

SP-driven QoS provision Mechanism for Real-Time Multimedia Streaming over P2P Networks using Admission Control

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Abstract. Enabling Quality of Service (QoS) for End-to-End real-time multimedia content delivery across heterogeneous P2P networks is challenging but vital for the efficient service provision. In this paper, we present an admission control based for the real-time multimedia streaming over Service Provider (SP) driven peer-to-Peer (P2P) networks. SP-driven P2P networks are the P2P networks where network providers have comprehensive control over network and provide the whole information to the service providers by explicit communication to deploy efficiently the real-time applications with improved QoS. The proposed QoS provision mechanism is based on the traffic descriptor admission control (TDAC) technique. The underlying video encoding scheme used to propose this solution is scalable video coding that is considered as the most promising video encoding scheme for real-time content delivery across heterogeneous networks as to ensure acceptable level of QoS.

1 Introduction

A rapid growth in the content distribution and sharing applications has been witnessed during the last decade. This exponential growth is the result of high popularity of peer-to-peer (P2P) phenomenon. P2P networks were initially designed for the content distribution across the networks but nowadays P2P architecture is widely used for many more real-time applications including multimedia streaming, multimedia / video on demand (VoD), P2P gaming, video conferencing, signaling of IP telephony, P2P-based IPTV services, etc. The delivery of these real-time applications is more challenging with an acceptable level of Quality-of-Service (QoS). These applications have adaptive bandwidth capabilities, but on the other hand have stringent delay, jitter, and packet loss requirements, which are not supported by current IP networks. Therefore, we need to design these multimedia applications capable of adapting to system and network resource constraints, (i.e., network-aware) whilst ensuring that end-user requirements are taken into consideration (user-centric).

There are different entities involved to ensure the end-to-end (e2e) service delivery with QoS provision that 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 the smooth service delivery along the end-to-end path. In general the CCs request for the media content directly to the SPs that provide the content to CCs in scalable way by establishing QoS-enabled aggregate pipes over end-to-end path. In the Service provider driven P2P framework, network providers have comprehensive view of network conditions and have explicit communication with P2P entities [1]. The available network resources are negotiated among the SPs and NPs regularly that are further used for the admission of new traffic over certain network links. These pipes are logically established by the Service Level Agreement/Specification (pSLA/pSLS) contracts between Service Providers and Network providers. SPs have a full global view of the available bandwidth over certain network links and such aggregate pipes are established on the basis of requested media content. Service provider maintains a traffic forecast matrix that provides the information regarding allocated and available bandwidth over certain network paths.

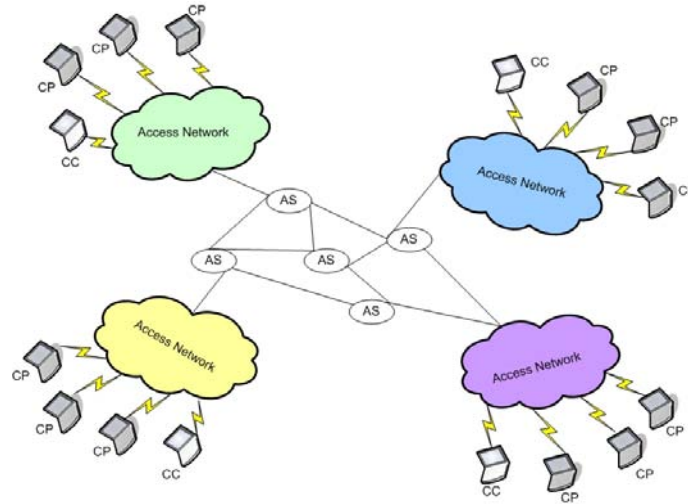


Fig. 1. Overall Network Topology

The end-to-end QoS provision for the real-time multimedia content delivery across P2P networks is even more challenging due to the heterogeneous and dynamic networking architecture. P2P framework is widely adapted for such service provision they provide an efficient infrastructure to use available networking resources in a more transparent, scalable and cost-effective way. Today, most of the multimedia services are provided by installation of set-top boxes at the content consumer's premises. These set-top boxes equipped with certain characteristics such as storage space, on which the client can store programs, movies, and other desired content. The service provider has full access for indexing, tracking and to use the content available in these set-top boxes. These characteristics enable service providers to use such boxes as independent content servers that lead towards efficient utilization of available network resources. Most of the CCs connect to the network using ADSL (Asymmetric Digital Subscriber Line) or CMTS (Cable Modem Termination System), where uplink capacity is 3-8 times lesser than that of the downlink. In this scenario SP

ensure the content delivery from multiple set-top boxes acting as individual sender peer (content provider). To ensure smooth service delivery service provider need to monitor the network conditions regularly because in P2P framework network conditions may change more often. Such abundant bandwidth degradation may affect the content delivery and is never acceptable for real-time applications that are more stringent against delay and jitter. Thus, to ensure smooth content delivery, we need to incorporate efficient collaborations among each content provider (sender peer) and content consumer (sender peer). A brief sketch of network topology is presented in Fig. 1 where any content consumer selects a certain number of content providers among the candidate list provided by the service provider and enables them to start the multimedia services.

In this paper, we present an admission control mechanism with end-to-end QoS for the multimedia streaming applications across P2P networks. This mechanism is based on the traffic descriptor-based admission control (TDAC) [2,3] where traffic descriptions are provided by the content consumer prior to the connection establishments. The service provider establish the end-to-end aggregate pipes on the basis of traffic descriptor and SP's traffic forecast matrix for the available bandwidth on certain network links. This paper is organized as follows: section 2 briefly describes the related work, section 3 describes QoS provision mechanism for multimedia streaming over P2P networks by introducing an admission control mechanism, section 4 concludes the paper along with highlighting some future perspectives.

2 Motivation and Related Works

There have been significant advancements observed for the deployment of real-time multimedia streaming application during last few years. Industry, Academia and research organizations are continuously putting their efforts to provide more reliable and cost effective solutions and are continuously introducing new application areas for such services. In this section, we discuss some of the existing work carried out in the context of the paper.

P4P [1] is a framework that is extension of the classical P2P framework. The P2P frameworks are widely used nowadays for a range of applications and are much more popular due to their nature but they possess highly dynamic characteristics where no one has complete control over the network. In the absence of global control of network resources even by the network providers, the assurance of QoS for real-time application is more challenging. 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 efficient service provision by the service providers.

In [2,3] an admission control mechanism is presented for the real-time traffic over different differentiated services domains. A comprehensive comparison for different available admission control mechanisms i.e. traffic descriptor-based admission control (TDAC), measurement-based admission control (MBAC) and end-point admission control (EAC) is presented. The proposed QoS provisioning scheme is

based on the MBAC due to its certain advantages over the other schemes for real-time traffic. The MBAC based mechanism is robust for the traffic heterogeneity.

An admission control algorithm based on the aggregated IP pipes is presented in [4]. The proposed algorithm install QoS enabled aggregated pipes in a multi-domain environment and takes into consideration the new service requests along with the actual utilization of the pipes for different services. The proposed admission control mechanism is more flexible and provides end-to-end QoS provision for designated services. The aggregated pipes are installed after the pSLA/pSLS negotiations between service providers.

In this paper, we intend to provide an admission control mechanism for the multimedia streaming and video on demand services over P2P networks. The use of P2P networks in providing these services allows the reduction of the network traffic load on the central servers (e.g. VoD head-end). The advancement in the P4P like framework where network providers have full view of the network resources, we focus to provide service provider-driven streaming mechanism over P2P networks. 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. These services are carried by end-to-end QoS-enabled aggregated pipes installation. To the best of our knowledge, no work exists for the QoS provision mechanism based on the admission control algorithms for the P2P networks.

3 QoS Provisioning for Multimedia Streaming over P2P Networks

P2P networks exhibit highly dynamic characteristics where network conditions change more often. The real-time application like multimedia streaming and video on demand services are highly stringent against such conditions and require certain QoS mechanism to ensure smooth content delivery. 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 CC seek to receive multimedia content from multiple CPs having distinct characteristics . This admission control mechanism is used to control the amount of multimedia traffic injected in each intra-domain and inter-domain to ensure the QoS. There are numerous well known techniques available to ensure QoS using admission control. These techniques include End-Point Admission Control (EPAC), Traffic Descriptor-based Admission Control (TDAC), Measurement-based Admission Control (MBAC), and Probe-based Admission Control (PBAC). Each of these techniques has certain advantages and disadvantages for the QoS provisioning [2,3,5].

We describe in section 1 that due to the difference between uplink and downlink capacities of participating peer, a content consumer select multiple content providers to receive the requested media content. The received media content are received at receiver side in the designated buffers before their actual decoding. A receiver side buffer is designated to select the video streams from different sources. This receiver side scheduler is responsible for key element of our proposed admission control mechanism. Scheduler receives all the information regarding play out time of each video stream of each tier and current buffer states. On the basis of the received

information, scheduler requests/assigns different video tiers to sender peers. In the case of the QoS degradation scheduler determine to apply stream switching mechanism to maintain the smooth delivery of content at receiver end. Scheduler also determines that how the received video packets from all tiers are ordered. The overall mechanism is presented in [5].

In this section, we described different aspects of QoS provision mechanism based on admission control. We present certain network constraints that should be considered for the provision of overall better QoS and smooth video delivery of at-least base video quality, i.e. minimum acceptable level of QoS. As, we propose that scalable video encoding scheme that is more suitable for real time streaming applications over heterogeneous networks like P2P networks. We apply token bucket model for the admission control of incoming traffic (video streams) from different content providers (sender peers). The admission control mechanism is based on the traffic specification (TSPEC) model [9].

TSPEC model is generally applied for QoS negotiations that are based on the pre-defined parameters which are passed to network layer. In such networks QoS is ensured by the network layer by resource allocation for certain applications. In our mechanism, the TSPEC parameters for the P2P streaming applications are negotiated between the sender and receiver peers before starting the actual streaming. TSPEC parameters for real time streaming application include: Peak Data Rate, Mean Data Rate, Maximum Burst Size, worst case delay, average packet size, maximum packet size, maximum packet error rate, and maximum worst-case delay, such information can be part of SLA (service level agreement) between the content consumer (CC) and the service provider (SP).

Although we introduced the traffic admission control mechanism but still incoming video traffic can be affected due to the dynamic nature of P2P systems. Thus, to maintain an acceptable level of QoS throughout the streaming session, we need to monitor incoming video streams constantly at receiver side to ensure the smooth delivery. In the following sub-sections, we describe a brief introduction to scalable video coding and present our proposed admission control mechanism.

3.1. Scalable Video Coding (SVC)

Video encoding schemes play vital role in the QoS provision in multimedia streaming application over heterogeneous networks. Many video encoding schemes have been proposed for real-time applications operating for heterogeneous networks and terminals to meet the requirements of heterogeneity. Layered encoding and multiple description coding [7] are considered suitable for many applications including P2P services. At present, Scalable Video Coding (SVC) is considered most promising video encoding format for streaming application in heterogeneous networks and terminals [8]. SVC video stream can be truncated to produce videos of different qualities, resolutions, and frame rates using respectively SNR (signal-to-noise ratio), spatial, and temporal scalabilities.

In this paper, we use SVC video encoding scheme to encode multimedia content. Content consumer receives different SVC based video streams that are combined later to decode with the higher quality. In SVC video coding, bit-streams are decodable at

different bit-rates, spatial and temporal resolutions. These SVC characteristics make it more suitable for heterogeneous environments especially P2P networks. For the admission control mechanism, the aggregate pipes are allocated on the basis of each video stream's TSPEC.

3.2. Admission Control Mechanism

The data flow model is an important aspect in the admission control mechanism. Our targeted streaming mechanism is receiver centric in the sense where the content consumer (receiver peer) requests media content from the service provider. On the other hand the QoS provision mechanism is service provider-driven where SP has a global view of available network resources. SP sends a list of content providers (sender peer) and CC selects a number of CPs on the basis of video characteristics. These parameters are exchanged between CC and SP and SP QoS-enabled aggregate pipes for the content delivery. CC selects these CPs on the basis of certain selection criteria and video packets are received from these CPs on the basis of relative scheduling of video packets (video quality tiers) according to their priorities, transmission time, actual playback time and coordinating among the sender and receiver peers. The scheduler determines two sets of parameters for each video packet; the first one is for transmission of video packet from sender peer on the basis of relative schedule and priority and the second one for the receiving of respective packet at receiver side. These set of parameters are used to determine the viability of the video packets before putting them in the receiver side buffer, i.e. if certain video packets are received after the respective playback dead line, these packets are discarded. We represented these set of parameters by $P_k^{schedule}$ and $P_k^{receive}$, where

$$P_k^{schedule} = \{s_k, p_k^{playback}, p_k^{transmit}\} \text{ and } P_k^{receive} = \{s_k, p_k^{playback}, p_k^{receive}\}.$$

Here, $p_k^{playback}$, $p_k^{transmit}$, and $p_k^{receive}$ represent relative playback time, transmission time, and receiving time of packet 'k' at the receiver end. This information is used for the transmission schedule of video packets and also used to determine the viability of received video packet at receiver side before putting into buffer. The scheduler ensures that each video packet arrives in the receiver side buffer on time to avoid buffers under flows problems and that is available before starting the decoding phase.

Sender peers who are providing the different video quality tiers communicate the 'TSPEC' for the respective video tiers with the receiver peer. The receiver peer selects a set of sending peers on the basis of offered video quality tier and relative 'TSPEC' parameters. These parameters consist of Peak Data Rate (PDR), Average Data Rate (ADR), Committed Data Rate (CDR), Peak Burst Size (PBS), Average Packet Size (X_{avg}), Maximum Packet Size (X_{max}), Average/Mean Data Rate (R_{avg}) etc. On the basis of the TSPEC specification the token bucket model is applied for the video streams to control the arrival process. The receiver peer receives video tiers from multiple senders and an independent token bucket is assigned for

each sender peer. The token bucket process is applied to control the incoming and outgoing data flow as shown in Figure 2.

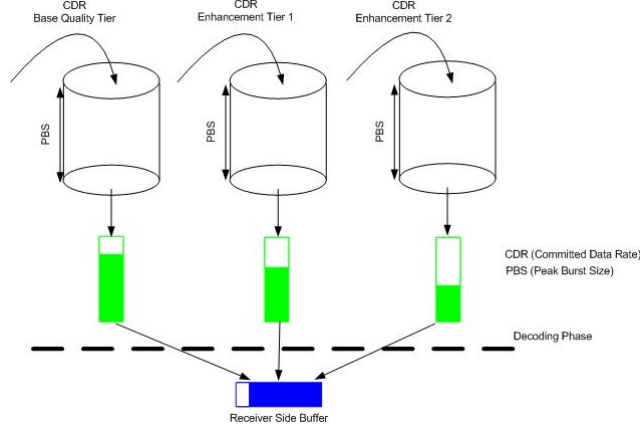


Fig. 2. Token Bucket Process for Admission Control

Now if w_i represents the active windows size advertised by the receiver for the transmission of video block 'i' and $W_i^{playback}$ represents the window size for all the packets: $\forall k, 0 \leq w_k \leq W_k^{playback}$, and $R.w_k \leq R_{max}$, where $i = 1, 2, \dots, k$ and R represents the data rate that is

$$R = \frac{\sum x_i}{(t_i + 1) - (t_i)}, \text{ that represents the number of packets received in one time interval.}$$

While assigning different video layers to sender peers, the receiver peer has to consider the following constraints to support the token bucket processes. An independent token bucket is assigned for each sender peer as showed in Fig. 2 video packets of different layered quality are put into the Video packets from different selected quality tiers are admitted according to the token bucket process at a constant rate. After the draining phase, video packets from all buckets are put into respective buffers separately designated to receive packets from selected quality tiers. We describe below certain constraints that should be satisfied for the overall end-to-end QoS provision for media streaming.

For any k where $i = 1, 2, \dots, k$, $\sum_{i=1}^k p_i^{transmit} \leq w_i$:

$$\sum_{i=1}^k \frac{S_i}{t_i} \leq PDR \sum_{k=d_1}^d p_i^{transmit} \quad (1)$$

Equation 1 represents the peak data rate constraint that the video packets admitted at any interval of time should be less than the peak data rate of video stream allocated in the token bucket process.

For any k where $k = d_1, d_2, \dots, d_K$, $\sum_{k=d_1}^{d_K} p_k^{transmit} \leq w_k$:

$$0 \leq \sum_{i=1}^K \frac{si}{ti} - R_{avg} \cdot \sum_{i=1}^k p_i^{transmit} \leq PDR \quad (2)$$

Equation 2 represents the constraint for the average data rate of each video stream while equation 3 represents the constraints related to end-to-end delay for the arrival video packets from sender to receiver. This constraint is more important as the real-time applications require having minimum delay and playing a major role in the overall QoS.

If D_{max} represents bounded network delay, then for all $d (1 \leq d \leq D_{playback})$, the delay bound will be:

$$0 \leq \sum_{i=1}^K p_i^{receive} - \sum_{i=1}^k p_i^{transmit} \leq D_{max} \quad (3)$$

where D_{max} can be measured according to the round-trip-time 'RTT' value and the respective transmission delay over the certain networking link.

$D_{max} = (RTT + \Delta) * R$, where R is the actual playing rate of video at receiver side while Δ is the transmission delay for video stream. This constraint is very important in streaming applications to find the deterministic schedules with the acceptable level of QoS.

Since the underlying network topology is based on the P2P architecture, there exist several intermediate nodes facilitating the content delivery along the end-to-end path between SP and CC. Thus, it requires the SLA/SLS negotiations between SPs and NPs based on the available network links capacities over the e2e path. We propose to use a cascade model [2] for the QoS negotiations as shown in Fig. 3. Here, e2e pSLA is determined on the basis of individual pSLA for all entities present along the path.

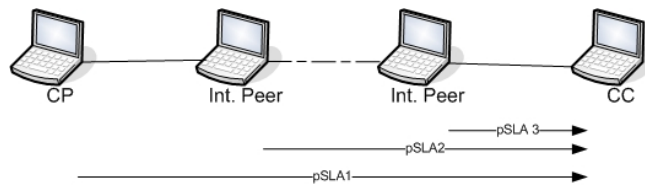


Fig. 3. A cascade model for QoS negotiations

The overall traffic descriptor-based admission control mechanism is described as under:

1. Media Content request initiation by CC
2. pSLA/pSLS negotiation between SPs and NPs
3. Traffic forecast matrix update based on TSPEC of CC's request and pSLA/pSLS agreement
4. SP send a list of CPs to CC
5. Selection of appropriate CPs
6. QoS enabled aggregate pipes installation over e2e paths after consideration of cascade model and constraints present in equations 1, 2, and 3.
7. Updating traffic forecast matrix by SP.

4 Performance Evaluation

This section describes some of the simulation results for the proposed streaming mechanism based on the proposed admission control mechanism. The simulations are performed using the ns-2 simulator [11]. To evaluate the proposed admission control mechanism, we use the SVC based video encoding for the MPEG-4 video trace file 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. We distribute SVC based tiers equally among different sender peers (content providers) to examine the performance of the proposed admission control based QoS mechanism. We perform simulations for two scenarios as follows:

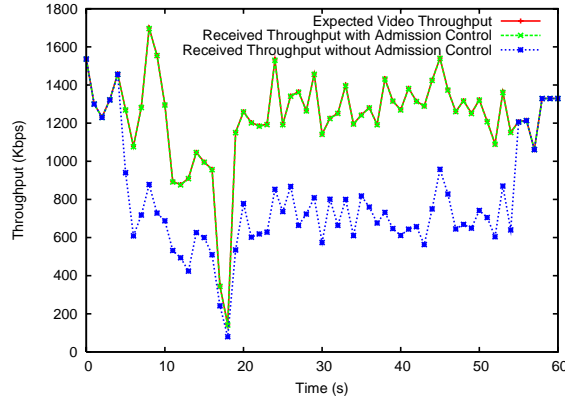
- **Scenario with Admission Control based Quality Adaptation:** Simulation without applying any quality adaptation mechanism. In this case, P2P system works as in downloading modes. The sender peers are selected on a random basis.
- **Scenario without Quality Adaptation:** Simulation with quality adaptation mechanism is performed by best peer selection and their switching based on the proposed hybrid overlay networks as described in section 4.

We focus to many-to-one streaming scenario where a single receiver (content consumer) intends to receive the real-time video packets from multiple sender peers. The content consumer requests the media 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 pSLA/pSLS negotiations. For the scenario with admission control based quality adaptation, the actual content delivery is carried out by allocation of end-to-end aggregate pipes on the basis of TSPEC (traffic specification) and no other traffic is allowed to pass through this dedicated path. 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 presented in equation 1, 2, and 3. Table 1 presents the peak and average data rates of video tiers used for the simulations.

Table 1. Peak and Average Data Rates of Video (kbps)

	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

Fig. 4 presents the received video throughput in both scenarios along with the expected video quality. Fig. 5 present the packet drops ratio for the received video at receiver end in both scenario. We notice huge packets 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 end-to-end 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. Fig. 6, Fig. 7 and Fig. 8 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, lower loss that improves the overall QoS for the received video.

**Fig. 4.** Received Video Throughput**Table 2.** Received Video Quality (%)

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

Table 2 presents a brief summary of simulations results for both received video quality in both simulation scenarios. We notice a clear difference between the received video qualities that is almost 100% when content delivery is supported by proposed admission control mechanism along with dedicated paths, while in the other scenario video content are delivered over shared network paths. The other traffic passing over the same networking link result into huge packet drop ratio and affect the received video quality drastically that is not acceptable for real-time content delivery.

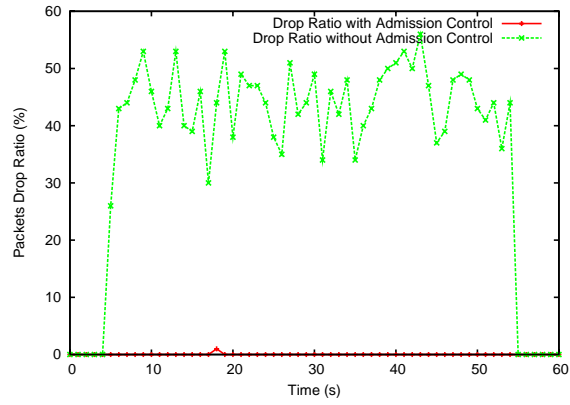


Fig. 5. Packets Drop Ratio

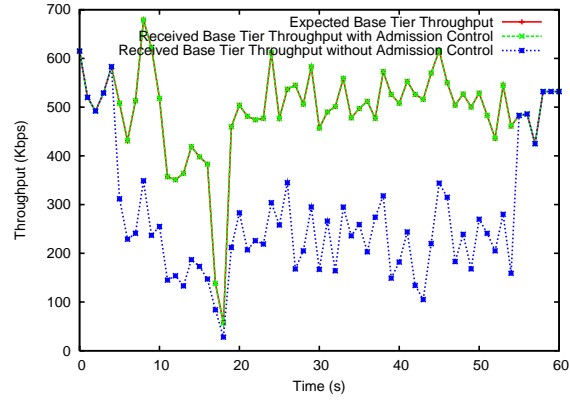


Fig. 6. Received Video Throughput for Base Tier

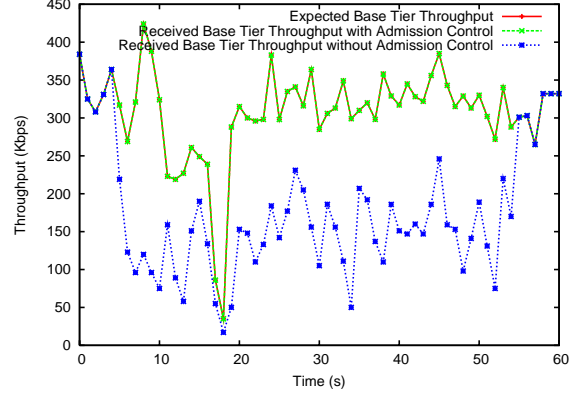


Fig. 7. Received Video Throughput for Enhancement Tier 1

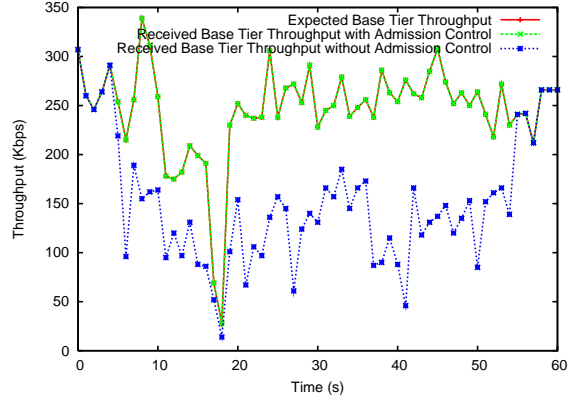


Fig. 8. Received Video Throughput for Enhancement Tier 2

5 Conclusion and Future Perspective

In this paper, we presented an admission control mechanism for the real-time multimedia streaming and video on demand services over SP-driven P2P environment. The admission control mechanism is based on the traffic descriptor admission control technique where the multimedia contents are delivered by enabling the dedicated aggregate pipes over the end-to-end delivery paths. We observe a remarkable improvement in QoS for the received video in the scenario when our quality adapted mechanism is applied. In future, we aim to extend this mechanism to the QoS provision for P2P based IPTV services delivery.

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