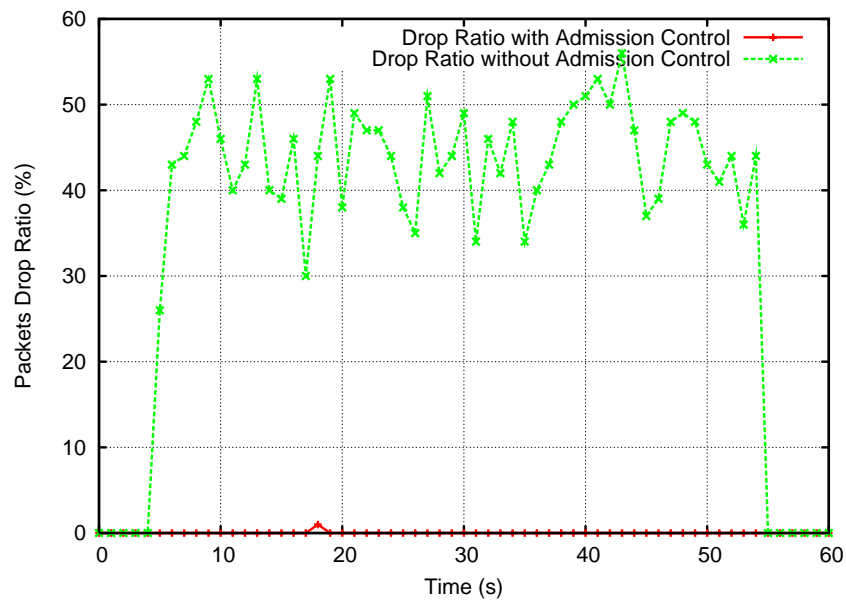


Figure 5-5: Packets Drop Ratio

Table 5-3 presents a brief summary of simulation results for received video quality in both scenarios. We notice a clear difference between the received video qualities that is almost 100% when streaming mechanism is supported by proposed admission control mechanism, while in the other scenario video content are delivered over shared network paths. The other traffic passing over the same networking link results into huge packet drop ratio and affect the received video quality drastically that is not acceptable for real-time content delivery. We notice that enhancement

video tier 3 is received 100% even in the scenario when we did not apply any admission control mechanism, but it cannot contribute to enhance the received video quality when the base video tiers not received completely. Figure 5-6, Figure 5-7, Figure 5-8 and Figure 5-9 present the comparison between received and original base tiers, enhancement tier 1, enhanced tier 2, and enhanced tier 3 respectively. We can see clearly that our proposed mechanism performs a smooth video delivery with higher quality and lower loss that improves the overall QoS for the received video. The enhanced packets throughput and lower packet drop ratio result into enhanced received QoS and justify the viability of proposed admission control based QoS provision mechanism over P2P networks.

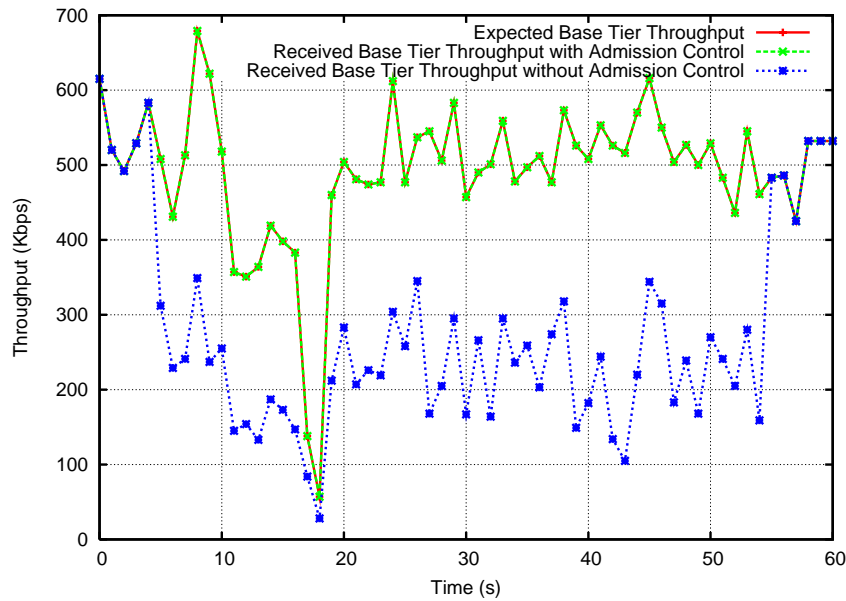


Figure 5-6: Received Video Throughput for Base Tier

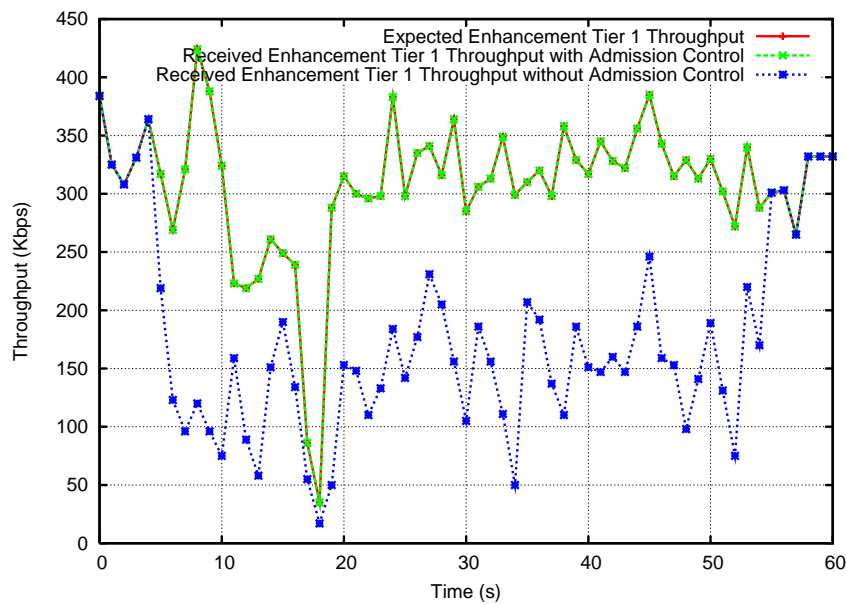


Figure 5-7: Received Video Throughput for Enhancement Tier 1

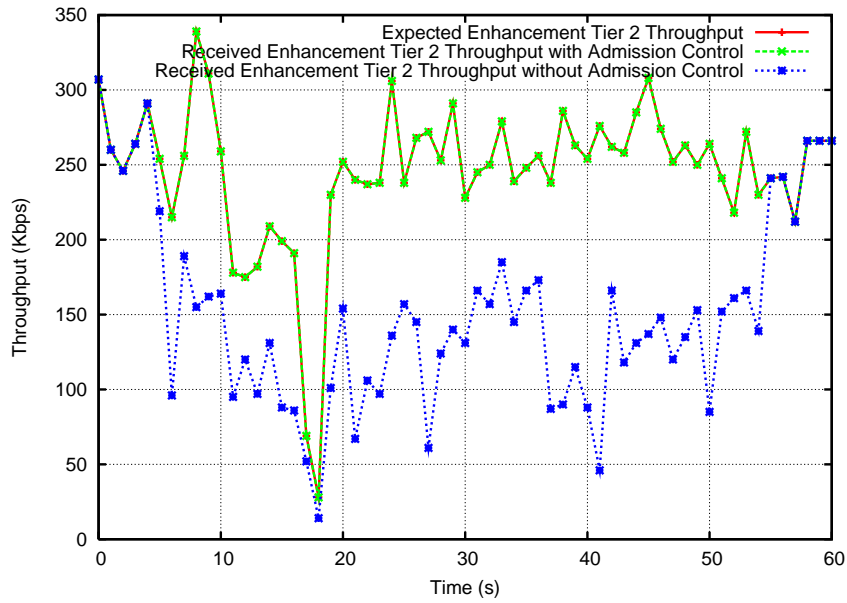


Figure 5-8: Received Video Throughput for Enhancement Tier 2

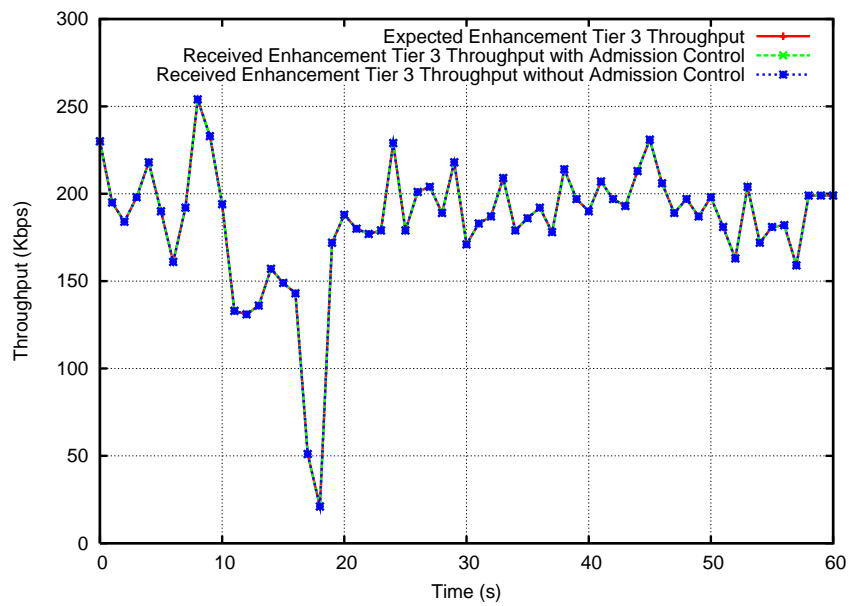


Figure 5-9: Received Video Throughput for Enhancement Tier 3

	Original Video	Base Tier	Enh. Tier 1	Enh. Tier 2	Enh. Tier 3
Video Throughput with Admission Control	100	100	100	100	100
Video Throughput without Admission Control	64	56	57	61	100

Table 5-3: Received Video Quality (%)

5.7 Conclusion

In this chapter, we presented a framework for the IPTV service delivery over P2P networks. The framework provides a comprehensive cooperation among the service providers (SPs) and P2P applications. Furthermore, we presented an admission control based QoS provision mechanism for the real-time video streaming over SP-driven P2P environment. The SP-driven approach makes our proposed mechanism distinguished to the other available solutions and provides efficient utilization of network resources. We observe a significant improvement in QoS for the received video in the scenario when our QoS provision mechanism is applied.

To ensure smooth QoS for the real-time streaming application over P2P networks is not trivial. Furthermore, free-riding is considered a challenging issue. Thus, to address this issue, we present an admission control mechanism over SP-Driven P2P framework. This admission control mechanism ensures the QoS for the real-time video streaming and prevents the unauthorized traffic to pass by the network links.

Currently, we have evaluated the proposed admission control mechanism for a simple scenario where a single receiver selects multiple sender peers to receive the video content. Currently, all peers belong to a single SP. In future, we aim to evaluate this mechanism for large scale networks with real implementation and to evaluate the complexity for installing this mechanism. Furthermore, we plan to extend this mechanism for the efficient cooperation among different SPs for IPTV service delivery.

Chapter 6

6 Conclusion and Perspectives

This chapter concludes the dissertation by summarizing our major contributions and suggesting some key directions for future perspectives.

P2P networks and related applications services are growing exponentially and are impacting every aspect of modern life. The video sharing among communities have revolutionized the modern societies by facilitating and providing the means to share their video content with each others. It is need of time to propose new architectures and mechanisms with the increasing demand of real-time content sharing across the networks.

Therefore in this dissertation, we have proposed an adaptive streaming mechanism to facilitate the selection of best sender peers among the available peers for improving user perceived quality of interactive video streaming applications over P2P networks. The proposed mechanism utilizes the characteristics of SVC standard and P2P frameworks to ensure the QoS.

6.1 Summary of Key Contributions

In this dissertation, we proposed an adaptive video streaming mechanism for QoS provisioning over P2P networks. The proposed receiver-centric streaming mechanism leverages the characteristics of the Scalable Video Coding (SVC) and P2P networks. The key components of the streaming mechanism include adaptive mechanism (1) organization of sender peers in hybrid overlay networks, (2) efficient sender peers selection and switching mechanism and (3) assignment of video parts to different sender peers. The best peer selection mechanism is based on the hybrid overlay networks organization and the sender peers are organized in these overlays on the basis of their respective priority towards offered QoS. The overlay organization makes it possible to select a best peer for individual SVC video tier in $\Theta 1$ time that is promising when there are a large number of sender peers available in the network. The main objective of this mechanism is to ensure smooth video quality delivery with the guarantee of the stable QoS during the streaming session. In this regard, we presented a peer/stream switching mechanism to cope with the dynamic behavior of the sender peer, after active monitoring of the network links to ensure the smooth streaming quality. The proposed quality adaptation is well suited for to offer video streaming to heterogeneous terminals.

We presented a receiver-side scheduling mechanism that is used to assign different video streams to different sender peers. The scheduler determines the video schedules on the basis of priority offered to QoS in streaming mechanism and is facilitated by different receiver side buffers. These buffers receive the video streams before their decoding. An admission control mechanism is incorporated on the basis of TSPEC model of the video streams. The proposed mechanism is

evaluated using simulations and a noticeable improvement in the received video quality is observed under the proposed mechanism. In fact, a significant improvement of the received throughput especially for the important video quality tiers, lower packets drop ratio for the important quality tiers, and a considerable improvement in the received packets delay are observed.

In the last chapter, we presented a framework for the SP-Driven P2P networks that is further used for the IPTV service delivery. The framework based on the cooperation among the service providers (SPs) and P2P applications. Furthermore, we presented an admission control based QoS provision mechanism for the real-time video streaming over SP-driven P2P environment. The SP-driven approach makes our proposed mechanism distinguished from the other available solutions and provides efficient utilization of network resources. We observe a significant improvement in QoS for the received video in the scenario when our QoS provision mechanism is applied.

To ensure smooth QoS for the real-time streaming application over P2P networks is not trivial. Furthermore, free-riding is considered challenging issues. This admission control mechanism ensures the QoS for the real-time video streaming and prevent the unauthorized traffic to pass by the network links.

We evaluated the proposed mechanisms using network simulations and evaluated the results for different parameters that contribute for offered QoS. The simulation results show noticeable improvements for the scenarios with our proposed mechanisms in terms of high video throughput, low packet drop ratio, and low packet transmission delay.

6.2 Open Issues and Future Perspectives

As part of our future perspectives, we plan to pursue our research activity in the following directions:

- **Minimizing the Impact of Loss and Jitter:** To ensure the high QoS for the real-time video streaming applications, it is necessary to maintain a very low packet loss (0% is always preferred) and to minimize the jitter. However, to maintain lower packet loss and lower jitter while coping with the dynamicity of P2P networks is not trivial. We need to incorporate more intelligent techniques to address these issues. In this regard, we should identify different receiver peer according to their priorities and their contributions in terms of content sharing with other peers. We also intend to incorporate different coding techniques to minimize the packet loss ratio that are used to regenerate the lost packet. In this regard, forward error correction (FEC) and network coding are techniques that can be used with our proposed mechanisms.
- **Packet Video Streaming in Wireless Ad-hoc Networks and Wireless Mesh Networks:** The main focus of our work remains in the scope of fixed P2P networks. However, currently P2P networks are comprised of different access networks including fixed networks (ADSL, Cable, ...) and mobile networks (UMTS, WiFi, WiMax, ...). Thus, a mechanism is necessary to support video streaming for all the network terminals connecting to the P2P networks through different access networks.

- **Analytical Modeling:** Analytical modeling is considered as an important tool to evaluate any proposed mechanism. Thus, a comprehensive analytical modeling is required to justify the proposed mechanism..
- **Implementation Issues:** The performance evaluations for the proposed mechanisms for guaranteeing QoS for the video streaming over P2P are done using network simulation. However, these simulations are not enough to determine the actual performance of any mechanism. Thus, we plan to implement the core components of the mechanism in real. This can be done by implementing the proposed mechanism on the top of some open-source P2P system.
- **Enhancement of SP-Driven P2P Framework:** The cooperation among service providers and P2P applications improves the performance for all key players involved in the content delivery chain. However, we need to extend our proposed SP-Driven P2P framework for the communication between service providers and network providers, communication between content consumers and service providers, and interaction between different service providers. Furthermore, we need to evaluate different parameters for the best peer selection in the context of cooperation between P2P applications and service providers.

Chapter 7

7 Résumé Detaille de la Thèse

Le réseau Internet se développe, change, s'organise et innove constamment de nouvelles applications et de nouveaux services. Ces dernières années, l'un des événements qui a le plus marqué la croissance d'Internet est certainement le développement des réseaux Pair-à-Pair ou Peer-to-Peer (P2P). Le terme pair-à-pair fait référence à une classe de systèmes ou d'applications qui utilisent des ressources distribuées. Dans ce type de réseaux, les utilisateurs s'organisent en groupe, en communauté ou en tant qu'utilisateurs simples. Ce modèle connaît un très grand succès dans la dernière décennie et est actuellement très connu dans la communauté des chercheurs et s'est traduit par l'apparition d'un nombre important de travaux de recherche et des projets. Ces dernières années, les réseaux P2P se sont généralisés. Ils se sont présentés comme une formidable opportunité de croissance du marché de l'accès haut-débit à Internet. Ils offrent aussi des possibilités et des perspectives de développement de nouvelles applications. Grâce aux réseaux P2P, de plus en plus de personnes partagent leurs documents, photos, vidéos, musiques, et autres données via Internet. Plusieurs systèmes ont ainsi vu le jour pour supporter cette nouvelle donne. Le phénomène P2P a vraiment révolutionné la façon dont les utilisateurs consomment et interagissent avec l'information en particulier les flux multimédia. Actuellement, on estime que le trafic Internet est dominé à 70% par des applications P2P.

Les réseaux P2P présentent différentes caractéristiques: la décentralisation, l'extensibilité et l'hétérogénéité, communication symétrique des utilisateurs, l'auto-organisation, l'autonomie, et la sécurité des utilisateurs. Les systèmes informatiques et plus particulièrement les systèmes P2P peuvent être classés selon la taxonomie de décentralisations des systèmes et de la structure des réseaux. De nombreuses applications utilisent les réseaux P2P pour collaborer et permettre la distribution du contenu vers un grand nombre d'utilisateurs.

Dans notre travail de recherche, nous nous sommes intéressés à l'application de distribution de contenus basée sur l'architecture P2P. Le but de cette application est la distribution massive de contenu pour alléger la surcharge des serveurs prévus initialement à cette tâche. La plupart des produits P2P connus rentrent dans cette catégorie à savoir : FastTrack (KaZaA, Morpheus), Napster, eDonkey, Akamai, BitTorrent, etc. Le contenu multimédia (vidéo, images, audio) et les applications logicielles sont les deux ressources les plus partagées par ce genre d'applications.

Les applications de distribution de flux multimédia classiques sont basées sur le principe de fonctionnement très simple. Le fichier doit être complètement téléchargé avant sa consommation par l'utilisateur. Ce qui pose certains problèmes relatifs au temps de téléchargement, à l'occupation de l'espace disque, à la qualité et au type de contenu. En revanche, actuellement, la notion de service de diffusion temps réel de contenu multimédia voit le jour avec l'apparition de plusieurs applications telles que : CoopNet, SpreadIt, ZIGZAG, PALS, and PROMISE. Le principe de fonctionnement repose sur diffusion temps réel de flux multimédia (i.e., téléchargement et lecture simultanée d'un flux). Le streaming des flux vidéo sur les réseaux P2P présente des problèmes spécifiques, comparé au streaming en mode client/serveur. En plus de tous les problèmes hérités du protocole IP : bande passante non garantie, présence de perte dans le réseau, et délai de transmission et gigue non bornés.

Nous nous sommes intéressés à la distribution de flux vidéo temps réel sur les réseaux P2P. Dans ce contexte, l'objectif de cette thèse est d'analyser les différents problèmes et de proposer un mécanisme de transport vidéo temps réel avec le provisionnement de la qualité de Service (QoS). Notre approche de streaming des flux vidéo rentre dans le mode de diffusion multi-arborescences. Ce choix a été motivé par l'utilisation du format de codage vidéo (SVC) qui organise le flux vidéo en couches. Ainsi, dans notre approche chaque couche SVC est mappée sur un arbre de diffusion P2P. Trois contributions majeures ont été proposées.

7.1 Mécanisme de sélection des pairs

Premièrement, il s'agit d'un mécanisme de streaming vidéo adaptatif permettant de sélectionner les meilleurs pair émetteurs. Ce mécanisme permet de structurer les pairs dans des réseaux overlay hybrides avec une prise en charge des caractéristiques sémantiques du contenu et des caractéristiques physiques du lien de transport. Ce mécanisme permet la sélection des meilleurs pairs (noeuds) participant à la session de diffusion vidéo. Ceci permet la réception du flux à partir des pairs disposants de la meilleure QoS. Ce mécanisme se base sur des mesures actives permanentes de la qualité de bout-en-bout des liens entre le récepteur et les clients pairs potentiels. Les pairs disposant de la même qualité de service sont organisés ensemble dans un même réseau overlay pour faciliter leur recherche et leur sélection. Ainsi, ce mécanisme est accompagné de stratégies de regroupement de pairs en fonction de différents critères (sémantique et physique) afin d'améliorer le fonctionnement global du système de diffusion vidéo.

Le mécanisme de sélection des pairs a pour but de choisir parmi l'ensemble de pairs possibles ceux qui fournissent la meilleure qualité de service. Les pairs peuvent être choisis sur la base de plusieurs critères physiques et/ou sémantiques. Pour cela, nous nous sommes intéressés, dans un premier temps, aux critères physiques pour la sélection des pairs, à savoir les mesures du nombre de sauts, et du RTT qui séparent le récepteur de ses émetteurs. Dans notre architecture, le noeud

récepteur de flux coordonne le streaming à partir des pairs émetteurs de flux. Pour cela, il exécute un ensemble d'actions pour déterminer les meilleurs pairs disponibles en affectant un poids pour chacun d'eux. Le calcul des paramètres utilisés est possible grâce à des techniques de mesures actives des temps aller-retour (RTT) et du nombre de sauts (Nbre_Saut). Même si la mesure RTT en elle-même peut suffire pour estimer la qualité de lien, il faut aussi considérer le nombre de sauts pour refléter une stratégie de choix géographique (i.e. le noeud le plus proche). Il faut noter aussi que les mesures RTT utilisées sont des moyennes exponentielles qui permettent de lisser toute variation instantanée des valeurs RTT mesurées.

Ainsi, pour faciliter le processus de transmission et de sélection des noeuds dans le cadre de notre approche de streaming multi-arborescence, nous avons organisé les noeuds dans des réseaux overlay. Chaque arbre de streaming représente un overlay contenant plusieurs noeuds. Cette organisation est dite hybride puisqu'elle tient compte, à la fois, de la couche vidéo SVC à transmettre (critère sémantique) et des contraintes physiques de lien en terme d'indice de priorité. À l'intérieur de chaque overlay, les noeuds sont organisés suivant un arbre binaire de type « MinHeap ». Un arbre « MinHeap » binaire ou un monceau (tas) est un arbre dans lequel il existe un ordre entre un noeud et ses descendants. Dans le « MinHeap », chaque noeud a une valeur plus petite ou égale à celle de ses deux fils. Ceci nous permet d'accéder directement à la racine de l'arbre contenant le noeud avec le plus faible indice de priorité (c'est-à-dire le meilleur noeud). Avec cette organisation, chaque overlay maximise le débit qu'il peut offrir. Rappelons que le but de cette organisation est de maximiser la capacité de réception du noeud récepteur par le choix des meilleurs pairs afin de faciliter la gestion de l'adaptation vidéo. Ainsi, la couche de base SVC, qui est considérée comme la couche la plus importante, est transmise à partir des meilleurs noeuds, et ainsi de suite pour les couches d'amélioration.

Le mécanisme proposé est évalué avec des simulations. Les résultats des simulations effectuées prouvent que, grâce à ce mécanisme de sélection des meilleurs noeuds et à la gestion efficace de la mémoire tampon de réception, une meilleure QoS est fournie au flux vidéo, comparé à un streaming P2P ignorant les contraintes physiques des liens et sémantiques de la vidéo.

7.2 Mécanisme d'adaptation vidéo sur P2P

Nous avons présenté aussi un mécanisme d'adaptation et d'ordonnement de paquets vidéo combiné à une commutation de pairs et/ou de flux pour assurer un transport lisse. Ce mécanisme joue un rôle très important. La variation de la qualité des liens de communication couplée au caractère dynamique des noeuds impose que le streaming vidéo soit adaptatif, dynamique, et attentif à tout changement intervenant dans le réseau P2P, aussi bien au niveau des caractéristiques physiques (RTT, nombre de sauts) que des caractéristiques sémantiques (distribution des couches SVC). Le codage vidéo SVC offre de très grandes flexibilités d'adaptation de flux vidéo pour des

terminaux et des réseaux hétérogènes. SVC fournit des fonctionnalités pour déterminer la qualité d'un flux vidéo en termes de SNR, résolution temporelle et résolution spatiale. Nous avons spécifié quelques scénarios permettant à l'utilisateur de choisir les adaptations possibles pour augmenter/réduire la qualité du flux reçu. Quatre modèles d'adaptation importants ont été définis : horizontal, vertical, diagonal, et ZigZag. Dans chaque modèle, le niveau de qualité SNR peut être modifié. Chaque modèle augmente/réduit l'un des critères de qualité vidéo (c'est-à-dire SNR, résolution spatiale, résolution temporelle) par des adaptations.

Chaque couche SVC doit être décrite par un modèle TSPEC (Traffic Specification) qui donne les paramètres essentiels de description du trafic à envoyer par un ou plusieurs nœuds P2P. Ce mécanisme de commutation de flux et de pairs qui permet de maintenir un niveau de QoS acceptable en cas de problème de réception dû au vidage imprévu des mémoires tampons de réception. Dans ce cas, deux solutions se présentent : (a) dégrader la qualité du flux reçu, par des mécanismes d'adaptation vidéo dans le cadre du codage hiérarchique SVC ou (b) essayé de maintenir la qualité originale par l'identification des pairs en défaillance et le remplacement par des pairs assurant la qualité originale.

7.3 Concept de collaboration P2P/Fournisseur

Finalement, une architecture de collaboration entre les applications P2P et les fournisseurs de services / réseaux est proposée pour supporter un contrôle d'admission de flux. Ce mécanisme est basé sur la réservation de ressources de bout-en-bout assisté par le fournisseur de services.

Une nouvelle génération de réseaux P2P est en train de naître, celle-ci préconise une meilleure collaboration entre le réseau P2P et le fournisseur de services ou SP (Service Provider) ainsi que le fournisseur des infrastructures réseaux ou NP (Network Provider). En effet, SP et NP ont réalisé qu'ils sont en train de perdre le contrôle sur ce qui se passe dans leurs réseaux à cause du déploiement massif des réseaux P2P. Cette situation pose aussi des problèmes aux utilisateurs P2P qui veulent télécharger des contenus rapidement et avec une certaine garantie de QoS. Un mécanisme collaboratif entre les applications P2P et le fournisseur de services permet la sélection des meilleurs nœuds pour l'ouverture d'une session de streaming temps réel. Ce mécanisme pourrait être effectué efficacement par le SP, du moment qu'il a une maîtrise totale des ressources disponibles et de celles contractées avec d'autres fournisseurs (grâce aux SLA SP-NP et NP-NP). Le but est de permettre aux utilisateurs P2P de mieux consommer les contenus disponibles par une collaboration entre le réseau P2P et le fournisseur de service. Cette initiative est en cours de concrétisation dans le cadre du groupe de travail P4P (Provider Portal for P2P applications). L'IETF n'a pas échappé à ce mouvement par la création du groupe de travail ATLO (Application-Layer Traffic Optimization). Le groupe de travail ATLO définit actuellement la problématique

d'optimisation du trafic P2P par des mécanismes qui préconise une collaboration entre le réseau P2P et les différents fournisseurs.

Dans ce contexte, nous avons proposé un nouveau concept d'allocation de ressources et d'échange de topologie entre SP et le réseau P2P pour faciliter la sélection des meilleurs noeuds. Nous proposons, dans cette architecture, d'utiliser le modèle de réseaux P2P hybride qui fait appel à des super-noeuds (SN) « tracker ». De plus, nous proposons que le SN soit désigné par le SP. Ainsi, pour chaque demande d'allocation de ressources, un noeud consommateur effectue sa requête via le SN. Ce dernier, informe le SP de la demande d'allocation du noeud récepteur. Une fois l'allocation confirmée, et la matrice de trafic est mise à jour au niveau du SP, le streaming vidéo P2P peut commencer. Nous proposons que le noeud SN fournisse trois types de services : (1) informe les autres noeuds sur les capacités du fournisseur réseau tel que les classes de services utilisées, (2) spécifie quand et comment les ressources du fournisseur réseau seront utilisées, et (3) décrit la topologie du réseau tels que le numéro du AS (Autonomous System), position géographique des noeuds, etc. En bref, ce mécanisme permet au fournisseur de services (SP) et au réseau P2P de collaborer pour mieux servir l'utilisateur final.

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