An Autonomous QoE-driven Network Management Framework

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Abstract

Recently, network researchers have taken a great interest in quality of experience (QoE) and in the new aspects it brings in the study of the link between network conditions and user satisfaction. Also, the realization that the information of users' satisfaction can be directly applied in the network management in a real-time manner has resulted in a fair amount of publications. Although the systems and frameworks presented in these publications tackle the subject of QoE-driven management quite successfully, they often concentrate on certain applications or technologies. We present a generic QoE management framework, which is applicable to a broad range of systems. We also demonstrate an instantiation of this framework as a network access point management system for RTP-based video. This system is not only able to positively affect the perceived quality of the multimedia application considered, but also to reduce over-prioritization and optimize resource usage.

Keywords: Quality of experience, multimedia, over-the-top, management system, access point

1. Introduction

Multimedia services, and in particular video, make up a large and ever-increasing portion of the total Internet traffic. The popularity of the so-called "Over-the-Top" (OTT) services, such as YouTube, Netflix, Hulu, and other Web-based video services has exploded and will continue to do so (e.g. with the adoption of WebRTC [1] for real-time browser-based communications), as has their importance to users and business. This fact, coupled with the availability of fast and cheap mobile connectivity and mobile devices capable of displaying high-definition content, poses a serious challenge to mobile operators, as users become accustomed to more resource-demanding services and demand better quality. In contrast to operator-run media services (such as IPTV, mobile TV, or IMS-based ones), where the operator has control over the whole chain of transmission, OTT services come from outside the operator's network, and the operators

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have very little, if any, control over them (some content providers such as Netflix work with operators to provide caching servers within the operator's network, but this is not generally the case with all content providers).

The operator's challenges are thus many; firstly, it has to deal with the increasing demand on their infrastructure, notably so over the last hop (base stations and WiFi hotspots, for example). Secondly, OTT video is most commonly delivered over HTTP, which makes it hard to separate from other web traffic. Finally, from the operator's point of view, OTT services are hard to monetize (the content providers get the revenue and the operator just sees an increased use of resources) and at the same time, if the quality of these services is not good enough for the users, the operator will face higher user churn.

In this paper we address these challenges by providing traffic control mechanisms based on a combination of Quality of Experience (QoE)[1] estimations, and subscriber and application-based traffic differentiation. We present a framework to instrument QoE-driven network management mechanisms, and in the context of this framework, we implement a concrete prototype for QoE-driven control. We expand upon our previous work [2] by incorporating network performance models, allowing the proposed approach to always make the right decision by predicting the possible outcomes, instead of just reacting to a drop in quality and hoping that the reaction will result in a positive change. The goal of the proposed work is to allow operators to properly address the needs of their users (in terms of QoE), while introducing subscriber differentiation as a means of increasing revenues and simplifying resource allocation (i.e. customers who pay more are prioritary). The proposed approach is able to a) identify the relevant media flows, b) estimate their current QoE, c) select the appropriate priority for the flows based on their application type, subscriber class, current QoE for it and other media flows, and expected QoE after the control mechanism kicks in (based on network performance models) and d) perform access control on new flows based on the current quality for existing flows, and the incoming flow's application and subscriber class.

The rest of the paper is organized as follows. Section 2 provides an overview of related works. Section 3 describes the proposed traffic management system. Sections 4 and 5 present a prototype implementation of the system and a testbed where the prototype was tested in. Section 6 presents its performance results. Finally, Section 7 presents our conclusions and future research lines associated with this work.

2. Related Work

Although the majority of the research papers concerning QoE are related to QoE assessment and monitoring [3–6], QoE management has recently gained more attention from the research community. Thus, different QoE management systems and frameworks for different network technologies and applications may be found in the literature.

In [2] we proposed a simpler version of the approach proposed herein, whereby control decisions were made based on current quality estimates. The main limitation of this approach (and motivation for this work) is the impossibility of making optimal decisions without a predictive performance model. In the general case, the mechanism proposed

[1] More specifically, perceived quality.

did result in an overall improvement in quality, but when load conditions become more serious, the decisions made could potentially result in worse quality for all users.

Kafetzakis et al. [7] proposed the QoE4CLOUD, a QoE-driven multidimensional framework for cloud environments, while Hoßfeld et al. [8] discussed challenges in QoE management for cloud applications. Gómez et al. [9] proposed an architecture that enables QoE-management in Long-Term Evolution (LTE) and LTE-Advanced (LTE-A) networks. Fajardo et al. [10] proposed a QoE-driven management system for VoIP over 3G UMTS services. However, in the forthcoming era of seamless mobility, solutions either for heterogeneous networks or independent from the network technology are needed.

Mu et al. [11] proposed a QoE management framework for end-to-end multimedia applications called Quality of Experience (QoE)-aware Real-time Multimedia Management (QoE2M). The proposed framework is based on a combined control of video assessment, Quality of Service (QoS) and QoE-based mapping and adaptation procedures. In QoE2M, congestion periods are detected through resource allocation controllers that gather information about the network status. Therefore, the QoE2M is able to adapt applications according to network conditions or user's device capabilities. However, despite this work proposing a dynamic approach and considering some QoE characteristics of the application, it does not discuss implementation or validation issues [12].

Vakili and Grégoire [13] proposed a QoE management framework for video coded with H264. The authors use subjective tests for measuring the level of video quality perceived by end users to investigate the effect of different factors such as frame rate and quantization (QP) on video data bit rate and perceived video quality. Based on the results obtained, the optimum video parameters are set given the network bandwidth and acceptable QoE level. They also introduce a straightforward technique to measure the minimum intermediate hop's bandwidth in each path while sending video data. Based on this estimation, the QoE control function will decide on the QP and the frame rate by which the video data should be sent to the other party. However, the applicability of this framework is limited due to the restrictions it sets on the type and size of videos (i.e. it supports only Quarter Common Intermediate Format (QCIF) resolution and medium motion videos).

Kim et al. [14] proposed a new QoE management framework, called "in-service feedback QoE framework". In this framework the end users give feedback instantaneously whenever service dissatisfaction occurs. By using the gathered feedback information from distributed end users, network parameters from routers, and application information from servers can be analyzed collectively to find out the reason and location of faults. However, this work lacks implementation and validation details.

Agboma and Liotta [15] presented a QoE-based management framework for the construction of QoE models for different types of multimedia contents delivered onto typical mobile devices. More specifically, the proposed framework is based on a statistical method which correlates QoS parameters directly with estimates of perceived quality and identifies the degree of influence of each QoS parameters on the user perception. By using this information the thresholds at which the user's perception of the service quality becomes unacceptable are defined. However, the authors consider only how to maximize end-user quality and they do not consider the impact of the network or how to optimize the usage of network resources [16].

Lloret et al. [17] proposed a QoE management system for IPTV networks that is designed to be placed between the Internet Service Provider (ISP) architecture and the consumer's network. More specifically, the system is based on something called "QoE process" which gathers information from both architectures. By using this information the QoE process performs the appropriate tasks in the network to improve the QoE parameter and vary some features of the IP network to avoid network congestion and, therefore, packet losses. However, the admission control mechanism aims to provide a correct IPTV service to the end users for every channel without considering the impact of network or how to maximize network resources.

Besides the QoE frameworks and architectures, several works proposing mechanisms for efficient QoE management, such as admission control, scheduling, bandwidth management and congestion control may also be found in the literature.

Piamrat et al.[18] proposed a QoE-driven admission control mechanism for IEEE wireless networks. In this research work the access point computes an average Mean Opinion Score (MOS) of all ongoing connections. If the MOS is higher than an acceptable level plus a threshold, then a new connection will be accepted; otherwise it will be rejected.

Gallo et al. [19] presented an ontology for a QoE-framework. The proposed ontology involves the user, application and network QoS aspects. Experimental work showed that semantic approach succeeds in the automatic QoE/QoS mapping and could be beneficial for QoE-management.

In addition, in order to provide the maximum perceived quality to the user, several adaptation techniques at the application layer have been proposed that automatically throttle the video quality to match the available resources in terms of bandwidth, CPU, etc. These techniques can be further classified into transcoded-based, scalable encoding-based and stream switching techniques [20]. Transcoded-based techniques imply the use of a transcoder for assuring guaranteed QoE to the users [21, 22]. However, the main disadvantage of this approach is that it requires increased processing resources and provides limited scalability, since the transcoding has to be done on a per-client basis [20].

In scalable encoding-based techniques, an encoded video is created by composing multiples sub-streams derived from the original video [23, 24]. The main disadvantages of this approach are [20]:

- It requires specialized servers implementing the adaptation logic.
- It is difficult to implement since it requires to be run on specialized servers and content cannot be cached in standard proxies.
- Since the adaptation logic depends on the codec employed, the content provider is limited to using only a limited set of codecs.

Stream switching techniques encode the raw video content in multiple streams with different quality. However, despite this approach does not rely on particular functionalities of the codec employed, it has the cost of increased storage requirements and that adaptation is characterized by a coarser granularity since video bitrates can only belong to a discrete set of levels [20].

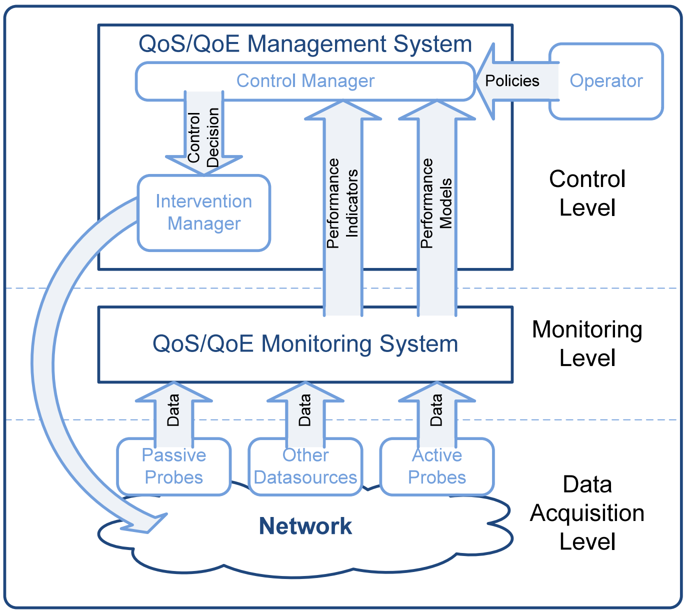


Figure 1: Customer Experience Management System Framework.

3. QoS/QoE Management System

The QoS/QoE management system presented in this paper is a part of a complete customer experience management system (CEMS) framework, which it is illustrated in Figure 1. The CEMS framework contains three layers: data acquisition, monitoring, and control. In order to fully define the QoS/QoE management system, the presence of the data acquisition and monitoring levels is essential. However, they are only briefly described in this paper, as the focus is on the control layer.

All raw data collection occurs on the data acquisition layer by probes or other means of data collection. Passive probes are non-intrusive network monitoring agents, which collect data without causing additional traffic overhead on the network, and usually being transparent to the other components of the network. Active probes, on the other hand, send dedicated measurement packets to benchmark the network performance. A measurement layout usually consist of two or more probes performing independent measurements and possibly exchanging information among themselves. However, even single-point measurement layouts are possible; this usually results in a smaller set of measurable parameters (e.g. accurately estimating one-way delay with a single measurement point is not feasible in general).

On the monitoring level, the raw data produced by the data acquisition layer is processed into knowledge about the state of the network, which is in turn passed to the control level. The control level performs actions upon the network based on this knowledge. There are three important components on this level: control manager, intervention manager and policies. These components are described in the following subsections.

3.1. Control Manager

This section presents the control manager (CM) component, which is responsible for providing management decisions for the system. The CM is responsible for determining the state of the network and deciding the corrective actions to be performed by the intervention manager. There are three parameters which control the decision process of the control manager: policies, performance indicators and performance models, so the CM can be seen as a relation of the form:

$$\text{CM} :: \text{PerfI} \times \text{Policy} \times \text{PerfM} \rightarrow \text{Decision}$$

Policy incorporates the guides and bounds set by the operator of the network. Without going into formal definitions, we consider policies as a set of network constraints on the possible field of operation. In the case of ISPs, policies are likely to be based on service level agreements policies (SLAP) and subscriber policies. Application policy is used more seldom but it is still important to this framework.

$$\text{Policy} = \text{ApplicP} \mid \text{SubscP} \mid \text{SLAP} \mid [\text{Policy}]$$

Performance indicators measure the current performance of the network. A common set of performance metrics are the quality of service parameters (packet loss, jitter, delay etc.). Other parameters are also discretionary, e.g. connection establishment time, address allocation time and average response times for certain services. Quality of experience indicators, such as mean opinion score (MOS), MOS estimations, peak-to-signal noise ratio, mean-squared error etc. are defined under QoE indicators. Since these options are not mutually exclusive, we define performance indicators to be a set of QoS, QoE and other metrics:

$$\text{PerfI} = \text{QoSI} \mid \text{QoEI} \mid \text{OtherI} \mid [\text{PerfI}]$$

Performance indicators express the current condition of the network, but we are also interested to know how control decisions affect the network QoS-parameters, and how a change in QoS-parameters in turn affects QoE. In the framework presented, these type of mechanisms are called performance models. Performance models can be one or more QoS or QoE models:

$$\text{PerfM} = \text{QoSM} \mid \text{QoEM} \mid [\text{PerfM}]$$

Performance models are a focal part of the control manager when optimizing other parameters, such as costs, resource utilization and power consumption, simultaneously attempting to maximize QoE. These models, along with a possible addition of usage models, can be used to estimate QoE or QoS performance:

$$\text{estimateQoE} = \text{PerfI} \times \text{QoEM} \rightarrow \text{QoEI}$$

$$\text{estimateQoS} = \text{PerfI} \times \text{QoSM} \rightarrow \text{QoSI}$$

The control manager integrates the data provided by the three information sources (i.e. policies, performance models and performance indicators). This data is used in the control decision module to make corrective actions on the network. The actions are executed by so-called intervention manager, which is presented in the next subsection.

Generally speaking, there is no restriction of what actions can or should be taken, but as the framework is designed for network-level operation, we limit the possible actions to the following items:

$$\begin{aligned}\text{Action} \quad = \quad & \text{ProvisionA} \mid \text{ShapingA} \mid \text{MarkingA} \mid \\ & \text{AccessCtrlA} \mid \text{HandoverA} \mid \text{AlertA}\end{aligned}$$

The decisions are, in respective order, dynamic network resource provisioning, packet or traffic shaping, packet marking for later control, per-client or per-connection access control, horizontal and/or vertical handover and alerting the network operator. These control mechanisms are also discussed in the next subsection.

The last element required for the control manager is the decision, which is simply a list of actions:

$$\text{Decision} = [\text{Action}]$$

In simple cases, the decision can be a single action. In the prototype presented in Section 4, however, we combine several actions in the decision process to deal with different scenarios and to mitigate quality degradations.

3.2. QoE Intervention Manager

The intervention manager is a physical device on the network which incorporates the necessary tools to actualize the decisions set by the control manager. This component may act upon all traffic traversing the point of control, traffic flows of certain users or applications, or individual packets, depending on the given task. The remainder of this subsection describes the actions defined in (3.1) in depth.

3.2.1. Dynamic Network Resource Provisioning

Dynamic network resource provisioning is the act of augmenting or re-distributing network resources, either adaptively during congestion or statically for certain services or users. These mechanisms can be, for instance, traffic re-routing or bandwidth re-allocation.

3.2.2. Shaping

Traffic shaping enforces certain performance bounds on different types of traffic. The focus of this mechanism is to limit the throughput of certain users or services, e.g. peer-2-peer traffic. The same method can also be used to throttle down other traffic in order to give better performance to certain users or services. In other words, shaping allows the establishment of traffic classes with different priorities. Also, an important aspect in traffic shaping is that if certain traffic is restricted to enter the outgoing link, the packets of that traffic are queued instead of just being dropped, which results in an increase in delay instead of in packet losses (unless a buffer overflow occurs).

3.2.3. Access Control

If the control manager detects that the network doesn't have enough capacity to meet any new resource requirements, access control (also known as admission control) is one way to prevent traffic from congesting the network. Access control can be implemented on the user level by preventing users from accessing the network, which is common in cellular and ATM (asynchronous transfer mode) networks. Another option is to allow all users to access the network but terminate new traffic flows if these flows would deteriorate the quality of existing ones. This model is more suitable in non- service-specific packet-switched networks, as the behaviour of users cannot be predicted beforehand: users may consume a large portion of the network capacity and use very quality-strict applications as well as simply idle in the network. Access control can be used in conjunction with other mechanisms to gain flexibility to the management system.

3.2.4. Handover

If a section of the network edge becomes overloaded, new or existing connections to the access point can be handed to other access interface (this procedure is called a “handover”). Handovers may be distinguished into horizontal and vertical, depending into whether a handover occurs between a single type of network interface (e.g. between two WiFi-hotspots) or a variety of different network technologies (e.g. from 4G to 3G) [25]. In this context the role of the QoE control manager is to make the decision of the best network selection during a vertical handover. To achieve this goal different vertical handover decision algorithms may be applied.

3.2.5. Marking and Alerting

Two actions which do not have a concrete impact on the network by themselves are packet marking and alerting. Alerting simply means that the system notifies the network operator under specific conditions, for example if the network or a part of it becomes heavily encumbered. Marking is the act of modifying packets by adding a specific mark in a protocol header. On the network layer, the differentiated service code point (DSCP) field is a good candidate for marking. There are also some technology-specific fields, like 802.11p or 3GPP QoS classes for mobile traffic. Other network components outside the system can be given instructions on how to deal with different markings. Naturally marking and alerting are not effectively actions (as defined above) by themselves, but they may lead to procedures outside of the system, so from this perspective they can be considered as such.

4. Prototype Implementation

Following the conceptual framework presented in Section 3, we developed an autonomous network access point management software, which is able to monitor and manage traffic traversing from the Internet to end users’ devices. The system can be placed at the edges of a 3G or 4G network before the radio channel, but because the system operates on the network layer and above, the underlying technology is not restricted in other way than requiring it to have an IP layer. This solution demonstrates some features of the framework, and gives a concrete example on how it can be brought into a concrete application.

An outline of the software design and its components are given in Figure 2, which follows the same layering principle as in Figure 1. The bottom level, network path, contains the physical path from the rest of the ISP’s network to the transfer media. The Control Manager and associated types described in Section 3.1 can be found in the Control and Monitoring levels in Figure 2. Clients connect to the Internet through this

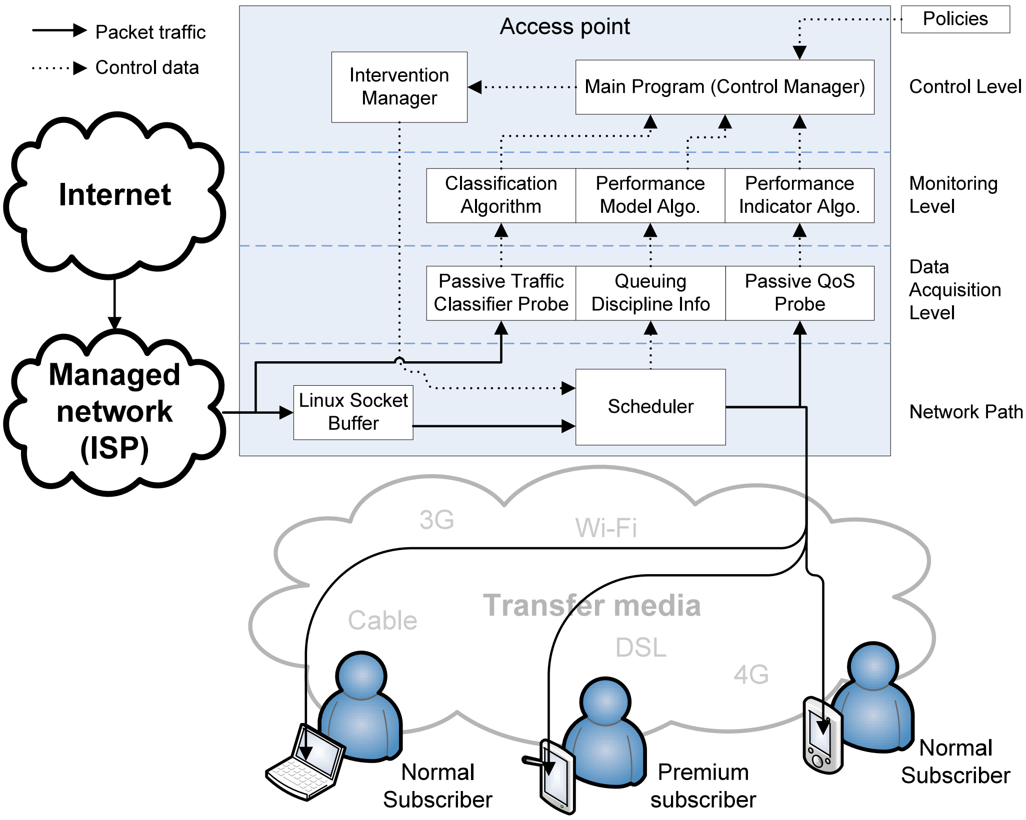


Figure 2: An overview of the access point software.

path using various devices. This path is marked with solid arrows (packet traffic). Other signaling between different software components are marked with dashed arrows (control data). The goal is to monitor and manage the downlink connections of the clients using several tools and methods in the access point. To improve scalability, the access point system operates almost fully autonomously: access points can be deployed and suspended independently.

There are two components in the network path: the Linux socket buffer and scheduler. The Linux socket buffer is a double-linked list which contains the packets traversing from the network towards clients' devices. This is the default network queuing component in Linux operating systems, both in Desktop computers and embedded systems (e.g. routers). Then, we use a queuing discipline to implement our traffic control. Although a queuing discipline resides on more than one level, the scheduler, which ultimately decides the dequeuing order of packets, can be placed in the network path. This is the component which physically realizes the traffic control.

The scheduler used in this implementation is a part of the hierarchical token bucket (HTB) queueing discipline [26]. This discipline allows us to form traffic classes with different real-time constraints. Also, this is the tool that the intervention manager uses to realize the control decisions. The use of this tool in this work is described further in the next subsection.

The next two levels in the system are the data acquisition and monitoring levels. All probes and sources which collect raw data are located in data acquisition level, and the components which refine the produced data into knowledge are located on the monitoring level. In this implementation, there are three data sources: a QoS-measurement probe, a traffic classification probe and a queuing discipline information collector. First, all incoming traffic is mirrored to the classifier probe, which sorts packets into flows identified by source and destination IP addresses, ports and transport layer protocol. The probe also collects flow-based statistical information, which is passed to the classification algorithm on the next level. The task of this algorithm is to classify each flow based on the acquired statistical information. The passive QoS-probe, on the other hand, is placed on the egress point of the AP after the scheduler. This probe collects a few QoS-parameters for quality assessment on the next level, namely packet loss rate and packet loss burst size. This information is used by the quality assessment component to produce a quality estimation for each relevant flow in real-time. The last component in the data acquisition layer is queuing discipline info collector, which retrieves the queuing discipline class information for performance model algorithms. The performance model algorithms produce a quality estimate, which helps in the decision process to determine the best course of action.

Starting from the general definition of the control manager defined in (3.1), the management decisions are governed by policies, performance indicators and performance models:

$$\mathrm{CM_{ap}} :: \mathrm{PerfI_{ap}} \times \mathrm{PerfM_{ap}} \times \mathrm{Policy_{ap}} \rightarrow \mathrm{Decision_{ap}}$$

The information provided by the classification algorithm is also required by the control manager, since it contains the information of each traffic flow. The following subsections present the four elements in detail.

4.1. Policy

The policy of the designed system consists of application policies and subscriber policies. Applications are divided into three categories based on their real-time constraints: interactive, streaming and bulk. Interactive traffic has high two-way real-time constraints in bandwidth, jitter, packet loss and delay. This type of traffic is generated by, for instance, VoIP-calls, videoconferencing and online games. The second category in priority, streaming, incorporates one-way streaming applications, which are intolerant to packet loss, low bandwidth and jitter. Applications such as IPTV and video-on- demand services fall in this category. It should be noted though, that applications in the same category may feature different behaviour during quality degradations. For example, a packet loss in an RTP video stream causes visual quality degradations, but a packet loss in an HTTP-based streaming causes TCP retransmission to occur, which results in a decrease in goodput, which in turn may manifest itself as stalls and long buffering times on the application layer, or a quality adaptation in the case of dynamic HTTP streaming systems such as DASH. The last category contains all elastic traffic and traffic which goes beyond the interest of the network operator. In this work we focused on over-the-top multimedia, especially in RTP-based video streaming[2].

In addition to application differentiation, we also have an implementation of a customer differentiation scheme. Users are divided into two classes in accordance to their prior subscription agreement with the ISP: premium users and normal users. Premium users have purchased a more expensive subscription type, which provides them a high-end connection type with better service quality. Normal users are customers who rely on the traditional "best effort" service quality. The goal is to make the premium subscription more appealing to "heavy users", who would consume a large portion of ISP's base station capacity. Although we want to place premium users above normal users in service quality, less constrained applications cannot supersede or preempt more constrained ones. Therefore, we settle on the following traffic category (in a descending order in priority):

1. Premium interactive media
2. Normal interactive media
3. Premium streaming media
4. Normal streaming media
5. Premium bulk traffic
6. Normal bulk traffic / Default class

The first four classes are QoE-managed. The lowest class is the default class where all the traffic is initially directed to (with the exception of premium bulk traffic, which is always directed to the second lowest class). It can be seen from the above that premium traffic always gets a higher priority in the same application class, but never over different application classes.

The tools required for implementing our policy scheme are a traffic classifier and a subscriber database. As the former one, we used a machine- earning algorithm called two-phased traffic classification tool. This tool relies on statistical features of packet traffic in the identification process, and it can be trained to classify individual applications or traffic which meets certain statistical criteria. The tool is described in detail in [27]. The subscriber database was left outside of this implementation, and the subscription information was managed by the access point software instead.

4.2. Performance Indicator

Perceived quality can be expressed in several ways. In this work, we use MOS as a way to express the estimated quality quantitatively as the average opinion of a group of users. The scale of MOS is often a number between 1 and 5. For giving a verbal presentation for the scale, we use degradation category rating (DCR) [28]. In this rating, a MOS value of 5 means that impairments in the given multimedia are "imperceptible", while a score of 1 means that the impairments are "very annoying".

In order to form a complete data path from the network to the CM, we need components on data acquisition level and monitoring level. We use the Qosmet tool [29] to perform single-point QoS-measurements in the access point with a passive probe. The

[2] Our scope limitation to RTP stems from the fact that, at the time of writing, we have a suitable quality model for RTP-based video, but not yet for HTTP-based streams. This does not, however, limit the generality of the proposed solution.

probe collects information on the data acquisition level, which is then transferred to the Qosmet controlling entity on the monitoring level.

Although the performance indicator of this system considers only QoE, we are using QoS along with application-level information to estimate the quality of experience in real-time. For this purpose, we use a machine-learning algorithm called pseudo-subjective quality assessment [30, 31]. The algorithm was trained to map a set of QoS and RTP-video stream parameters into a single MOS value. These parameters were packet loss rate, packet loss burst size, video resolution and the amount of movement in the video. Data collected from a subjective video assessment campaign was used to teach the algorithm to map these parameters into a MOS value. We trained and validated the algorithm with 98 training samples and 20 validation samples, and the validation showed that the method correlates well with the subjective data, resulting with a correlation coefficient of 0.91. Further details about the quality model can be found in [32].

4.3. Performance Model

It is the scheduler's responsibility to coordinate the traffic according to the QoE requirements. In our demonstration, we consider 6 different queues (i.e. four for each QoE-managed traffic class, one for premium bulk traffic and one for default traffic class). Since the traffic pattern is bursty, we are interested not only in calculating the packet loss probability but also the mean loss burst size. This information is sufficient in our implementation to estimate of the perceived quality of RTP-based video streams. The difference between a performance model and performance indicator is that the indicator expresses the current flow-based QoE, while the performance model can be used to predict the impact of changing the priority class of a flow. Thus, with the help of the performance model, we can make more intelligent decisions when controlling traffic. It is indeed the addition of the network performance models that ensures that decisions are done correctly, instead of just assuming that changing the class of a stream will result in an improved overall quality, as we did in our previous work [2].

In our prototype implementation we expand upon previous work integrating traditional network performance models with quality estimates [33], by representing the scheduling process as 6 different separate $M/M/1/K_i$ queueing systems, where K_i denotes the storage capacity of each queue. For simplicity, we assume that the arrival rate of each type i follows a Poisson process, denoted as λ_i and the service time of each type i is exponentially distributed with mean $1/\mu_i$. Thus, the load ρ generated by type i is given by

$$\rho_i = \lambda_i/\mu_i. \tag{1}$$

Also, the loss probability p_{L_i} for each traffic type i can be expressed as

$$p_{L_i} = \begin{cases} \frac{\rho_i^K (1-\rho_i)}{1-\rho_i^{(K+1)}} & \rho_i \neq 1 \\ \frac{1}{K+1} & \rho_i = 1 \end{cases} \tag{2}$$

If we assume that a burst of losses for the traffic type i is j_i, then the mean loss burst size for the traffic type mlb_i is equal to

$$mlb_i = \sum_{j=1}^{\infty} \left(\frac{\rho_i}{1+\rho_i} \right)^{j-1} \frac{1}{(1+\rho_i)} = \frac{1}{(1+\rho_i)} \tag{3}$$

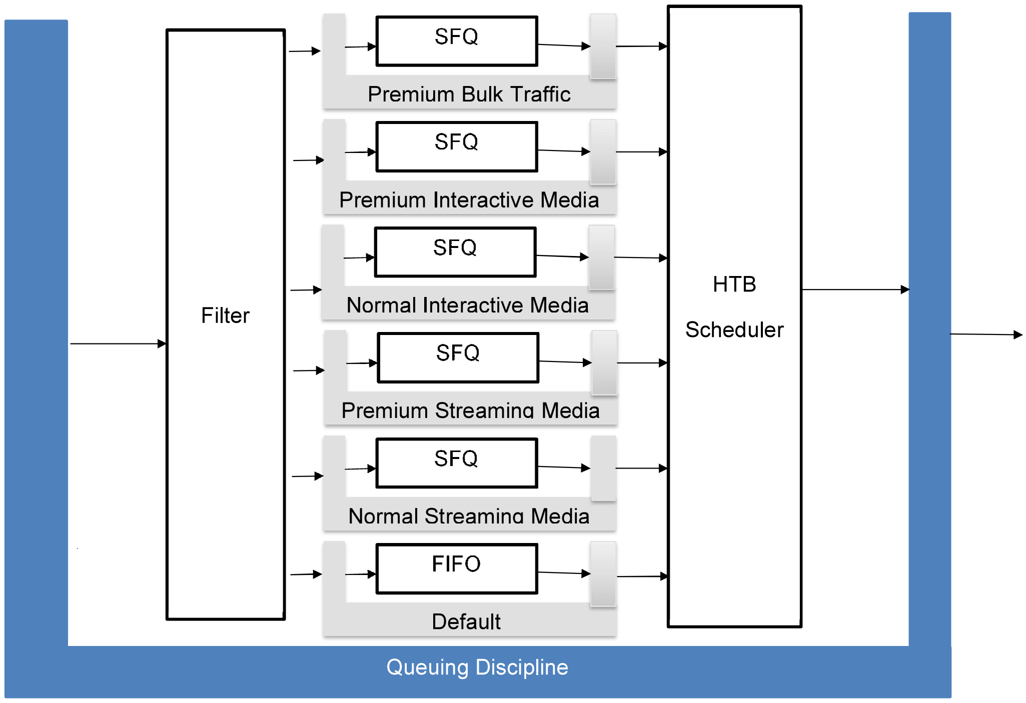


Figure 3: The queueing policy.

While this is a simplification of the way the HTB scheduler works, it is a reasonable one for the short timescales involved in the decision making process.

4.4. Decision

As defined in Section 3.1, a decision is a list of actions. There are three types of control in this implementation: dynamic network provisioning, shaping and access control. Access control may only be performed when a new flow arrives. Also, each traffic class is treated differently from the management perspective, and this is when provisioning and shaping step in. Higher-priority classes gain a larger portion of the AP's capacity, and they are also dequeued with a higher priority. If no flow is promoted to a higher class, the AP operates in a "best-effort" manner. The network resources are managed by hierarchical token bucket algorithm [26], which distributes the available bandwidth to the classes, while keeping utilization ratio high (that is, no bandwidth is ever left unused as long as there are packets waiting for transmission). The shaping action incorporates inter and intra-class scheduling: classes of higher priorities are always dequeued first, as long as they have bandwidth available. Moreover, classes are also scheduled internally by using the stochastic fairness queuing discipline. This queuing mechanism is described in a few online sources (e.g. [34, 35]). The complete queuing system is depicted in Figure 3.

When a new flow attempts to enter the AP, the control manager must make a decision on whether the new flow can be supported or not, and thus if it can be permitted to enter. This decision is based on the application policy and the performance model. The admission control decision tree is depicted in Figure 4.

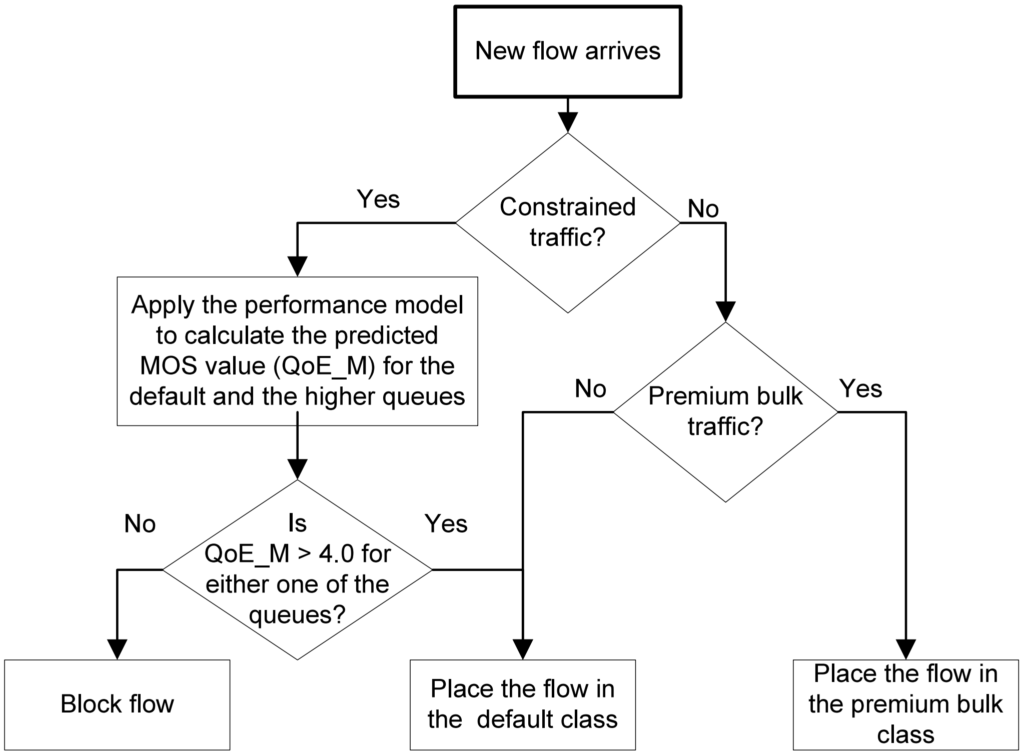


Figure 4: Access control decision process.

First, flows that are identified as bulk traffic are automatically permitted to enter the AP. Since these flows are elastic, their resources can be re- allocated to other flows whenever needed, so there is no reason to reject them. If the bulk traffic belongs to a premium user, the flow is permanently placed in the second lowest class. On the other hand, if a flow belonging to a more constrained application class attempts to enter, the performance model is used to predict the QoE this flow would receive either in the default traffic class, or the higher class respective to the application type of the flow. There are two possible queues for each constrained flow, either the corresponding higher class of the flow (e.g. an interactive application flow generated by a premium user is promoted in the highest class etc.) or the default queue. If it is determined that the quality in either one of these queues would be less than the admission threshold (which in this system was set to 4.0), the flow is rejected. In the other case we can conclude that the quality of the flow can be maintained in a satisfactory level in the system in at least one queue, so the flow is permitted to access.

As stated, premium bulk traffic is placed in the second lowest class, and normal bulk traffic and all permitted constrained traffic is placed in the default class. Bulk traffic flows stay in these classes permanently, but this is not the case with constrained application flows (i.e. streaming or interactive flows). Figure 5 depicts the decision process for constrained flows, which is processed every one second for each flow. First we check which class the constrained flow is currently in. If it is in the default class, it stays there

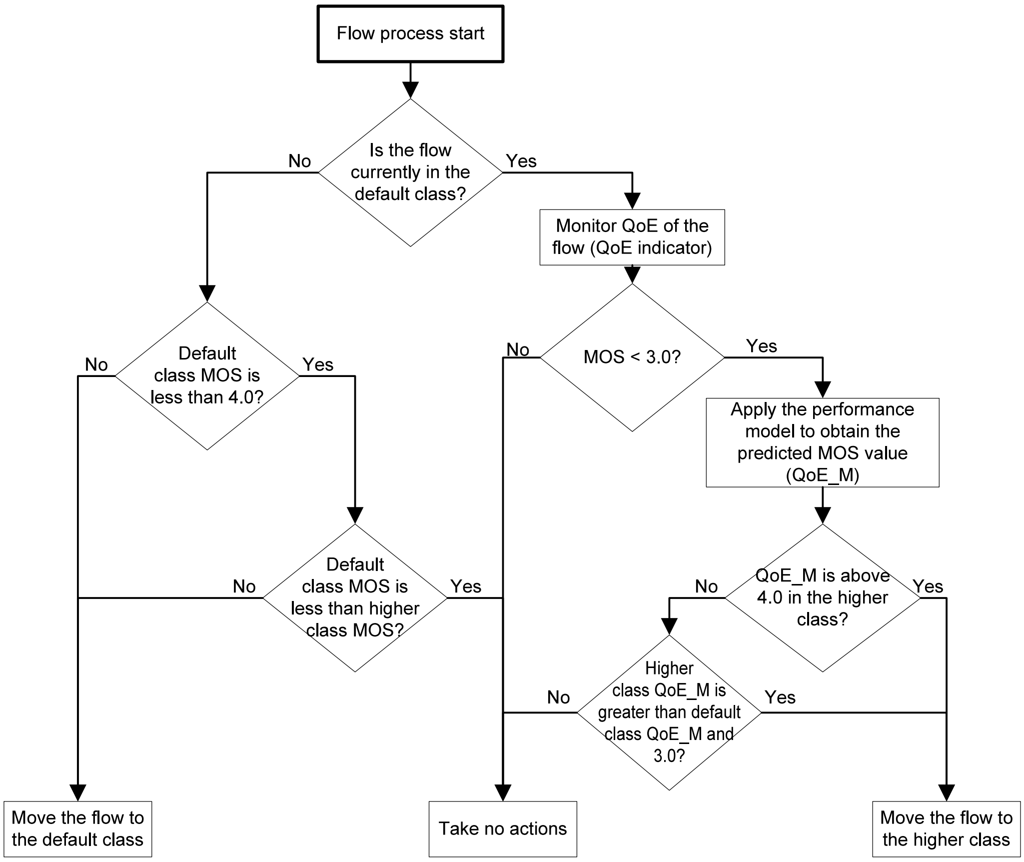


Figure 5: Per-flow QoE decision process.

until the estimated MOS of the flow drops below "bad" threshold (which was set to 3.0 in this demonstration). Now we need to determine if the flow should be moved to the higher class or keep it in the current class. For this task we use again the performance model. The estimated MOS-score is calculated for both queues, and the flow is moved to the higher queue if the predicted MOS value of this queue is greater than "good" threshold (which was set to 4.0). If, however, this is not true, then the system will compare whether the predicted MOS is still above the bad threshold and also above the predicted MOS of the default class. If these conditions are met, then moving the flow to the higher class is still beneficial. If all these tests fail, then the best option is to take no action and keep the flow in the default class.

The process of determining the best course of action for flows in higher classes is somewhat similar. But in order to avoid over-prioritization, which would result in too many flows being promoted to higher classes, we must determine when the conditions in the default class are viable for flow de- promotion. First, the system checks if the predicted MOS of the default class is above the good threshold. If so, the flow can be moved to the default class right away. In the other case, moving the flow back to the

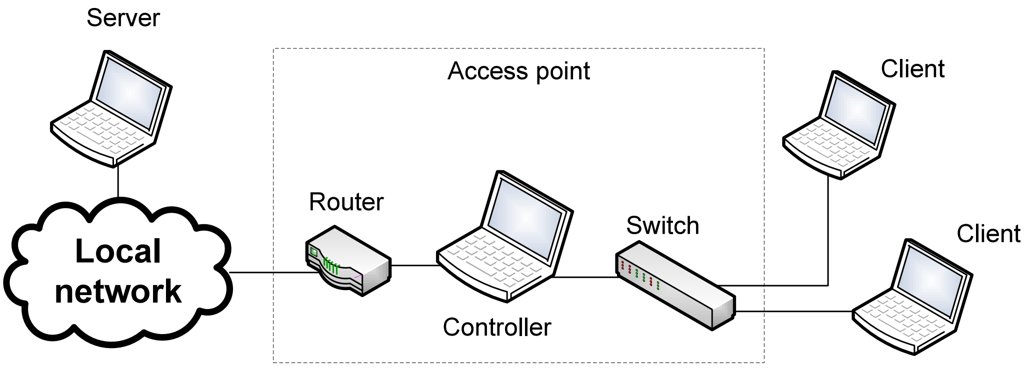


Figure 6: Testbed

default class is still a good option if the predicted MOS of the default class is above the MOS of the higher class. With this decision tree, we can limit the amount of flows in the higher classes so that the quality of these flows remain good. Otherwise flows could overload the higher queues during congestion periods.

5. Evaluation setup

The testbed used in this demonstration is illustrated in Figure 6. We used two client computers to emulate up to 5 (using one end device for each client would have the same result, as the management software differentiates flows and subscription types, not end devices). The access point software was implemented in a laptop (shown in the center of the figure). An external computer was used as a D-ITG and VLC media player server, which were used to transmit bulk traffic and video streams, respectively. The router was used to implement NAT on the subnet, and the switch to allow connecting more than one device on the AP. It should be noted though that the software can be also implemented in a single Linux-enabled router as well.

For all the tests, we used the same reference video, which can be obtained freely from [36]. The video features an MP4-encoded, 1280x546 resolution video stream with 24 frames per second. For emulating TCP bulk traffic, we used a tool called Distributed Internet Traffic Generator (D-ITG) [37], which was set to generate 2048 packets per second, each having a payload of 256 bytes. For emulating packet losses, we used a tool called netem, which is integrated to Linux iproute2 toolset [38] along with the queuing disciplines needed to implement the desired behaviour.

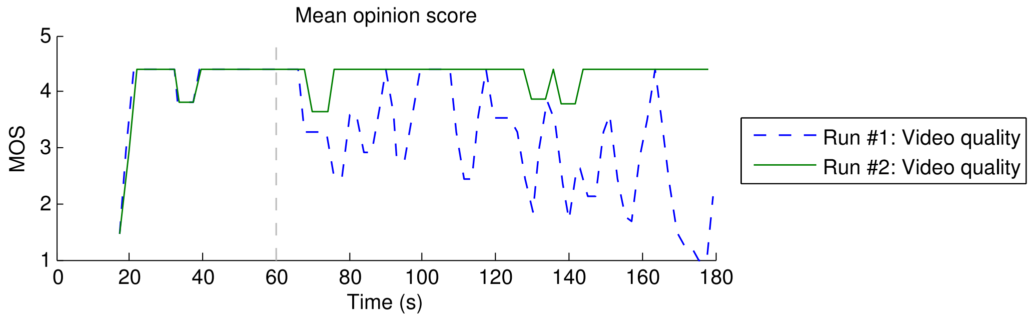
We ran several tests in order to study the developed system. Completing a holistic test for this type of system would be extremely complex and so would be extracting the desired information from it. We therefore chose several simpler scenarios to validate the different aspects of the system's functionality: we tested the differentiation between traffic classes to observe how inter-class prioritization impacts the perceived quality. Then we ran an intra-class test where normal and premium users have to share common resources. We then performed a more complex test where the AP had to resolve the distribution of resources to a mix of different applications and users with different classes. We also tested the admission control algorithm to validate that it didn't allow the AP to become overburdened. Finally, an important thing was to measure how fast the system could react to a quality degradation, as the reaction time is a critical component in user satisfaction. Each test except the performance test was run in pairs; once without the management system (i.e. a simple "best effort" FIFO-system) and once with the management system. These pairs were repeated 10-40 times depending on the test to acquire statistically significant values showing that the system operates as demonstrated in the figures. For each test, we calculated standard deviations of MOS scores to show that the results are consistent and reliable. The deviations are calculated for each video stream as follows: In a test run, samples are taken every second. Each sample is rounded to the nearest second and MOS values are windowed with 4 samples. For each second where there are at least 2 samples a standard deviation is calculated, and all samples are finally averaged to get a mean standard deviation.

6. Results

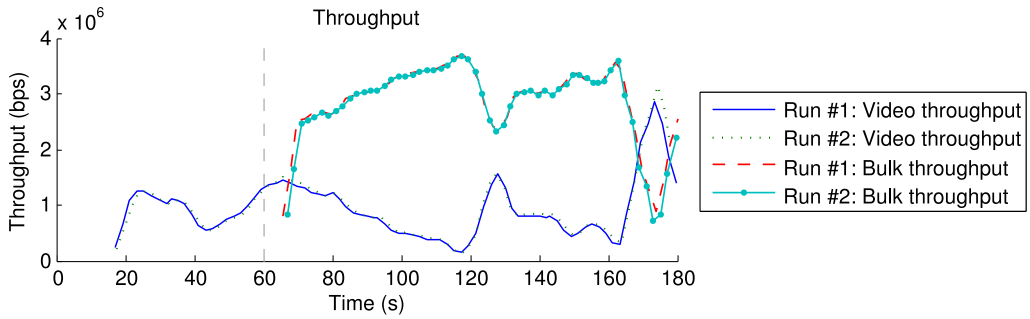
To show the impact of the developed QoE-driven access point management system, we present some testbed results in this section. First, we present two simple test cases where traffic and client differentiation are demonstrated. Then we introduce a performance test which measures the reaction time of the system. In the last two tests we show how the admission control prevents traffic from over-burdening the AP, and how the system is able to balance the load of different traffic types in a more complex scenario.

The first test shows how the AP performs with and without the management system during congestion. Shown in Figure 7, the test was run using an RTP-video stream against background traffic. The RTP-stream was initiated at the beginning of the measurement, and background bulk traffic at 60 seconds marked by the vertical dashed line. The management system is disabled in run 1 and enabled in run 2. Without the management system in run 1, the quality of the RTP stream begins to decline once the background traffic is initiated. Although the background traffic operates on TCP, which features congestion control mechanisms, the adaptation does not facilitate the situation fast enough, and the RTP- stream begins to suffer from packet loss and increased delay. In run 2, the drop in quality is detected by the management system, and the RTP- stream is promoted to a higher class. Now these two traffic types are given different bandwidth and dequeuing priorities, which results in an improved QoE of the RTP-stream. 15 Repetitions of this test showed that the MOS value of the RTP-video increased approximately 0.9 points on average (0.05 standard deviation), and the variance of MOS reduced from 1.92 to 0.21, indicating that the quality didn't only improve but also remained more stable during the test. One thing which should be noted from Figure 7b is that while the quality of the video stream increased significantly, there was hardly any difference in bulk throughput between these two test runs. This is most like due to the fact that in both cases the bulk transfer suffers from packet losses due to rate adaptation, but in the latter case the reaction is faster because of prioritization. It can be noted, especially between 170 and 180 seconds, that the rate of the bulk transfer slows down faster when the management system is on, but the difference is marginal since the RTP video only requires a slightly larger bandwidth to operate with good quality.

Another test was also executed which demonstrates the differentiation between user classes instead of applications. In this test, two RTP-stream were initiated simultane-



(a) MOS value of the RTP video in both runs

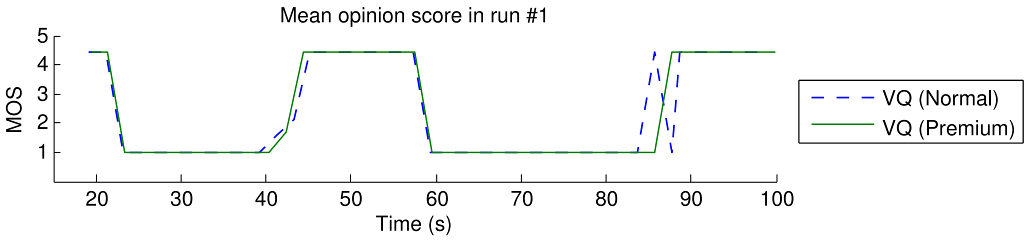


(b) Throughputs of the RTP video and the bulk transfer in both runs

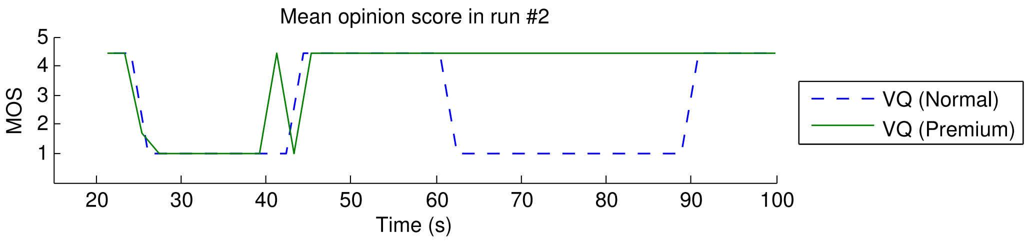
Figure 7: Congestion test results

ously, one belonging to a premium user and the other to a normal user. The results of this test are illustrated in Figure 8. For this test, the access point bandwidth was set to 2048 kbps – enough to support only one RTP-stream, and the test was repeated 15 times, resulting in a standard deviation less than 0.21 for both video streams when the management system is on. The variable bit rate of the video stream causes two congestion events to occur, one at 25–45 seconds and another at 60–90 seconds. Without the management system, both video signals become intolerable in quality during these congestion periods. When the management system is enabled in run 2, the quality drops are detected and the streams are promoted to higher classes. However, although both of the streams have the same application, the premium user gains the preference during the second congestion event. Further 15 repetitions of this test showed that the quality of the premium flow increased on average from 2.41 to 3.81, similarly reducing the MOS variance from 2.83 to 1.68. There are no statistically significant changes in the quality of normal user's flow, which can be also noted from Figure 8a and 8b, although the changes in throughput are significant (Figure 8c and 8d). This can be explained by the simple fact that once the quality drops to 1, the throughput of the flow can be distributed to other flows, as the quality is already intolerable and cannot become worse if more bandwidth is distributed elsewhere.

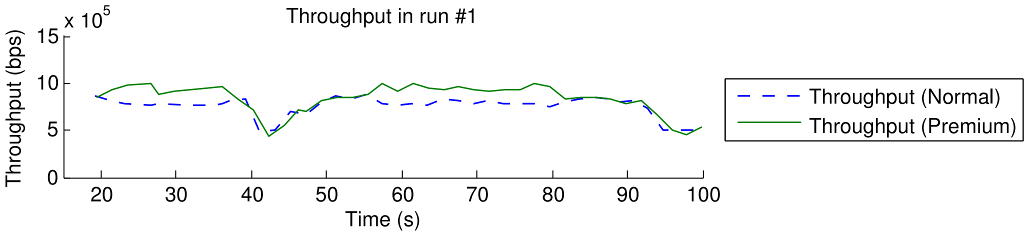
The next test, performance analysis, was executed to study how fast the management system can react to a quality drop. The test was performed using an RTP-video stream and assigning a static packet loss rate to the default bulk traffic class with netem [39].



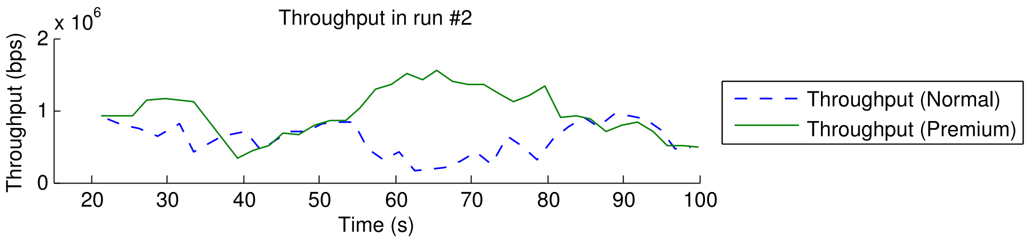
(a) MOS value of the RTP videos in the first run



(b) MOS value of the RTP videos in the second run



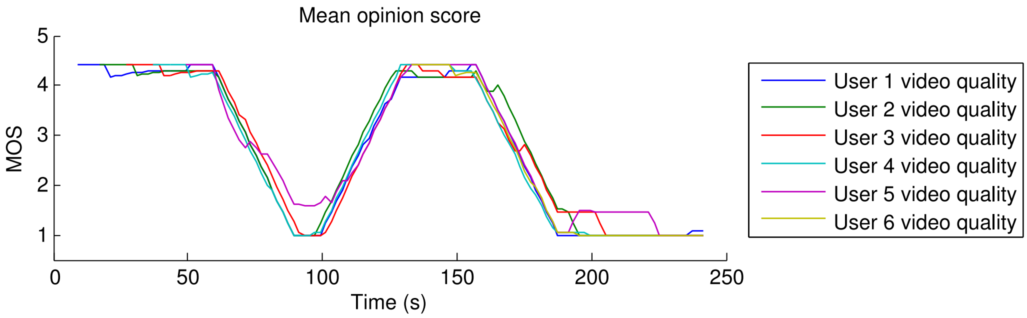
(c) Throughputs of the RTP videos in the first run



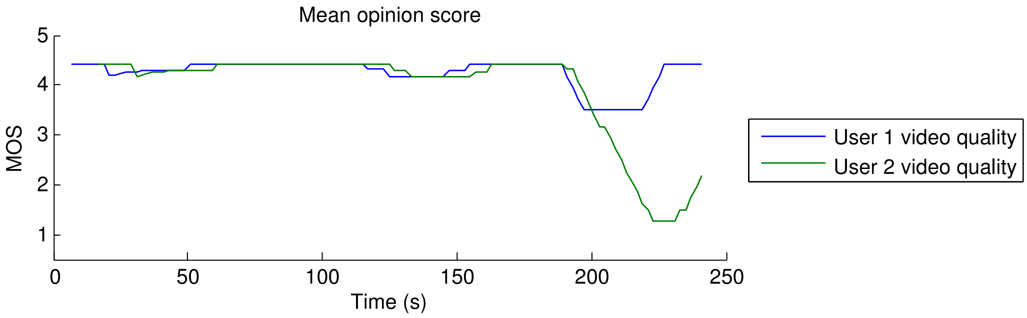
(d) Throughputs of the RTP videos in the second run

Figure 8: Client priority test results

The test was repeated 40 times. It was discovered that the system detects quality deteriorations in 2.8 (with 1.44 seconds of standard deviation) seconds on average and it can restore the quality to an acceptable level (i.e. above 4.0) in 4.0 seconds on average (with 1.78 seconds of standard deviation). But during these tests we realized that the reaction time is much larger in a congestion event that in this test (as one can witness for example in Figure 8b). We studied the issue and found out that when the default class consumes all the bandwidth of the AP, a flow which is promoted to a higher class



(a) MOS value of the videos when admission control is inactive



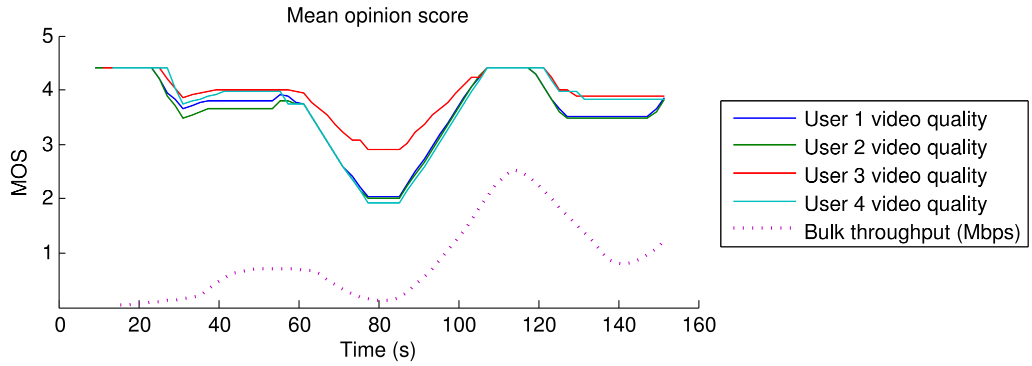
(b) MOS value of the videos when admission control is active

Figure 9: Admission control operation

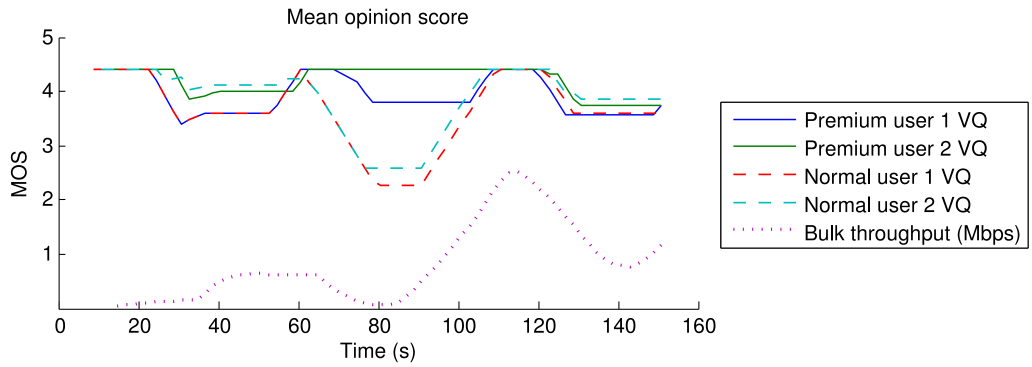
does not instantly receive the bandwidth the class is supposed to gain. This results in heavy packet losses to the promoted flow for approximately 5 seconds, which deteriorates the results somewhat acquired in congestion tests. This issue is related either to HTB or Linux iproute2 implementation, but we are still able to demonstrate the benefits of the system.

Although the AP management system improves the perceived quality of selected applications, the system may fail nevertheless if there is simply not enough capacity to support all the flows. If we know beforehand that a new arriving flow cannot receive a satisfactory quality because of congestion, we can reject the entry of that flow immediately. We have demonstrated how the admission control works in Figure 9. We repeated this test 10 also times to ensure consistency of the results. In 8 runs, the admission control permitted 2 flows to enter the AP, and in 2 runs 3 flows were permitted. When the management system was off, the average MOS score of all video streams was 2.29 with a standard deviation of 0.38, while the respectable values were 4.17 and 0.14 when the management system, and therefore admission control, was on.

Figure 9a depicts the situation where multiple RTP flows attempt to enter the access point in 10-second intervals. Without admission control, these flows begin to suffer from unacceptable quality quite soon, especially during peak bit rates. Figure 9b depicts the same situation when the admission control is on. Only two flows are permitted to enter the AP, as the admission control algorithm calculates that there is enough capacity to maintain a satisfactory quality. However, at 190 seconds the bit rates of the videos



(a) MOS value of the videos when the management system is inactive



(b) MOS value of the videos when the management system is active

Figure 10: System demonstration with both types of clients

increase significantly, and the management system has to step in and differentiate the flows.

In order to use performance models in quality prediction, we need to know how much capacity the new arriving flow would consume. This is difficult to predict beforehand, but there are several ways to make a good guess. One possibility is to perform DPI on packets sent during the handshake procedure to see what kind of data the flow will contain. If the data is unavailable then one can make assumptions based on other information, such as the URL field or the history of the client. We predicted the flow throughput based on the average throughput of flows already in the AP, assuming that a new RTP flow will consume approximately the same amount of throughput as all the other flows on average.

Finally, we introduce a test where both traffic and client differentiation is present. In this test, we have two premium and two normal users, each downloading an RTP video. Additionally, there is a TCP bulk transfer running in the background. This test was also run 10 times, resulting in an average standard deviation less than 0.27 in MOS scale. The benefit of having the management system is shown in Figure 10.

Figure 10a shows the basic scenario without any traffic management. Several flows enter the AP and use approximately an even amount of resources. The TCP bulk transfer adapts to the bandwidth requirements of these flows, which can be seen as a correlation between bulk throughput and video quality (i.e. videos enter a small bit rate phase, which improves the perceived quality and allows more capacity to the bulk transfer). Regardless, the quality of the video streams is unacceptable during higher bit rates.

The management system is enabled in Figure 10b, and an improvement can be seen for streams belonging to premium users. At approximately 30 seconds, the management system perform a promotion of premium flows to a higher class (but as stated earlier, the flow promotion during congestion can cause additional packet losses, hence the slightly worse quality compared to normal users). At approximately 70 seconds, the system runs a check using performance models and notices that it cannot fully support the premium flows if both of them reside in the higher class. Therefore, one of these flows is depromoted back to the default class. Now, one of the flows receives the best possible quality during congestion, while the other premium user will have a satisfactory video quality. For the normal users, there is hardly any difference between having and not having the management system. From these results we can conclude, that the system is able not only to perform prioritization, but to intelligently optimize the loads of different classes, whenever feasible [3].

7. Conclusions

In this paper, we presented a generic QoS/QoE framework for enabling quality control in packet-switched networks. We focused especially in the management component of this framework to show how data collected from a network can be refined into knowledge of quality perceived by users, and how to take corrective measures when necessary. We also showcased this framework by introducing and demonstrating a network access point management software, built around the proposed framework's architecture.

We executed several tests where RTP video streams were subject to quality-driven network control. We evaluated the performance of the system and functionalities of traffic differentiation, client differentiation, admission control and quality optimization. According to the results, the system performed as expected, and each of these tests resulted in improved quality for the relevant clients. Especially good results were observed for customer differentiation, which is an important aspect to ISP's; the tests showed that it is possible to improve the perceived quality of premium users without sacrificing the quality of streams belonging to normal users.

Currently, the demonstrated AP management system supports quality-driven control for RTP streams only. This is due only to the availability of a suitable model for RTP video quality, and could trivially be extended by adding other models (e.g. for HTTP-based video streaming). Although more applications need to be supported, fortunately there is a need to support only applications which are most relevant to the ISP and customers. Namely, these are real-time multimedia applications like games, VoIP and video, since in these applications quality control is relevant. A particularly good application would be HTTP video streaming, as it has recently become very popular among OTT content providers.

[3] Starting from approximately 120 seconds, all flows suffer from bad quality, because the video bit rate rises so much that there is not enough capacity to support even a single flow.

Another limitation in this work is the throughput estimation of new flows, as explained in the admission control test. Regardless of how the estimation is done, with this kind of management system it is more important not to under-utilize the access point. The management software can deal with large amounts of traffic, but it is more beneficial for the user to know at the very beginning if the stream will be supported or not. This also enables the system to offer more bandwidth to non-constrained traffic when the AP is not filled with high-priority traffic which constantly features intolerable quality. This is why admission control is essential although not seemingly necessary from the point of quality management.

When congestion occurs, distributing the resources QoE-wise becomes an optimization problem. Customers may have different priorities, but so have applications. When the perceived quality of one application drops below an acceptable level, the application can be considered useless and therefore all resources distributed to that application are, in a way, wasted. Traditional approaches of conflating application quality with the bandwidth allocated to it no longer apply; different multimedia applications require not only capacity but also other guarantees in terms of delay, jitter and packet losses. When these factors are combined human perception, optimization becomes even more difficult. A momentary good QoE doesn't mean much to users if their application is operational only occasionally, as seen in the tests performed, but the quality also needs to remain stable. We introduced a framework and a system which is not only able to make a good decision for given time instant, but also to predict the outcome of different decisions and pick the most optimal one. The system is not only able to improve the quality of selected streams, but also to be conservative with the available resources and identify when they are really needed.

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