Performance Evaluation of Scheduling Mechanisms for Broadband Networks

by

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Dedicated to
My parents

Abstract

While today's computer networks supports only best-effort service, future packet-switching integrated-services networks will have to support real-time communication services that allow clients to transport information with performance guarantees expressed in terms of delay, delay jitter throughput and loss rate. An important issue in providing guaranteed performance service is the choice of packet service discipline at the switching nodes. Recently, a number of new service disciplines that are aimed at providing per-connection performance guarantees have been proposed in the context of high-speed packet switching networks. Of these, fair queuing algorithms have gained significance in that they offer differentiated services to different classes of traffic and attempt to maintain fairness among competing users. Weighted Fair Queuing (WFQ) is one such fair queuing scheme. This thesis introduces a modification to the original WFQ and characterizes its delay and throughput performance in comparison with the traditional WFQ. The new queuing scheme is called Hybrid FIFO-Fair Queuing (HF^{2}Q) since it exhibits both First-In First-Out (FIFO) and WFQ behaviors under different conditions. The modification introduced in the virtual time updation in WFQ results in packets getting ordered as in a *FIFO* queuing system when the total system load is less than unity. However, when the system is overloaded, HF^{2}Q behaves as traditional WFQ. These conclusions have been established from via simulation studies. This thesis also analyses the performance of Data Over Cable Service Interface Specification 1.1 (DOCSIS 1.1).

DOCSIS 1.1 is a standard interface for cable modems, on which the IEEE 802.16 standard for Broadband Wireless Access (BWA) systems was based. DOCSIS 1.1 provides Quality of Service (QoS) guarantees to different classes of traffic via new scheduling features like packet priorities and packet fragmentation. This thesis presents a detailed performance analysis via simulations of these QoS scheduling features applicable to specific classes of traffic. The performance of two classes of service namely Unsolicited Grant Service (UGS) and Best Effort introduced by DOCSIS was studied under different network conditions. The impacts of the different DOCSIS scheduling features on the different traffic classes are thoroughly investigated and analyzed.

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1 Introduction and Motivation

In the future, broadband networks will be an integral part of the global communication infrastructure. With the rapid growth in popularity of wireless data services and the increasing demand for multimedia multi-service applications (voice, video, audio, data) it is expected that future broadband networks, both wired and wireless, will provide services for heterogeneous classes of traffic with different quality of service (QoS) requirements. Currently, there is an urgent need to develop new technologies for providing QoS differentiation and guarantees in such networks. Among all the technical issues that need to be resolved, packet scheduling is one of the most important. Scheduling algorithms provide mechanisms for bandwidth control, and congestion control policies are all dependent on the specific scheduling algorithm used. Many scheduling algorithms capable of providing certain guaranteed QoS have been developed for wireline networks.

The focus of this thesis is two-fold. The main focus is to characterize the performance of a new scheduling algorithm called Hybrid FIFO Fair Queuing (HF^2Q). This new algorithm was stumbled upon by a fellow student [1] while studying the properties of Weighted Fair Queuing, an existing queuing algorithm for packet-switched networks. HF^2Q has very interesting properties in that it is a combination of two existing queuing schemes, namely the conventional First-In-First-Out (FIFO) queuing scheme and WFQ. Detailed descriptions of all these are provided in the subsequent chapters. Doing a complete performance characterization is important since it helps us understand the operation and employ it in suitable networks where such a behavior would be beneficial.

The other aspect of this thesis is to study the performance of IEEE 802.16 standard (standardized in 2001), which is the Medium Access Control standard for wireless Metropolitan Area Networks (MAN) or fixed Broadband Wireless Access (BWA) networks. IEEE 802.16 was largely based on Data Over Cable Service Interface Specification 1.1 (DOCSIS 1.1), which is the *defacto* standard for delivering broadband services over Hybrid Fiber Coax (HFC) networks. Hence the basic protocol operation and QoS features of both the protocols are the same. IEEE 802.16 specifies a number of new features for providing QoS guarantees over a broadband wireless network. It is important to understand the basic operation and the various QoS features provided by this complex standard. This in turn helps in design of wireless access systems with required QoS guarantees for specific end-users.

QoS and Scheduling

Many future applications of computer networks will rely on the ability of the network to provide ``Quality of Service'' (QoS) guarantees. These guarantees are usually in the form of bounds on end-to-end delay, bandwidth, delay jitter (variation in delay), packet loss rate, or a combination of these parameters. Broadband packet networks are currently enabling the integration of traffic with a wide range of characteristics -- from video traffic with stringent QoS requirements to ``best-effort'' data requiring no guarantees – all within a single communication network. QoS guarantees can also be provided in conventional IP packet networks by the use of proper packet scheduling algorithms in the routers (gateways).

Providing QoS guarantees in any packet network requires the use of traffic scheduling algorithms in the switches or routers. The function of a scheduling algorithm is to select, for each outgoing link of the switch (or router), the packet to be transmitted in the next cycle from the available packets belonging to the flows sharing the output link. These scheduling algorithms are implemented in a packet scheduler in which several inputs are buffered and there is a single server. The basic scheduler architecture is shown in Figure 1.1. The implementation of the algorithm may be in hardware or software.

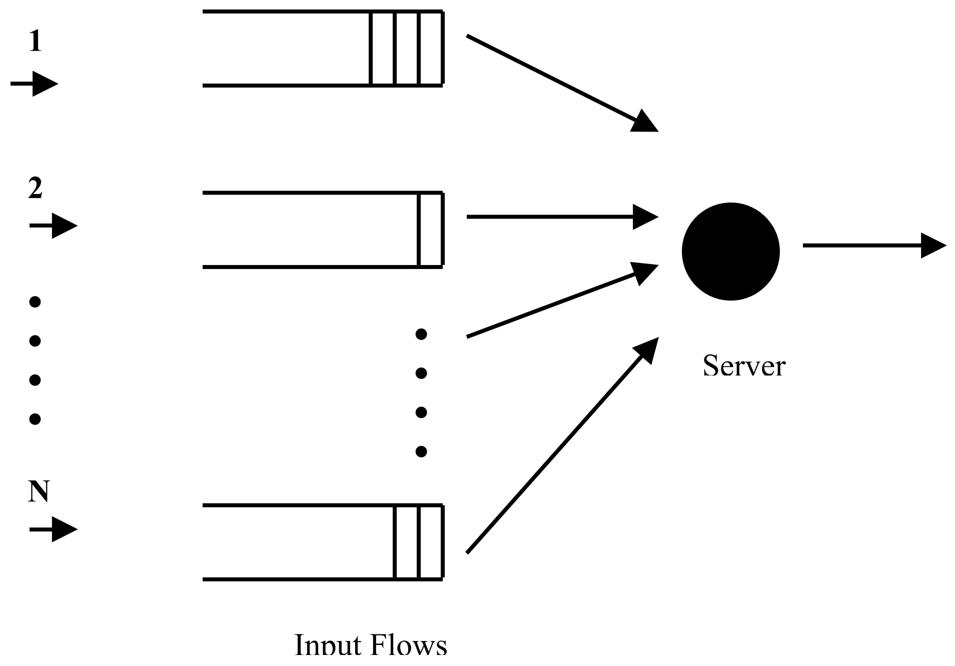


Figure 1.1 Scheduler Architecture

Although it is possible to build a guaranteed performance service [2] on a broad class of service disciplines, it is desired that a service discipline to be efficient, protective, flexible, simple and fair.

Efficiency. To guarantee certain performance requirements we need a connection admission control policy to limit the guaranteed service traffic load in the network. A service discipline is more efficient than another one if it can meet the same *end-to-end* performance guarantees under heavier load of guaranteed service traffic.

Protection. Guaranteed service requires that a network protects well-behaving guaranteed service clients from three sources of variability: ill-behaving users, network load fluctuation and best-effort traffic. It is essential that the service discipline should meet the performance requirements of packets from well-behaving guaranteed service clients even in the presence of ill-behaving users, network load fluctuation and unconstrained best-effort traffic.

Flexibility. The guaranteed performance service needs to support applications with diverse traffic characteristics and performance requirements. Because of the vast diversity of traffic characteristics and performance requirements of existing applications, as well as the uncertainty about future applications, an ideal service discipline should be flexible to allocate different delay, bandwidth and loss rate quantities to different guaranteed service connections.

Simplicity. The service discipline should be both conceptually simple to allow tractable analysis and mechanically simple to allow high-speed implementation.

Fairness. Fairness is a desirable property in queuing algorithms that enable adequate congestion control in networks even in the presence of ill-behaved sources (sources that transmit at a rate more than what it is supposed to transmit at). Thus fairness in the sense of allocating bandwidth and buffer space in a *fair* manner automatically ensures that ill-behaved sources get no more than their fair share. WFQ provides this sort of protection in the form of bandwidth allocation. But the concept of fairness can also be viewed from a different perspective. While WFQ provides fair bandwidth allocation, FIFO provides *fairness* in the sense that it provides equal *mean waiting time* for packets from all flows irrespective of their offered load. In a sense, FIFO can be deemed as a fair queuing scheme in that it does not discriminate between various packets in terms of mean delay. Furthermore with FIFO queuing the order of arrival determines the bandwidth and buffer allocation. The remainder of this thesis is organized as follows.

Chapter 2 discusses the relevant work in the area of packet service disciplines and IEEE 802.16 in the context of BWA networks.

Chapter 3 discusses Weighted Fair Queuing [WFQ] and the proposed modification to it, resulting in a new queuing discipline Hybrid FIFO Fair Queuing (HF^{2}Q). Chapter 3 discusses the operation and features of HF^{2}Q in detail.

Chapter 4 presents the analysis and results of the performance characterization of HF^{2}Q.

Chapter 5 presents a detailed discussion on the operation and performance analysis of QoS aspects of IEEE 802.16 for Broadband Wireless Access networks.

Chapter 6 presents a summary of results obtained, conclusions drawn and possible future work.

2 Background and Related Work

A number of service disciplines have been proposed in the past in the context of high-speed packet switched network. The performances of these service disciplines have been studied and new analysis techniques have been proposed to address their performance issues. Since the one focus of this thesis is to discuss the performance of a new service discipline as a modification of Weighted Fair Queuing, relevant work in the area of queuing algorithms will be presented.

2.1 Classes of Queuing Algorithms

Queuing algorithms can be thought of as allocating three nearly independent quantities [3]: bandwidth (*which* packets get *transmitted*), promptness (*when* do those packets get *transmitted*), and buffer space (*which* packets are *discarded* by the gateway). Multiple packets exist in one or more buffers sharing a common outgoing link; the scheduler chooses a packet for service. There are several queuing methods, including FIFO (first-in, first out), priority queuing, and fair queuing. All the queuing algorithms discussed below are *work conserving*, that is, the server is never idle when a packet is buffered in the system

1. FIFO queuing

The simplest queuing algorithm is the first-in, first-out queuing technique in which the first packet in the queue is the first packet that is processed i.e. transmitted. When the queue becomes full due to traffic congestion, incoming packets are dropped. FIFO relies on end systems to control congestion via congestion control mechanisms. FIFO queuing essentially relegates all congestion control to the sources, since the order of arrival completely determines the bandwidth, promptness and buffer space allocations. It thus does not offer any protection and it is not very flexible in the sense that it does not provide performance guarantees for different kinds of traffic. FIFO is extremely simple to implement and has a fairly tractable analysis.

2. Priority queuing

This technique uses multiple queues, but queues are serviced with different levels of priority, with the highest priority queues being serviced first. When congestion occurs, packets tend to be dropped from lower-priority queues first. The only problem with this method is that lower-priority queues may not get serviced at all if high-priority traffic is excessive. Packets are classified and placed into queues according to information in the packets, such as flow ID, explicit priority, class/service indication, etc. Priority queuing is efficient in the sense that it provides some amount of performance guarantees, i.e. gives preferential treatment to higher priority queues compared to lower priority ones. It is also easy to implement.

3. Fair queuing

The fair queuing concept was originally proposed by Nagle [4] in order to solve the problem of malicious or erratic TCP implementations. The goal of the algorithm was to guarantee each session a portion of the network resources even though some sessions are transmitting at a higher rate than the allocated one. A round-robin approach is used to service all queues in a fair way. This prevents any one source from overusing its share of network capacity. Fair Queuing service disciplines address this scheduling problem by allocating bandwidth fairly among competing flows regardless of their prior usage or congestion. In particular, these disciplines do not penalize flows for the use of idle bandwidth. Fairness offers protection from "misbehaving" flows and leads to effective congestion control and better services for rate-adaptive applications. Strict QoS guarantees such as throughput or delays can also achieved. Some of the important work related to fair queuing is described below.

A. Virtual Clock

The Virtual clock discipline [5] aims to emulate the Time Division Multiplexing (TDM) scheme. Each packet is allocated a virtual transmission time, which is the tine at which the packet would have been transmitted were the server actually doing TDM. The packets are transmitted in the increasing order of virtual transmission times. Virtual Clock algorithm ensures that a well-behaving connection gets good performance in the presence of connections that send packets at a rate higher than they are supposed to.

B. WFQ and WF^{2}Q

Weighted Fair Queuing (WFQ) [3] is a packet scheduling policy that tries to approximate Fluid Fair Queuing (FFQ) or Generalized Processor Sharing (GPS) policy [6]. FFQ is a general form of head-of-line processor sharing service discipline [7]. There is a separate FIFO queue for each connection sharing the same link. GPS serves the non-empty queues in proportion to their service shares. GPS is impractical since it assumes that the server can serve all connections with non-empty queues simultaneously and the traffic is infinitely divisible. WFQ, also known as the Packet Generalized Processor Sharing or PGPS [8], is the most well known approximation of the GPS service in a packet system. In WFQ when the server is ready to transmit the next packet at a particular time t, it picks the first packet that would complete service in the corresponding GPS system if no additional packets were to arrive after t.

While WFQ uses only finish times of packets in the GPS system, WF^2Q [9] (Worst Case Weighted Fair Queuing) uses both start times and finish times of the packets in the GPS system to achieve a more accurate emulation. In WF^2Q, when the next packet is chosen for service at time t, rather than selecting it from among all the packets at the server as in WFQ, the server only considers the set of packets that have started receiving service in the corresponding GPS system, and selects the packet among them that would complete service first in the corresponding GPS system.

WFQ and WF^2Q have most of the properties desired in a queuing algorithm. They offer protection to well-behaving flows under overloaded conditions and they are very flexible in the sense that they can support applications with different performance requirements. The main drawback is that it is not very easy to implement.

C. Self-Clocked Fair Queuing

A simpler packet approximation algorithm of GPS is Self-Clocked Fair Queuing (SCFQ) [10] also known informally as "Chuck's Approximation" [11]. To reduce the complexity in computing the reference times in the GPS system, SCFQ introduces an approximation algorithm based on the observation that the system's time in the reference GPS system at any moment may be estimated from the service time of the packet currently being serviced. Moreover, SCFQ scheme is nearly optimal in the sense that the services received by any pair of backlogged sessions, normalized to the corresponding service shares, stay close to each other.

4. CBQ (Class-Based Queuing)

CBQ [12] is a class-based algorithm that schedules packets from several queues and guarantees a certain transmission rate to each queue. If a queue is not in use, the bandwidth is made available to other queues. From this simple description, CBQ is seen to be similar to WFQ. But CBQ is a hierarchical queuing algorithm. In hierarchical link sharing [13] there is a class hierarchy associated with each link that specifies the resource allocation policy for that link. A class represents a traffic stream or aggregate of traffic streams that are grouped according to administrative affiliation, protocol, traffic type or other criteria. A CBQ-compliant device looks within packets to classify packets according to addresses, application type, protocol, URL, or other information. Thus CBQ is more than a queuing scheme; it is also a QoS scheme that identifies different types of traffic and queues the traffic according to predefined parameters.

5. Delay-Earliest-Due-Date (Delay-EDD)

Delay-EDD [14] is an extension to the classic Earliest-Due-Date-First scheduling [15], in which each packet from a periodic traffic stream is assigned a deadline and the packets are sent in the order of increasing deadlines. In Delay-EDD, the server negotiates a contract with each source wherein if a source obeys its promised traffic specification, then the server will provide a delay bound. The key lies in the assignment of deadlines to packets. The server sets a packet's deadline to the time at which it should be sent had it been received according to the contract.

2.2 MAC Standard for Fixed Broadband Wireless Access

Broadband Wireless Access (BWA) systems have gained an increased interest during the last two years. This has been fuelled by a large demand on high frequency utilization resulting in a crowded spectrum as well as a large number of users requiring simultaneous multi-dimensional high data rate. A BWA system uses new network architectures to deliver broadband services in a fixed point-to-point or point-to-multipoint configuration to residential and business customers. BWA networks support voice, data, video distribution services and emerging interactive multimedia communications. Large bandwidth, lower installation cost and ease of deployment, coupled with recent advancements in semiconductor technologies for wireless applications, make BWA an attractive solution for broadband service delivery. The wireless medium is a shared medium, which demands a Medium Access Control (MAC) protocol to co-ordinate the transmission of multiple traffic flows over it. The downlink flows are simply multiplexed by the access point and there is no contention.

The uplink direction is more challenging because of potential contention, and the MAC protocol is needed to minimize the contention probability. The medium access control protocol also has to include the uplink scheduling in order to accommodate the demand assignment multiple access and dynamic resources allocation. The IEEE 802.16 standard [16] has formally been approved as the standard for BWA networks by the IEEE Standards Association in 2001. It is worth mentioning that IEEE 802.16 is a consolidation of two proposals, one of which was based on Data Over Cable Service Interface Specification (DOCSIS) [17]. DOCSIS is the *de facto* standard for delivering broadband services over Hybrid Fiber Coax (HFC) cable networks. DOCSIS was developed by a group of major cable operators called Cable Labs. DOCSIS was later adopted by the ITU and is now supported by many vendors. Versions 1.0 and 1.1 of DOCSIS were introduced in 1999, and version 2.0 was introduced in 2000.

Not much relevant work has been done directly in the area of performance analysis of Medium Access Control standard (IEEE 802.16) for BWA. [18] discusses the performance of possible MAC protocols for Fixed Broadband Wireless Access Networks. [19] describes new Quality of Service Scheduling architecture for Cable and BWA systems.

3 Description of HF^{2}Q

Hybrid FIFO Fair Queuing (HF^{2}Q) is a new service discipline for packet-switched networks that exhibits the properties of both WFQ and First In First Out (FIFO) under different load conditions. This new behavior is a result of a modification in the operation of regular WFQ. This chapter presents a detailed description of the notion of FIFO in comparison with fair queuing, the evolution of WFQ and its principles of operation, a description of HF^{2}Q operation and its interesting properties, and a summary of comparison of both the service disciplines.

3.1 First In First Out (FIFO)

First-Come-First-Serve (FCFS) or First-In-First-Out (FIFO) is one of the simplest scheduling policies. Its operation is that packets are served in the order in which they are received. It is quite simple to implement. In particular, insertion and deletion from the queue of the waiting packets are constant time operations and do not require any per-connection state to be maintained by the scheduler, so it is one of the most commonly implemented policies. However, the "best-effort service " offered by FIFO scheduling does not readily provide delay or throughput guarantees.

In terms of delay, FIFO results in the same expected waiting time for every arriving packet (at least under the assumption of Poisson arrivals). FIFO provides no mechanism for providing different waiting times for different flows. In some sense, this is "fair" treatment of the flows. One way to provide a delay *bound* (for all flows) is to limit the buffer size so that once a packet is queued up for transmission it is guaranteed to be sent out in the time it takes to serve a full queue of packets. But in this case, packets have to be dropped if the limited buffer is full when packets arrive.

Neither does FIFO explicitly provide any mechanisms for fair sharing of link resources, but with the help of some buffer management schemes it is possible to control the sharing of bandwidth. Buffer management schemes [20] can be achieved by monitoring the congestion levels of queues (queue occupancy or queue occupancy increase rate) and using this mechanism to modify buffer portioning or discarding policy. FIFO does not provide flow isolation and essentially relegates all congestion control to the sources, since the order of arrival completely determines the bandwidth promptness and buffer allocations. So there have to be flow control algorithms which when universally implemented can overcome these limitations. Unfortunately, no matter what congestion control is used at the sources, FIFO does not protect well-behaved sources against ill-behaved ones. A single source sending packets at a sufficiently high speed can capture a high fraction of bandwidth of the outgoing line [3]. As mentioned before, for a finite buffer case the delay is then very high but is bounded, but for an infinite buffer the delay of all the packets grows infinitely.

3.2 Fair Queuing Criterion

Fair queuing was originally developed [4] as an attempt to maintain fairness in the amount of services provided at a service point to the competing users. Unlike FIFO queuing discipline where a session can increase its share of service by presenting more demand and keeping a large number of packets in the queue, the primary goal in fair queuing is to serve sessions in proportion to some pre-specified service shares, independent of queuing load presented by the sessions. Fair queuing algorithms provide protection, so that ill-behaved flows have only limited negative impact on well-behaved flows. Allocating bandwidth and buffer space in a fair manner automatically ensures that ill-behaved sources can get no more than their fair share.

A queuing algorithm is said to be fair if the *fairness criterion* holds. We begin by introducing some notations. Consider a queuing system served by an access link with a total output link capacity of C (bits/second). Let us denote by K the set of flows setup on this link, and by r_k, $k \in K$, the service rate (in bits/second) associated with each flow k. Define $W_k(t_1, t_2)$, $t_2 > t_1$, as the aggregate service (in bits) received by flow k during the time interval $[t_1, t_2]$. The value of $W_k(t_1, t_2)$ is simply equal to the time spent by the server on flow k during $[t_1, t_2]$ times the server speed C. $W_k(t_1, t_2) / r_k$ is then the total service provided to flow k during the time interval $[t_1, t_2]$ normalized to its corresponding transmission rate. We call this quantity the *normalized service* received by k during $[t_1, t_2]$. At any time t, a flow may be *backlogged* or else *absent*. A flow is backlogged at time t if a positive amount of session's traffic is queued at time t. We denote by $B(t)$ the set of flows that are backlogged at time t and by $B\ (t_1, t_2)$ the set of flows that are backlogged during the entire interval $[t_1, t_2]$. The fairness criterion is said to hold if the following condition is satisfied:

For intervals of time $[t_1, t_2]$, in which both the flows k and j are backlogged,

$$\frac{W_k(t_1,t_2)}{r_k} - \frac{W_j(t_1,t_2)}{r_j} = 0, \qquad j,k \in B(t_1,t_2) \qquad (3.2.1)$$

In real packet networks, an entire packet of a certain flow must be transmitted before service may be shifted to another flow. Therefore, it is not possible to satisfy the fairness criterion above exactly for all intervals of time. Instead, the objective of a fair *packet-scheduling* algorithm is to ensure that the quantity $|W_k(t_1,t_2)/r_k - W_j(t_1,t_2)/r_j|$ is bounded for all intervals of time and is as close to zero as possible.

3.2.1 Generalized processor Sharing

Generalized-Processor-Sharing (GPS) is an ideal scheduling discipline that is defined with respect to a fluid-model, where packets are considered to be infinitely divisible. Thus it is also called as Fluid Fair Queuing (FFQ) [2]. With a fluid flow model of traffic, the service is offered to sessions in arbitrarily small increments. Equivalently, it may be assumed that multiple sessions can receive service in parallel. As a result, it is possible to divide the service among the sessions, at all times, exactly in proportion to the specified service shares. A GPS server is work conserving and operates at a fixed rate C. It is characterized by positive real numbers r_1, r_2, r_3,..............r_N. Let $W_k(t_1, t_2)$ and $W_j(t_1, t_2)$, be defined as before , then a GPS server is one which

$$\frac{W_k(t_1,t_2)}{W_j(t_1,t_2)} \geq \frac{r_k}{r_j}, \qquad j = 1,2,3...,N \qquad (3.2.1.1)$$

for any session k that is continuously backlogged in the interval $[t_1,t_2]$.

Session k is guaranteed a rate of

$$g_k = \frac{r_k}{\sum_j r_j} C \qquad (3.2.1.2)$$

A problem with GPS is that it is an idealized discipline that does not transmit packets as entities. It assumes that the server can serve multiple sessions simultaneously and that traffic in infinitely divisible. But in actual packet-based traffic scenarios, only one session can receive service at a time, and an entire unit of traffic (referred to as a packet) must be served before another unit is picked up for service.

3.3 Weighted Fair Queuing

Weighted Fair Queuing (WFQ) and its derived algorithms are excellent approximations to the more generalized and ideal form of fair queuing viz, Generalized Processor Sharing (GPS) even when the packets are of variable length. Round Robin, which is an early form of fair queuing, assumes an equal service share for all the sessions and an equal length for all packets. It provides service to the sessions in a round robin fashion, picking one packet for service from each session (or queue) with backlogged traffic, and then proceeding to the next session. When the lengths of the packets are not the same and/or the service shares assigned to the sessions are not equal, the definition of fair queuing and the right order of providing service becomes a more subtle matter. To formulate fair queuing in this more general case, the notion of fairness was applied to an idealized fluid-flow traffic environment and then the outcome was used to specify fair queuing for the actual packet-based traffic scenario. Packet-by-Packet GPS (PGPS) or WFQ does not make the infinitesimal packet size assumption in GPS, and with variable-size packets, they do not need to know a connection's mean packet size in advance.

3.3.1 Transmission Scheme

Let F_p be the time at which packet *p* will depart (finish service) under GPS. Then, a very good approximation of GPS would be a work-conserving scheme that serves packets in increasing order of F_p. Now suppose the server becomes idle (completes service of some packet) at time *t.* The next packet to depart under GPS may not have arrived at time t and, since the server has no knowledge of when this packet will arrive, there is no way for the server to be both work conserving and serve the packet in increasing order of $F_{p.}$ The server picks the first packet that would complete service under the GPS simulation if no additional packets were to arrive after time *t.* This scheme is accomplished as follows.

3.3.2 Virtual time

WFQ operation is linked to the reference fluid-based GPS system by defining a system-wide function called *virtual time*. The virtual time, denoted by $v(t)$, is associated with the reference GPS system and measures the progress of work in that system; it is dependent on system load. The virtual time is used in WFQ to define the order in which packets are served in the packet-by-packet scheduling algorithm. The definition of the virtual time is that $v(t)$ has a rate of increase in time equal to that of the normalized service received by any backlogged flow *k* in the reference GPS system. In the reference fluid-based system, all backlogged flows receive exactly the same normalized service with time. Mathematically,

$$\frac{d}{dt}v(t) \equiv \lim \Delta t \to 0 \left[\frac{W_k\left(t, t \ + \Delta t\right) - W_k\left(t\right)}{\Delta t . r_k} \right], \quad k \in B(t, t + \Delta t) \qquad (3.3.2.1)$$

In other words, during a busy period of the server, we can say that:

$$v(t_2) - v(t_1) = \frac{W_k(t_1, t_2)}{r_k}, \quad k \in B(t_1, t_2) \qquad (3.3.2.2)$$

where [t_1, t_2] is an arbitrary subinterval of the busy period. It can be shown that the above definition leads to the following expression for evaluating $v(t)$:

$$v(t_2) - v(t_1) = \frac{C}{\sum_{k \in B(t_1, t_2)} r_k} \cdot (t_2 - t_1) \qquad (3.3.2.3)$$

where C is the total output link capacity in bits/s. Hence, $v(t)$ is a piecewise linear increasing function of time with a slope that changes whenever the set of backlogged flows $B(t)$ changes and is inversely proportional to the sum of service shares of sessions backlogged. Figure 3.3.2.1 shows the variation of $v(t)$ with time. Note that $v(t)$ does not in general have a monotonically increasing slope (as illustrated in the figure).

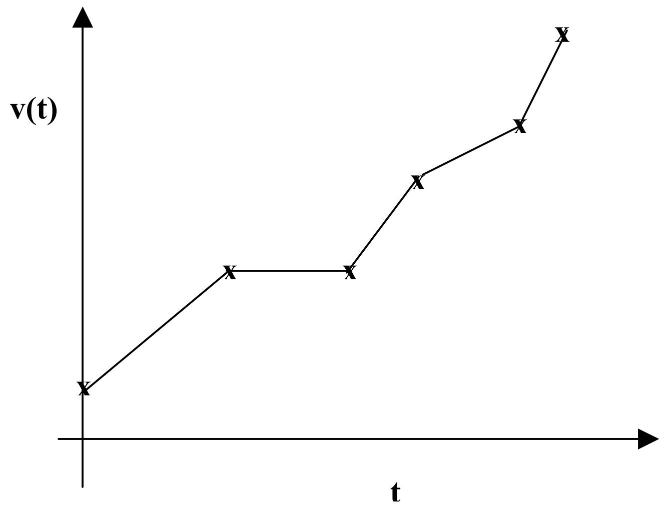


Figure 3.3.2.1 Variation of v(t) with time

Now that the virtual time of the reference GPS system has been defined, the scheduling operations of packet-based WFQ are introduced. Denote the *i*th packet of flow *k* by p_k^i, its arrival time by a_k^i and its length by L_k^i. Whenever a packet arrives at the WFQ scheduler, it is assigned a *virtual finish time*, F_k^i, i = 1, 2, 3, …, given by:

$$F_k^i = \frac{1}{r_k} L_k^i + S_k^i \qquad i = 1,2,\cdots, \quad k \in K \tag{3.3.2.4}$$

and,

$$F_k^0 = 0, \qquad k \in K \tag{3.3.2.5}$$

where,

$$S_k^i = \max(F_k^{i-1}, v(a_k^i)) \qquad i = 1,2,\cdots, \quad k \in K \tag{3.3.2.6}$$

S_k^i denotes the Start Number which is the maximum of the finish time of the (i-1) th packet of the kth flow and $v(a_k^i)$ which is the value of $v(t)$ at the time of packet p_k^i

arrival. If the packet arrival is to an inactive queue, then the start number would be the current virtual time (i.e. $v(a_k^i)$). If the arrival is to an active queue, then the start number would be the finish time of the last packet in the queue on arrival. It can be shown [2] that serving packets in the order of increasing virtual finish times F_k^i, as computed by (3.3.2.4) and (3.3.2.5), is equivalent to serving them in the increasing order of their actual finish times in the reference GPS system. Hence, all WFQ has to do is compute a virtual finish time tag for each arriving packet and then serve packets in the increasing order of their virtual finish time tags. Computing finish time tags for each arriving packet is a simple operation that requires knowing only the packet's length, virtual finish time of the previous packet, and the system's virtual time when the packet arrives. Some of properties of the virtual time interpretation from the standpoint of implementation are

- The virtual finishing times can be determined at the packet arrival time.
- The packets are served in the order of virtual finishing time.
- We need to update the virtual time when there are events in the GPS system, where each arrival and departure from the GPS system is denoted as an event.

However the price to be paid for these advantages is some overhead in keeping track of the set of backlogged flows. Another troublesome point in the implementation of WFQ, as outlined above, is the requirement of evaluating $v(t)$ of the reference GPS system at each packet arrival instant. As mentioned earlier, $v(t)$ is a piecewise linear function, the slope of which changes whenever some flow k becomes backlogged or absent in the reference GPS system. Such slope changes constitute breakpoints in the piecewise linear form of $v(t)$.The computational complexity of evaluating $v(t)$ depends on the frequency of such breakpoints.

3.4 Hybrid FIFO Fair Queuing (HF^{2}Q)

This promising new queuing algorithm was stumbled upon during the process of development of new simulation module for WFQ [1]. It turns out that introducing a minor modification to the operations of WFQ transforms such a fair queuing policy into a totally different queuing algorithm that exhibits very interesting properties. We now call the new algorithm the Hybrid-FIFO/FQ system (or HF^{2}Q for short) because it exhibits desirable properties from both WFQ and FIFO scheduling policies. It exhibits fairness in both perspectives, i.e. in bandwidth allocation (protecting well-behaved sources in an overloaded system as WFQ does) and also offering equal mean waiting times to all flows (as FIFO does) when total offered load is less than unity. The description of this new queuing algorithm is presented here and its properties and behavior are investigated in the next chapter. The following sections describe the modifications required to transform WFQ into HF^{2}Q and explain the preliminary observations made about this new queuing system.

3.4.1 Modifications made to formulate HF^{2}Q

The operations of packet-based FQ algorithms (including WFQ) explained in previous sections are dependent on *virtual time*, *v* (*t*), which in WFQ is calculated using (3.3.2.2)

$$v(t_2) - v(t_1) = \frac{C}{\sum_{k \in B(t_1,t_2)} r_k} \cdot (t_2 - t_1) \tag{3.4.1.1}$$

where *C* is the total output link capacity in bits/s and [t_1, t_2] is a of a busy period of the associated reference GPS system subinterval in which the set of backlogged flows *B(t)* does not change. Hence, *v*(*t*) is a piecewise linear increasing function of time with a slope that changes whenever the set of backlogged flows *B*(*t*) changes. When a packet arrives, virtual time (represented by a variable called *roundNumber* in the actual implementation) is updated. The implementation keeps track of instants of time when the *roundNumber* is updated using the *lastRUpdateTime* variable, which indicates the last time when the *roundNumber* was updated.

$$roundNumber = roundNumber + \frac{(currentTime - lastRUpdateTime)}{weightSum} \cdot capacity \tag{3.4.1.2}$$

In HF^{2}Q, the *lastRUpdateTime* variable is not updated in a scenario when the *roundNumber* variable is not updated at a new packet arrival *because* there are no backlogged flows in the reference GPS system, i.e. the GPS system is idle in an interval [t, t_1] and a packet arrives at instant t_1. For WFQ, the *lastRUpdateTime* variable has to be updated at every packet arrival irrespective of the queuing system status.

3.4.2 Comparison of WFQ and HF^{2}Q operations

In WFQ, calculating the virtual time during a *busy period* is done using (3.4.1.1), where the break points t_1 and t_2 are the instants of new arrivals and/or departures as seen by the reference GPS system. The *arrivals* at the GPS system are the same as those at the WFQ system. On the other hand, *departure* instants might be different for both systems. Since both GPS and WFQ are work conserving disciplines, their busy periods coincide, i.e., the GPS server is in a busy period if and only if the WFQ server is in a busy period. Let τ_e be the end of a busy period and at this time updating *v*(*t*) should be stopped, which means it would remain at $v(\tau_e)$. When a new busy period starts again by a new arrival (say flow a$_1$) at τ_s, the first packet arrival is processed based on $v(\tau_s) = v(\tau_e)$. Let the next arrival be at time τ_2 and assume that the first packet has not completed service by τ_2. Then *v*(*t*) would be correspondingly updated to

$$v(\tau_2) = v(\tau_s) + \frac{C}{\sum\limits_{a_1 \in B(\tau_1, \tau_2)} r_{a_1}} \cdot (\tau_2 - \tau_s) \qquad (3.4.2.1)$$

Calculating the virtual time in H-F^{2}Q is exactly the same as in WFQ until the end of a busy period. The virtual time function for HF^{2}Q is denoted by $v'(t)$. At the end of a busy period τ_e, the value of the virtual time is maintained as $v(\tau_e)$, but the variable *lastRUpdateTime* is not updated. This means that at a new busy period, the first arrival at τ_s receives a time stamp based on $v(\tau_s) = v(\tau_e)$ exactly as in WFQ. However, for the next (second) arrival at τ_2, $v(t)$ would be updated to

$$v(\tau_2) = v(\tau_s) + \frac{C}{\sum_{a_1 \in B(\tau_1,\tau_2)} r_{a_1}} \cdot (\tau_2 - \tau_e) \qquad (3.4.2.2)$$

rather than to the expression mentioned in (3.4.2.1). This means that the second arrival gets a much bigger timestamp than it would under WFQ (i.e., is delayed more) since $(\tau_2 - \tau_e)$ is greater than $(\tau_2 - \tau_s)$ as indicated in figure 3.4.2.1. Figure 3.4.2.1 illustrates the change of virtual time with time in both the queuing disciplines, i.e. WFQ and HF^2Q. *v(t)* for HF^2Q is denoted as $v'(t)$ and is shown in bold in the figure. As can be seen in the figure, from time τ_2 onward, the virtual time $v'(t)$ is offset from *v(t)* by a value of $\frac{C}{r_{a_1}}(\tau_s - \tau_e)$ due to the fact the *lastRupdateTime* variable was not updated; that is, from τ_2 onward both *v(t)* and $v'(t)$ increase with the same slope. The figure 3.4.2.1 assumes that the second packet is from a different flow than the first packet, so that the slope of *v(t)* and also $v'(t)$ changes at time τ_2.

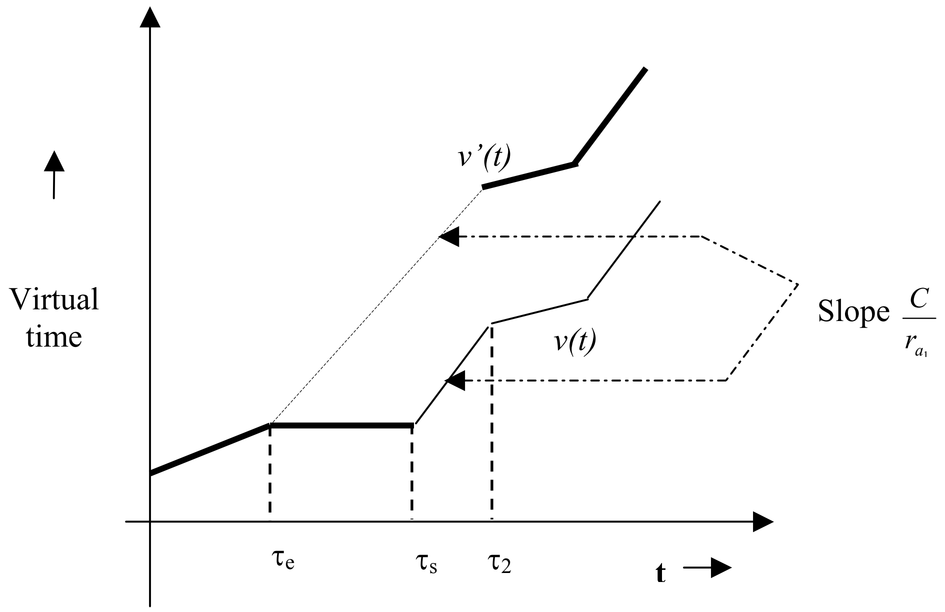


Figure 3.4.2.1 Variation of virtual time with time

Two different scenarios have to be considered with regard to the packet arrival to the system (reference system) at τ_2. The packet that arrives may be from the same flow as the packet that arrived at τ_s or may be from a different flow. It is to be noted that if the packet is from a different flow then it would get a start time that equals the value of $v'(t)$ at t = τ_2 i.e. $v'(\tau_2)$, since as explained in section 3.3.2 , an arrival to an inactive queue gets a start time that is equal to the current virtual time. On the other hand if the next arrival were to be a packet from the same flow as the packet that arrived at $\tau_{s,}$ then the start time of the packet would be the finish time of the previous packet and *not* the virtual time since this is an arrival to backlogged queue. But we already know that the for the first packet that comes to the system at the end of the busy period, the start time would be the same for both WFQ and HF^{2}Q since $v(t)$ and

$v'(t)$ are the same at t= τ_S. In both the scenarios the packet ordering would eventually be different than the packet ordering for normal WFQ. This has a cumulative effect on the order of scheduling of future packets from the different flows. As we will see in the next chapter, packets get ordered as in a FIFO queuing system if the system load is less than unity. Of course, if the first packet already left the GPS system by τ_2, this means that we reached the end of another busy period.

3.4.3 Properties of HF^{2}Q

Preliminary investigation of the HF^{2}Q queuing system in chapter 4 indicates that if the normalized system load is *smaller* than unity (i.e., $\rho \leq 1$), the queuing system provides the same mean packet delay for *all* supported flows (similar to the behavior of first-in first-out FIFO queuing). However, if the system load rate grows beyond unity (i.e., when the system encounters a congestion period), the system reverts back to operate as a WFQ system, providing protection against misbehaving flows. In such mode, the mean packet delay of misbehaving flows (flows that send above their capacity reservations) grows dramatically while the delay of other behaving flows is kept finite, in the same way WFQ operates.

4 Simulations And Analysis of Results

Simulations studies were conducted under various scenarios to characterize the performance of HF^{2}Q. The simulations were performed using the Extend simulation software. Extend (Imagine That Inc.) is a simulation environment with many features like model libraries, hierarchies of models, and its own modeling language (ModL) which resembles C. It targets both continuous systems and discrete event systems simulation, also offering the possibility of combining the two paradigms. This research was conducted using discrete event simulation techniques.

4.1 Simulation Setup

Each experiment basically had a certain number of flows with each making its own bandwidth reservation. Since the study was being done in a Broadband Wireless Access context, a link speed of 1Mbps was chosen for all the experiments. Speeds in wireless broadband can go from sub one megabit (< 1 Mbps) to 45 Mbps range. All the flows transmitted packets whose lengths were uniformly distributed between 4000-12000 bits so that the mean packet length was 8000 bits. The arrival process for each flow was Poisson. The simulations were run for an average of 2000 s. This ensured that even for both low and high load cases more than 100,000 packets were generated for a given run. In each simulation there were three flows (or sources), and the incoming load of one of the flows was varied, keeping the loads of the other flows constant. In all of the simulations, the flow (flow 1) whose load keeps varying is named the *tagged flow* and the other flows are named *non-tagged.*

The aim of the different simulation experiments was to characterize the performance of HF^2Q in terms of two performance requirements, namely the average delay and throughput. The average packet delay was measured for each flow for varying incoming loads of the tagged flow. Throughput measurements were also performed for certain cases. Each simulation was setup for a specified number of runs wherein the incoming load of the tagged flow is different for each run and the different flows had specified bandwidth reservations (indicated by weights). The aim of the simulations was to characterize the performance of HF^2Q in two scenarios:

- When the total load of the system is less than 1
- When the total load exceeds one (i.e. when the system is overloaded).

Also a comparison with WFQ is also shown for each of the different cases.

4.1.1 Queuing Model

A M/G/1 queuing model is considered, in which arrivals from flow 2 constitute a Poisson process with rate λ_i (exponential interarrival times) and service times are independent, identically distributed with a general probability distribution with mean service rate μ. In our case, the service times are uniformly distributed with a mean of 8 ms. Figure 4.1.1.1 illustrates the queuing model. There are a total of N queues and a single server to process packets from all of these queues. Each queue has its own arrival rate ($\lambda_1, \lambda_2, \lambda_3, \ldots\ldots\ldots \lambda_N$).

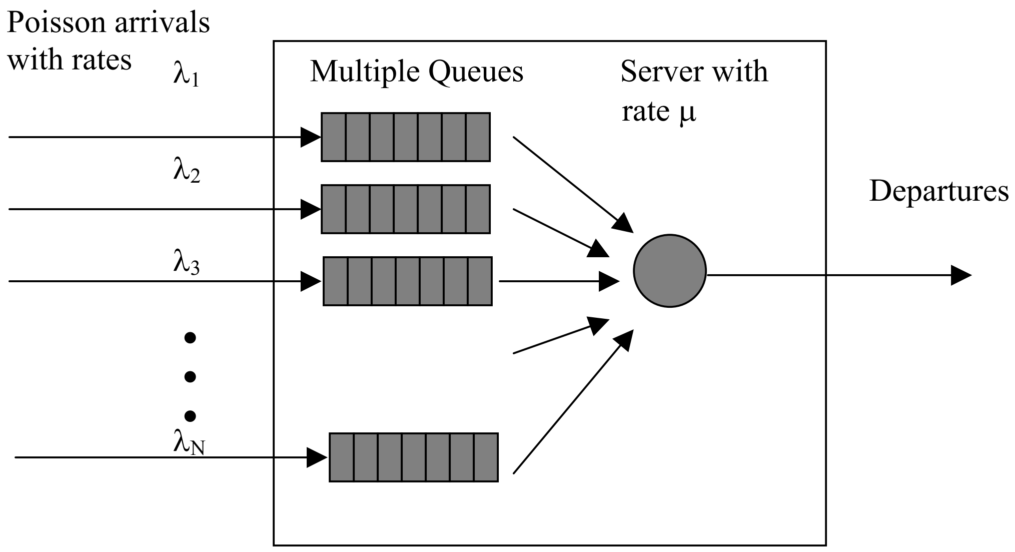


Figure 4.1.1.1 Queuing Model

4.2 Mean Delay Measurements for total load < 1

This section presents the mean delay experiments for different reservations made by the incoming flows. The mean packet delay is measured as

Mean packet delay = Mean packet waiting time + Mean service time. Since the mean packet length for all the packets is the same, the mean service time is constant for all flows and it is the waiting time in the queue that affects the mean packet delay. In all of the simulations for which the results are presented in this section, the total normalized load offered to the system (sum of incoming loads of the three flows) was less than 1. Several simulations were carried out to study the performance of HF^{2}Q in detail, with different bandwidth reservation setups. For each experiment, the bandwidth reservation and the offered load of each flow are summarized in a table. For each of the experiments, case 1 presents the HF^{2}Q results and case 2 presents the WFQ behavior for the same setup

4.2.1 Experiment 1

In the first experiment the untagged flows are identical in terms of reservation and load: both have relatively small reservation (0.2 Mbps) and an even smaller load (0.15 Mbps). Meanwhile, the tagged flow has a large reservation (0.6 Mbps) and a load that varies from 0.05 to 0.65 Mbps in 0.05 increments, for a total of 13 individual conditions. Note that the tagged flow exceeds its reservation for its larger loads but the system load never exceeds 0.95. The table 4.2.1.1 summarizes the flow setup for experiment 1.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1	0.6	0.05-0.65
2	0.2	0.15
3	0.2	0.15

Table 4.2.1.1 Flows setup: Exp. 1

Case 1: HF^{2}Q

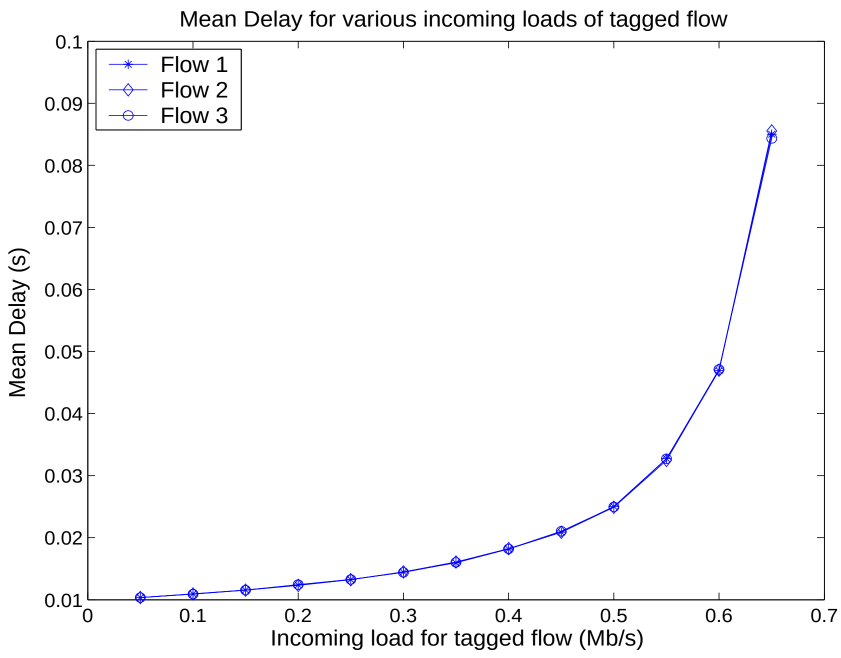


Figure 4.2.1.1 Mean Delay for HF^{2}Q: Exp. 1

Case 2: WFQ

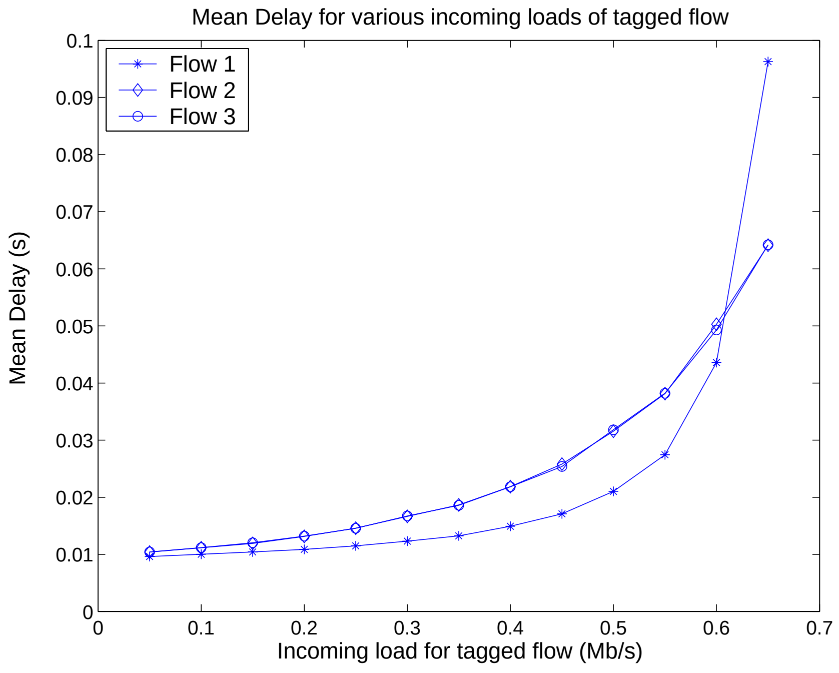


Figure 4.2.1.2 Mean Delay for WFQ: Exp. 1

Analysis:

From Figure 4.2.1.1 it is clear that HF^{2}Q provides equal delays for all flows, i.e. HF^{2}Q system behaves as a single FIFO queue so long as the total load offered to the system is less than 1. Packets from the different flows get FIFO treatment even after the tagged flow exceeds its bandwidth reservation of 0.6 Mbps. We will demonstrate later (section 4.5) that after a short initial transient period, HF^{2}Q does indeed serve packets in the order of their arrival (FIFO). WFQ on the other hand (see figure 4.2.1.2) offers lesser packet delay to flow 1, since it has a higher weight of 6 compared to flow 2 and flow 3, which have weights of 2;this remains true so long as the flow conforms to its bandwidth reservation. It is evident that WFQ offers protection to the behaving flows against the ill-behaving flow. It is seen that flows 2 and 3 are protected and get lower delays when flow 1 exceeds its bandwidth reservation, ie when the incoming load of flow 1 is greater than 0.6 Mbps, whereas the delay of packets belonging to flow 1 increases. Also flows 2 and 3 get equal delays since they have the same bandwidth reservation and same offered load. A few more experiments which are slight modifications to experiment 1 (different reservations, and non-tagged flows having equal reservations) were performed. These experiments are not discussed separately since they have the same basic results as experiment 1 does. Detailed results (tables, graphs) for these experiments are presented in Appendix 1.

4.2.2 Experiment 2

In this experiment the tagged flow has a reservation of 0.5 Mbps and the two untagged flows have different reservations (but close to each other) and offered loads. The tagged flow load varies from 0.05 to 0.55 Mbps again in increments of 0.05 for a total of 11 different conditions. Here the tagged flow exceeds its reservation when its incoming load is 0.55 Mbps and the experiment was conducted till the total system load reached 0.95. The table 4.2.2.1 summarizes the flow setup for experiment 2.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1(tagged)	0.5	0.05-0.55
2	0.3	0.25
3	0.2	0.15

Table 4.2.2.1 Flows setup: Exp. 2

Case 1:

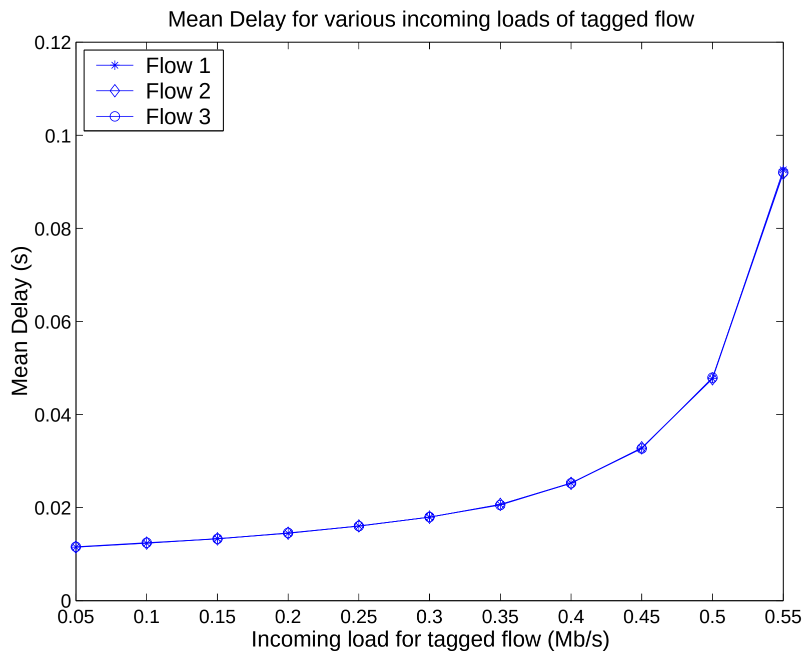


Figure 4.2.2.1 Mean Delay for HF^{2}Q: Exp. 2

Case 2:

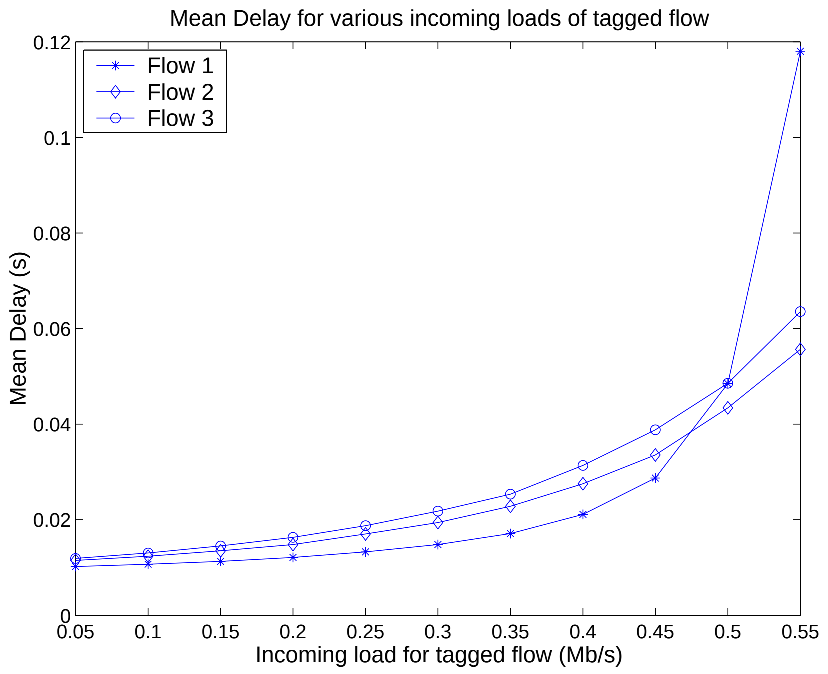


Figure 4.2.2.2 Mean Delay for WFQ: Exp. 2

Analysis:

In this experiment we see that HF^{2}Q exhibits the FIFO behavior irrespective of what the reservations of the individual flows are. In this case, the tagged flow eventually exceeds its reservation of 0.5 Mbps. We see that WFQ on the other hand provides delays to the flows according to their individual bandwidth reservations. Flow 1 which has a reservation of 0.5 Mbps gets the least delay so long as its incoming load is less than its reservation. Flow 2 is seen to get a lower delay than flow 3 since its reservation is higher and both offered loads are below their respective reservations.

4.2.3 Experiment 3

In this experiment the three flows again have three different reservations, but flow 1 (tagged) has a small reservation (0.2Mbps), while flow 2 has a large reservation (0.7Mbps) and flow 3 has a very small reservation (0.1Mbps) The experiment has two parts, both with the same weights but with different incoming loads for the three flows. Basically this experiment differs from experiment 2 in the fact that there is a vast difference between the reservation made by flow 2 and the reservations of flows 1 and 3. This experiment was basically performed to test the behavior of HF^{2}Q even if the bandwidth reservations are largely different and to ensure that it treats all the flows equally as FIFO does irrespective of the weights and incoming loads.

4.2.3.1 Part 1

In part 1 of experiment 3 the load of the tagged flow varies from 0.05 Mbps to 0.31 Mbps; again the tagged flow exceeds its reservation. The incoming loads of flows 2 and 3 are very close to their reservation. The experiment was repeated for 7 different tagged flow load conditions till the total system load was 0.99 Mbps. The table 4.2.3.1.1 summarizes the flow setup for this experiment.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1(tagged)	0.2	0.05-0.31
2	0.7	0.6
3	0.1	0.08

Table 4.2.3.1.1 Flows setup: Exp. 3

Case 1:

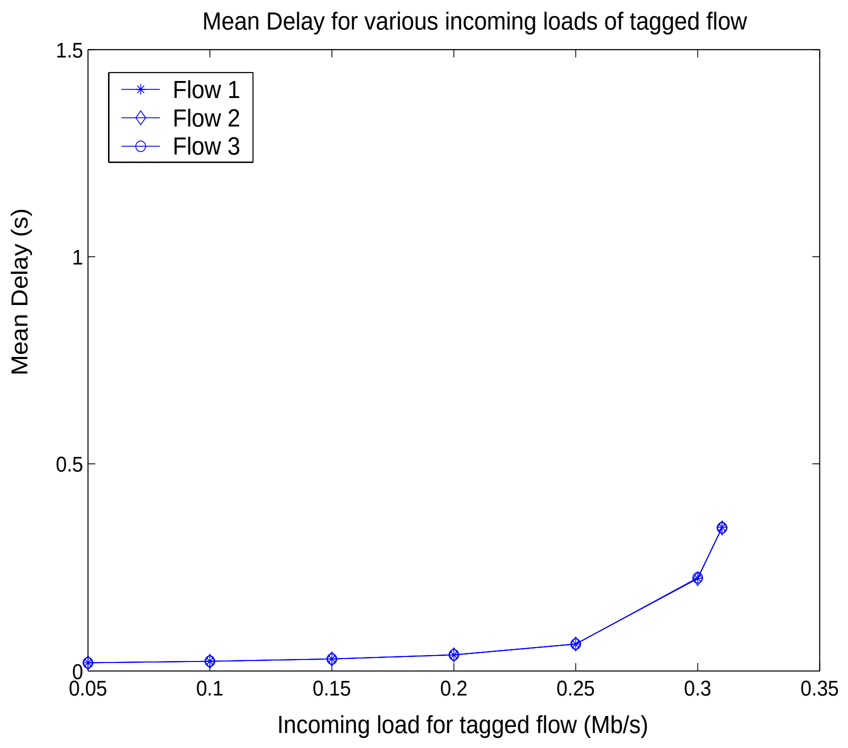


Figure 4.2.3.1.1 Mean Delay for HF^{2}Q: Exp. 3

Case 2:

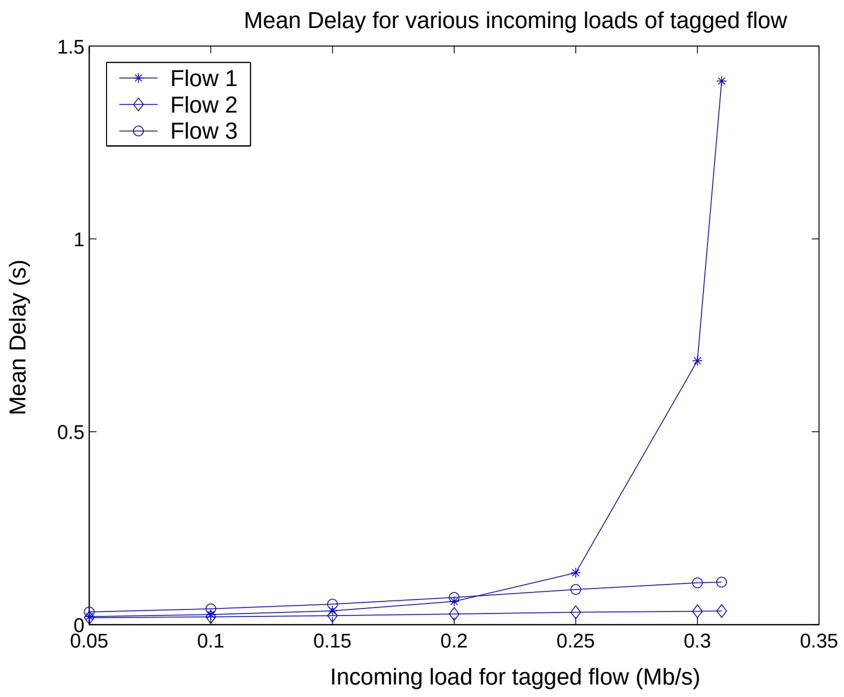


Figure 4.2.3.1.2 Mean Delay for WFQ: Exp. 3

Analysis:

This experiment basically demonstrates that HF^{2}Q behaves as FIFO irrespective of how different the reservations and the incoming loads of the different flows are. For WFQ flow 2 gets the lowest delay throughout since it has the highest reservation of 0.7 Mbps, even if its incoming load is much higher, but well within its reservation. Flow 1 gets lower delays compared to flow 3 since its reservation is higher than the other flows. But after a load of 0.2 Mbps, flow 3 gets lower delay since flow 1 has begun to exceed its reservation and WFQ does not guarantee lower delays for misbehaving flows.

4.2.3.2 Part 2

In part 2 the non-tagged flows though having different reservations, have equal incoming loads. The incoming load of flow 3 is close to its reservation while that of flow 2 is far less. The load of the tagged flow is increased from 0.05 to 0.8 Mbps. Hence the tagged flow exceeds its reservation to a great extent. The experiment was repeated for 16 different load values of the tagged flow.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1(tagged)	0.2	0.05-0.8
2	0.7	0.09
3	0.1	0.09

Table 4.2.3.2.1 Flows setup: Exp. 3

Case 1:

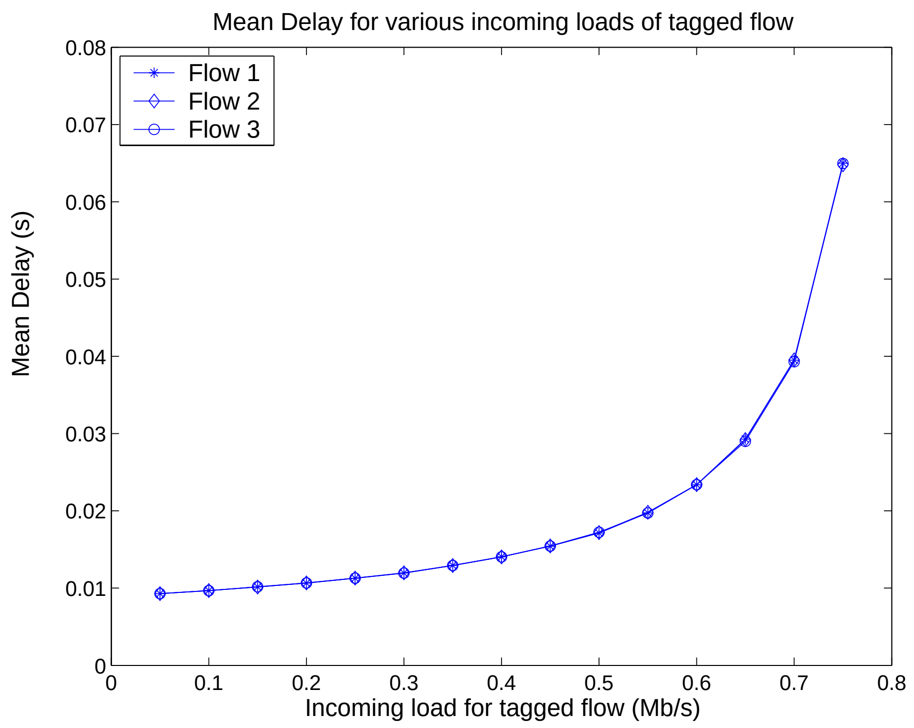


Figure 4.2.3.2.1 Mean Delay for HF^{2}Q: Exp. 3

Case 2:

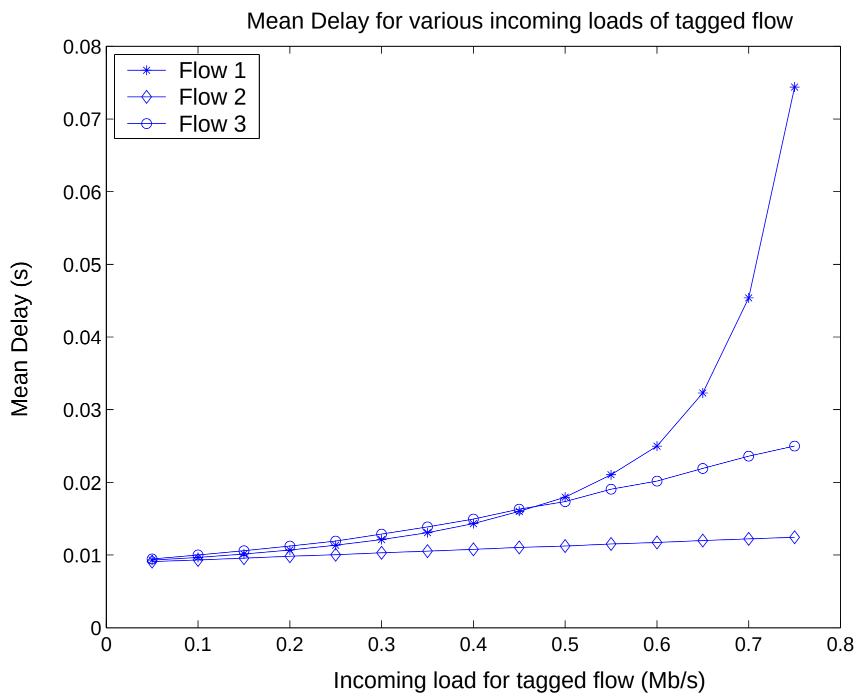


Figure 4.2.3.2.2 Mean Delay for WFQ: Exp. 3

Analysis:

In this case flow 2 and flow 3 have the same offered load of 0.09 Mbps, while the reservation of flow 2 is much higher than that of flow 3. Therefore we see a huge difference in the delays of flows 2 and 3 in the case of WFQ. At this point it is worth comparing the HF^{2}Q results for parts 1 and 2. Since HF^{2}Q basically behaves as FIFO, it is expected that for a given value of system load the delay value obtained from parts 1 and 2 (since the reservations are equal) must be equal for any flow. It is found from the experiment results that the delay values are in fact equal. For instance, for a system load of 0.78 Mbps flow 2 has a delay of 0.02328 s measured from results of part 1 (at a tagged flow load of 0.1 Mbps) and has a delay of 0.02339 s measured from results of part 2 (at a tagged flow load of 0.6 Mbps).

4.3 Mean Delay Measurements for total load > 1

The same experiments were repeated for an overloaded system, wherein the total load offered is greater than 1. Again a comparison was made between the performance of HF^2Q and WFQ. With the current Internet and broadband revolution, the network must handle increasingly large volumes of data, which can lead to network overload and signal degradation. Hence queuing algorithms that control the order in which packets are sent, and the usage of the gateway buffer space, and the way in which packets from different sources interact play an important part in congestion control. Hence it is important that the performance of this new queuing algorithm under overloaded conditions is characterized and a comparison made to the existing WFQ. In all of these experiments two sets of results for the HF^2Q and WFQ are shown – one set (shown in case 1) that includes the all the delay values of the flows including the values for a total system load of 1, and another set (shown in case 2) that does not show the values of load for which the system exhibits a transitional behavior of HF^2Q. In all the results shown below, the delay of the tagged flow, which rises infinitely, is not plotted. In the following description, the experiments correspond to those described in section 4.2 (underloaded case).

4.3.1 Experiment 1

The load of the tagged flow is increased from 0.6 Mbps (100% system load) to 0.8 Mbps (120% system load), while the load of the other two flows are maintained below their reservations. The experiment was repeated for 11 different loads of the tagged flow. Table 4.3.1.1 summarizes the flow setup.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1	0.6	0.6 – 0.8
2	0.2	0.15
3	0.2	0.15

Table 4.3.1.1 Flows setup: Exp. 1

Case 1:

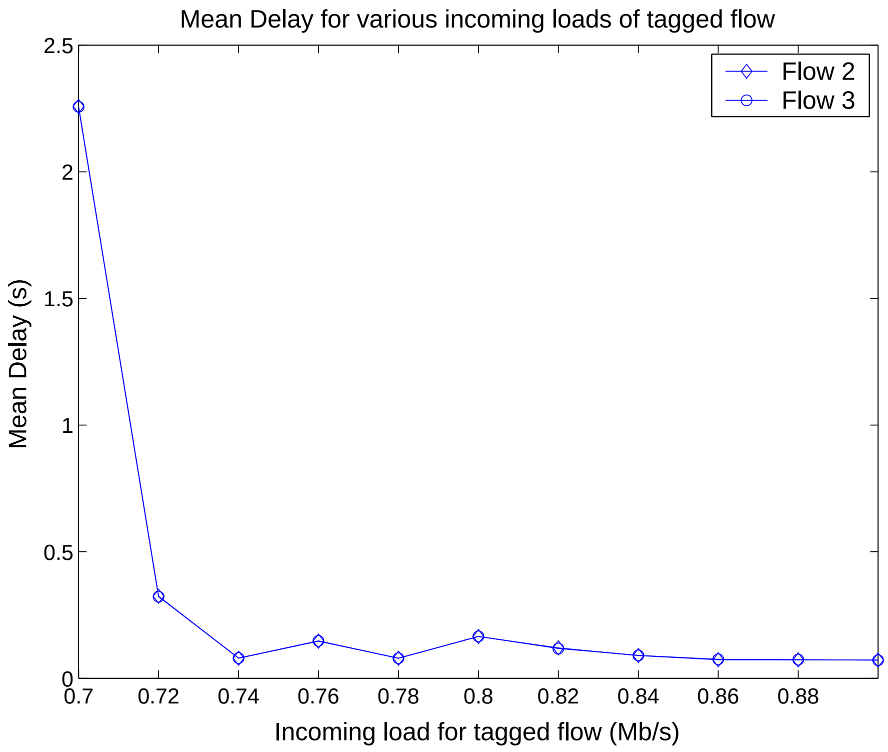


Figure 4.3.1.1 Mean Delay for HF^{2}Q: Exp. 1

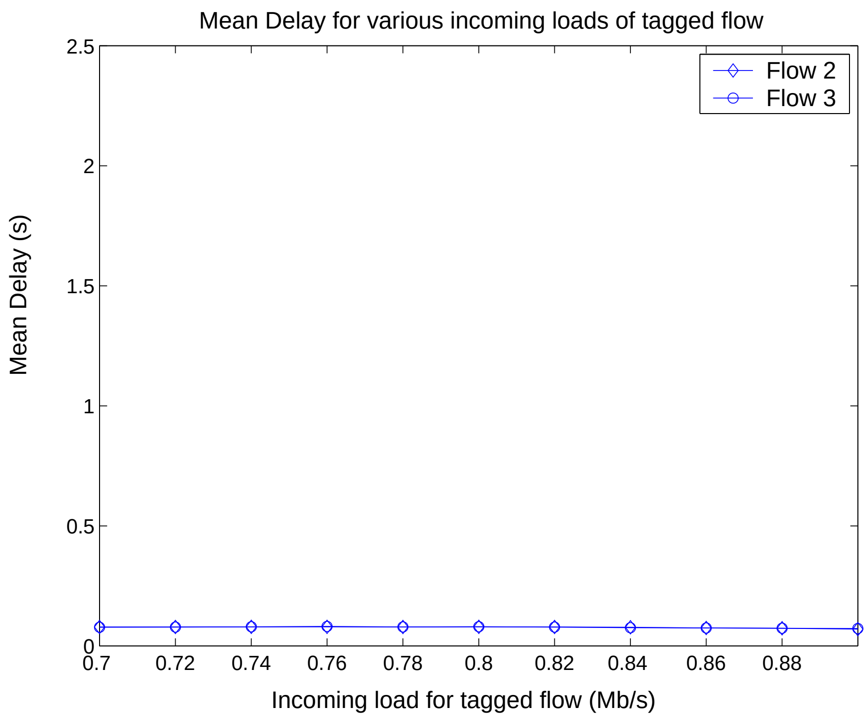


Figure 4.3.1.2 Mean Delay for WFQ: Exp. 1

Case 2:

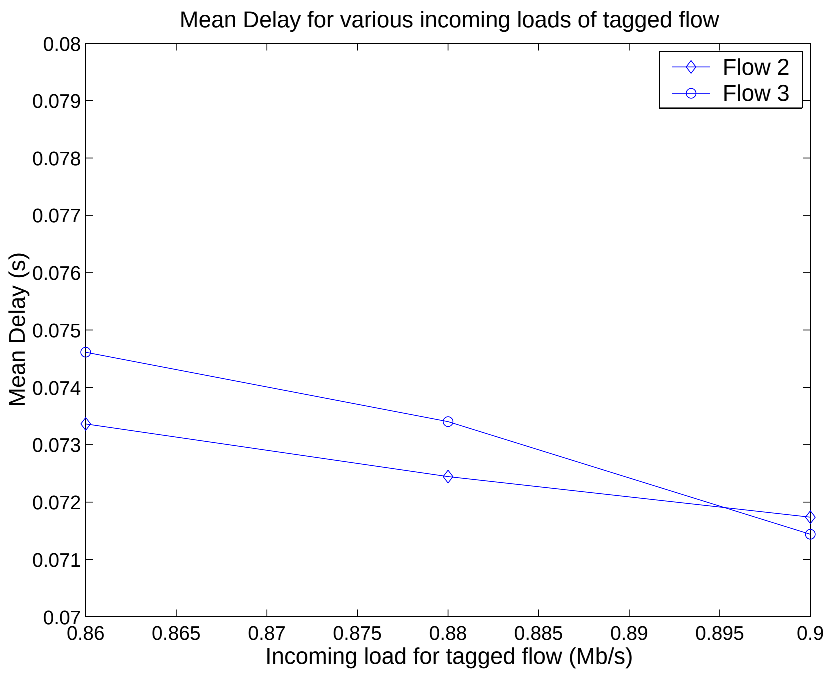


Figure 4.3.1.3 Mean Delay for HF^{2}Q: Exp. 1

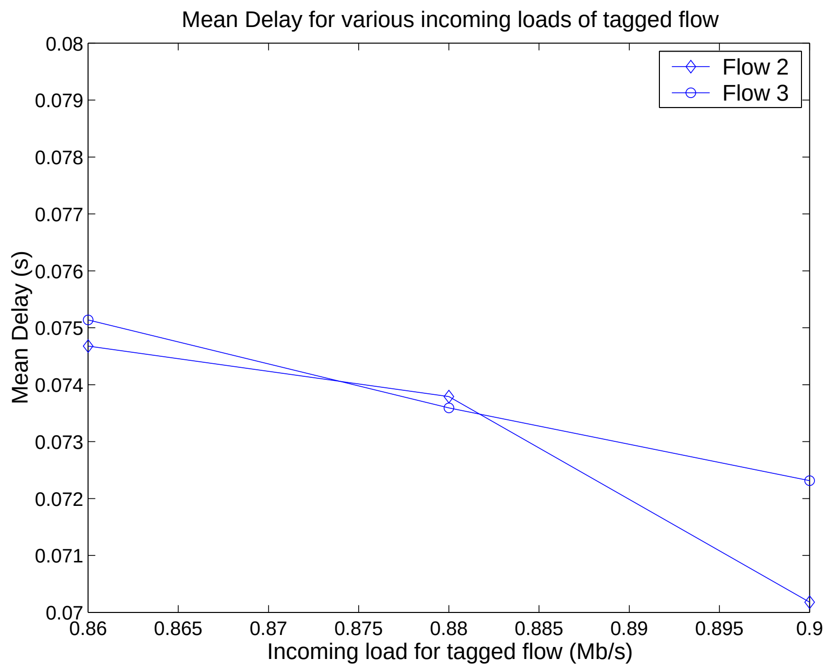


Figure 4.3.1.4 Mean Delay for WFQ: Exp. 1

Analysis:

We see that when the system is overloaded HF^{2}Q exhibits the properties of WFQ but only after a transitional period, i.e. till a total system load of approximately 110% (in this case). But we see that when the system is heavily overloaded (a total system load of 112%), the delays in the case of HF^{2}Q and WFQ are essentially the same (Figures 4.3.1.3 and 4.3.1.4). Thus HF^{2}Q offers the same protection as WFQ does to the behaving flows even when the system is overloaded. We observe that both flow 2 and flow 3 have equal reservations and equal offered loads and hence their delays are approximately equal. It is the misbehaving flow whose delay rises infinitely, whereas the delays of the other flows remains bounded. Possible reasons as to why HF^{2}Q exhibits the transitional (compare Figure 4.3.1.1 and 4.3.1.2) behavior are presented in the section 4.5. As for the underloaded case the detailed results for similar experiments are presented in Appendix 1.

4.3.2 Experiment 2

In this experiment the load of the tagged flow is varied from 0.6 Mbps (100% system load) to 0.8 Mbps (120 % total system load). However the non-tagged flows are well within their respective reservations. Table 4.3.2.1 summarizes the flows setup for this experiment. The experiment was repeated for 11 different tagged flow load values.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1	0.5	0.6 – 0.8
2	0.3	0.25
3	0.2	0.15

Table 4.3.2.1 Flows setup: Exp. 2

Case 1:

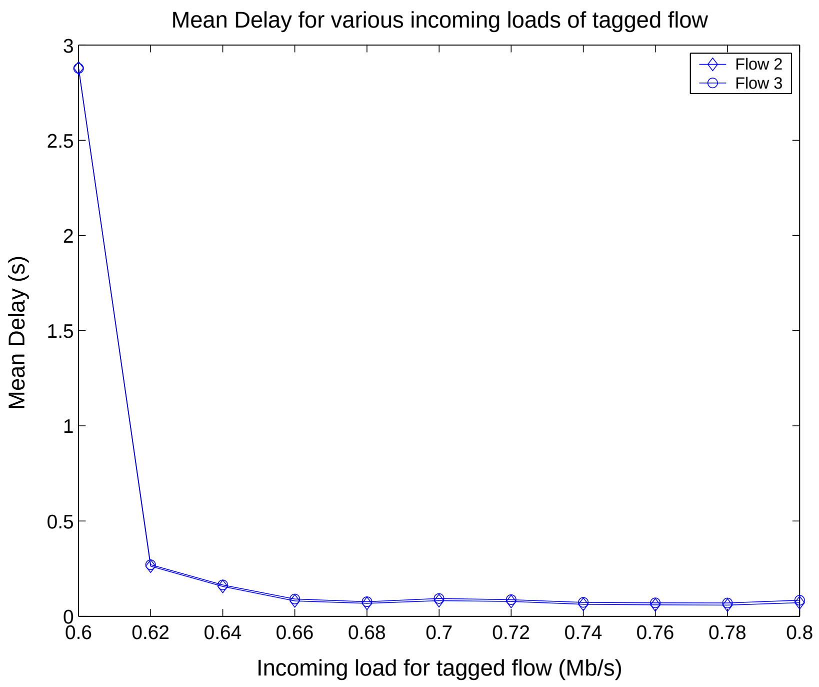


Figure 4.3.2.1 Mean Delay for HF^{2}Q: Exp. 2

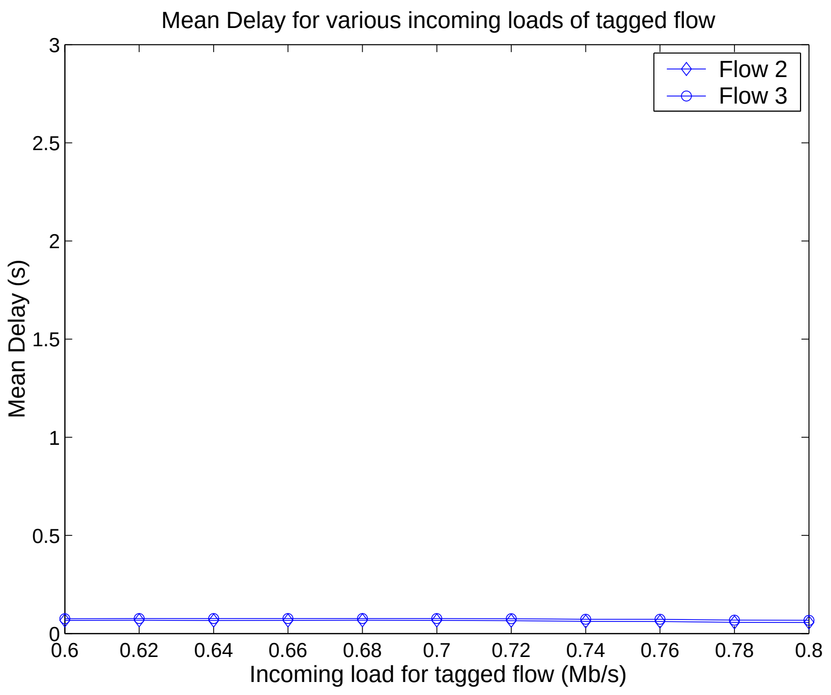


Figure 4.3.2.2 Mean Delay for WFQ: Exp. 2

Case 2:

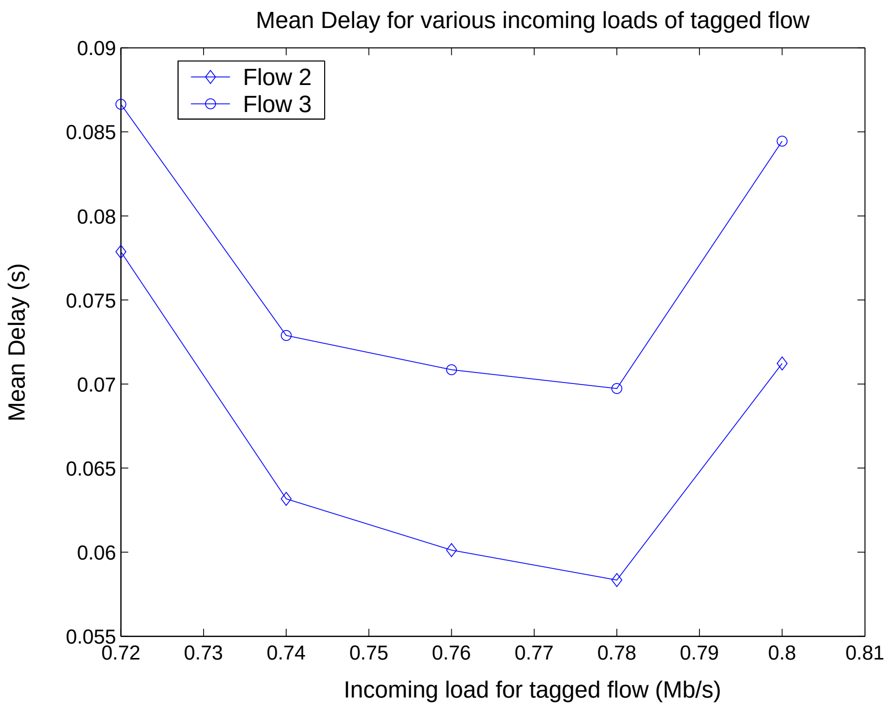


Figure 4.3.2.3 Mean Delay for HF^{2}Q: Exp. 2

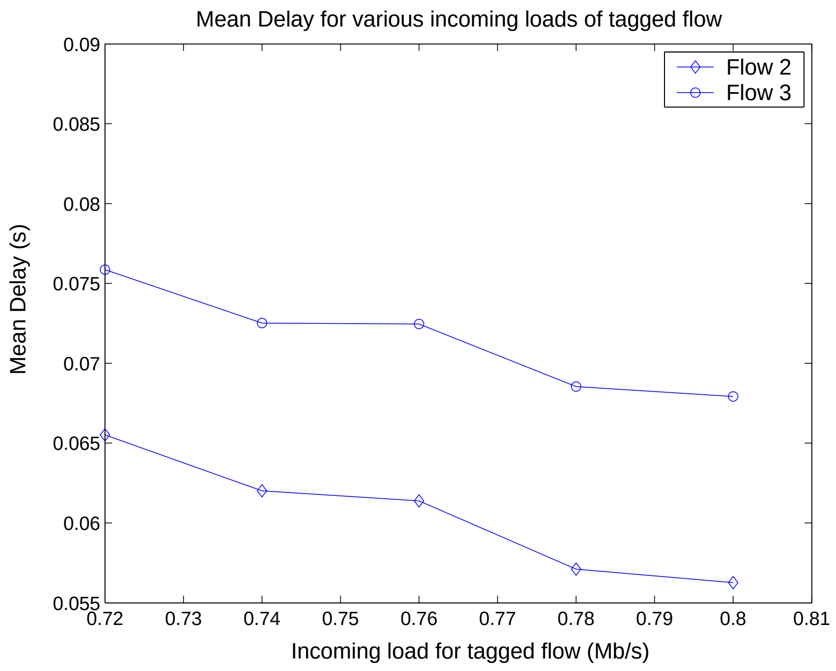


Figure 4.3.2.4 Mean Delay for WFQ: Exp. 2

Analysis:

In this experiment, after the transitional period, we see that HF^{2}Q behaves as WFQ. The transitional period exists till a total system load of 114% (incoming load of tagged flow is 0.74 Mbps). After that we again see similar behavior for HF^{2}Q and WFQ. Both offer lower delay to flow 2 as compared to flow 3 since flow 2 has a greater reservation and their delays are bounded since their incoming loads are well within their respective reservations.

4.3.3 Experiment 3

In this experiment the three flows have three different reservations. The experiment has two parts, both with the same weights but with different incoming loads for the three flows.

4.3.3.1 Part 1

In this part the load of the tagged flow is increased from 0.32 Mbps (100% total system load) to 0.5 Mbps (115% total load). Both non-tagged flows have incoming loads close to their reservations. The table 4.3.3.1 summarizes the flows setup. The experiment was repeated for 10 different tagged flow loads.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1(tagged)	0.2	0.32-0.5
2	0.7	0.6
3	0.1	0.08

Table 4.3.3.1.1 Flows setup: Exp. 3

Case 1:

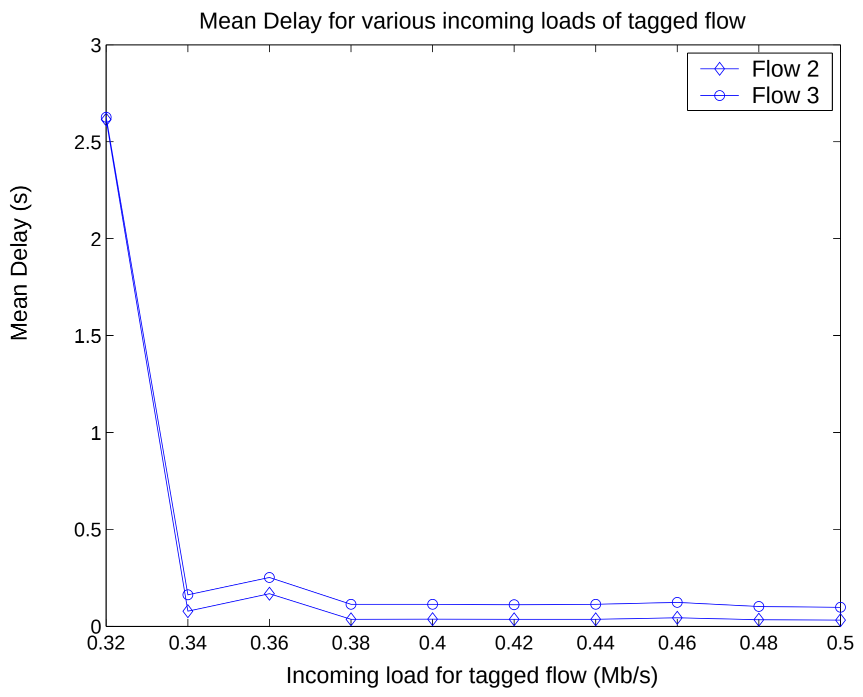


Figure 4.3.3.1.1 Mean Delay for HF^{2}Q: Exp. 3

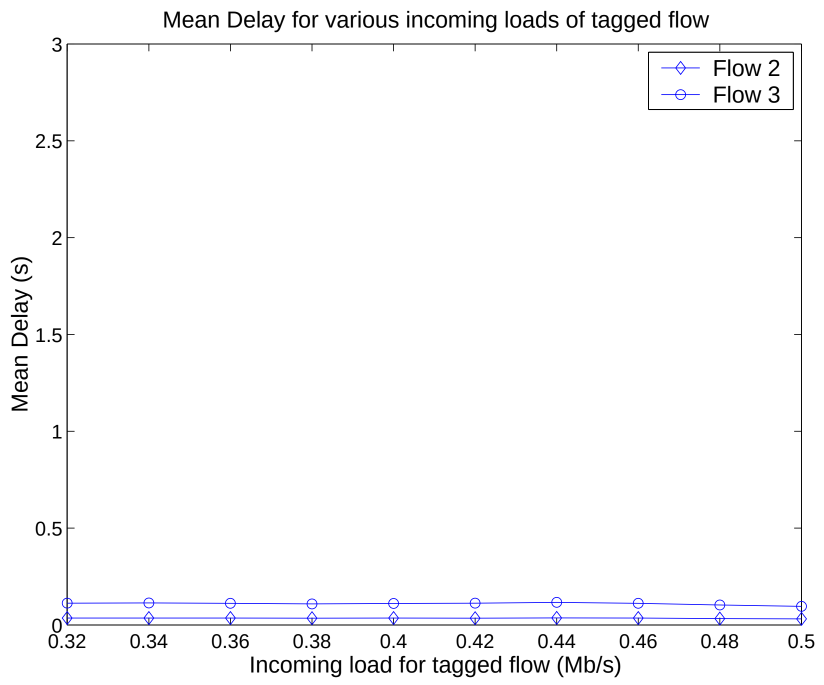


Figure 4.3.3.1.2 Mean Delay for WFQ: Exp. 3

Case 2:

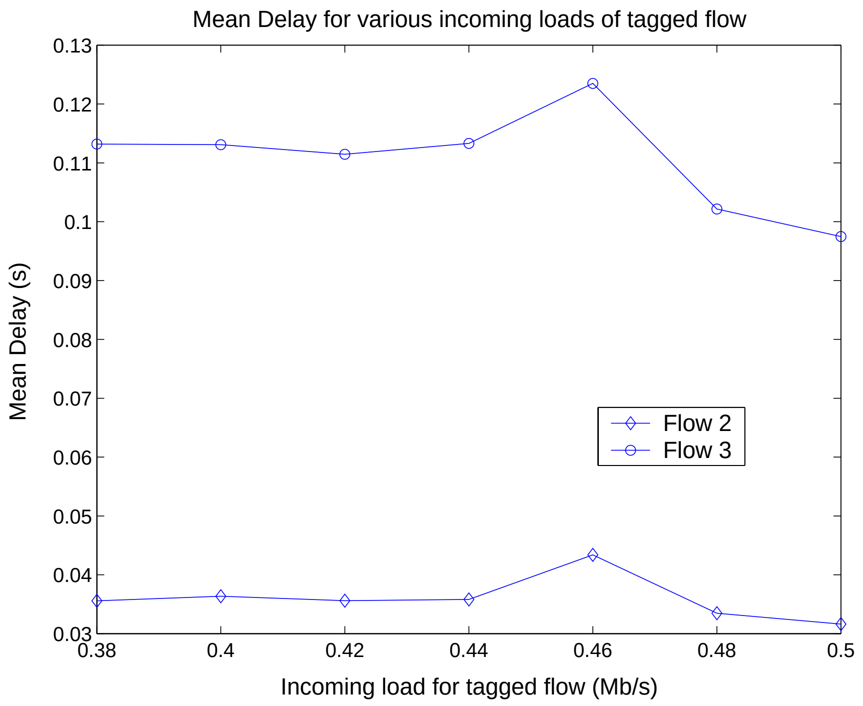


Figure 4.3.3.1.3 Mean Delay for HF^{2}Q: Exp. 3

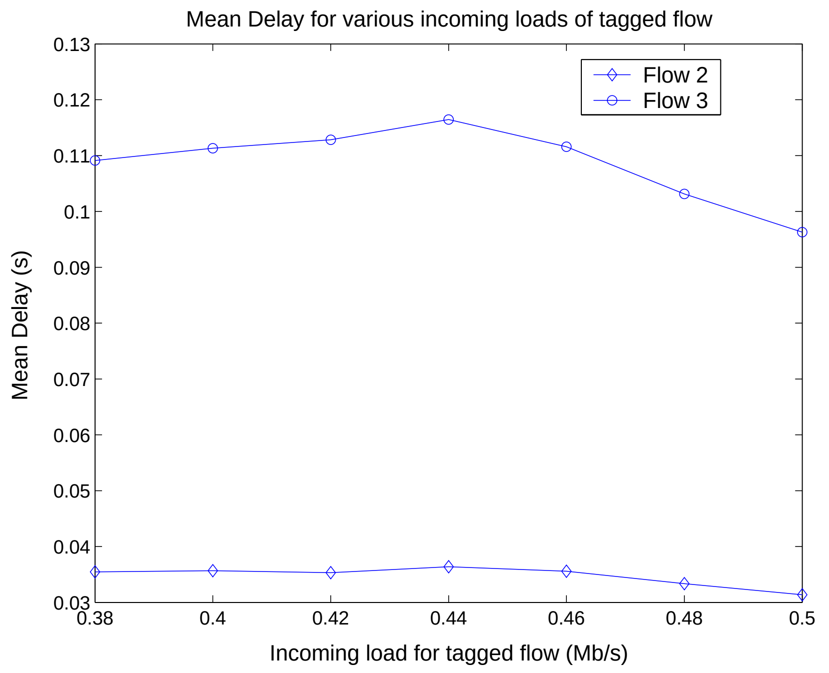


Figure 4.3.3.1.4 Mean Delay for WFQ: Exp. 3

Analysis:

In this experiment the incoming loads of both flow 2 and flow 3 are close to their respective reservations. We see that in this experiment the transitional behavior for HF^{2}Q exists till a total system load of 106% (i.e. when in the incoming load of the tagged flow is 0.38 Mbps). Again we see in both cases that flow 2 gets a much lower delay compared to flow 3 since its reservation is much higher. The difference in average delays of flow 2 and flow 3 is approximately 80ms whereas in the previous experiment when the reservations were close enough (flow 2 had a reservation of 0.3 Mbps and flows 3 had a reservation of 0.2 Mbps) the difference in average delays was approximately 10ms.

4.3.3.2 Part 2

In this part the load of the tagged flow is increased from 0.82 Mbps (100% total system load) to 0.98 Mbps (116% total load). The system is more heavily loaded than the previous part. The tagged flow heavily exceeds its reservation. Both the non-tagged flows have equal reservations. The table 4.3.3.2 summarizes the flows setup. The experiment was repeated for 9 different tagged flow loads.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1(tagged)	0.2	0.82-0.98
2	0.7	0.09
3	0.1	0.09

Table 4.3.3.2.1 Flows setup: Exp. 3

Case 1:

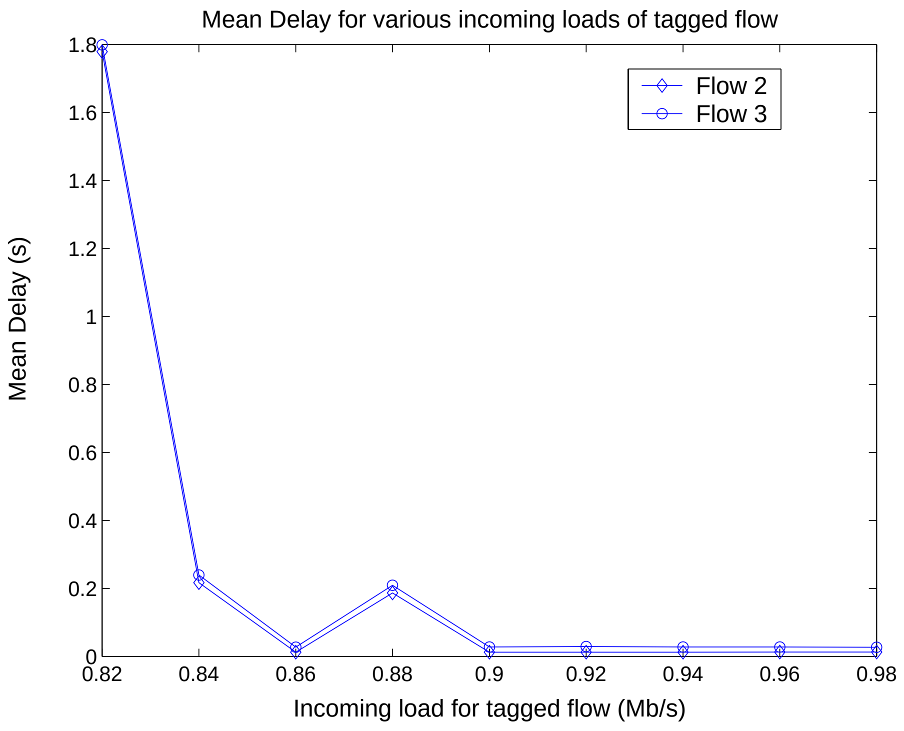


Figure 4.3.3.2.1 Mean Delay for HF^{2}Q: Exp. 3

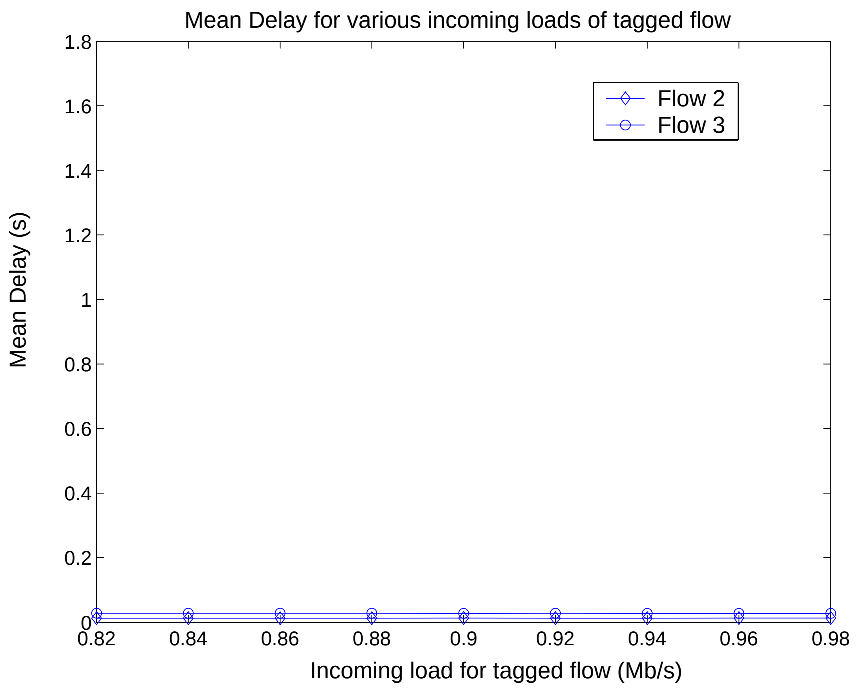


Figure 4.3.3.2.2 Mean Delay for WFQ: Exp. 3

Case 2:

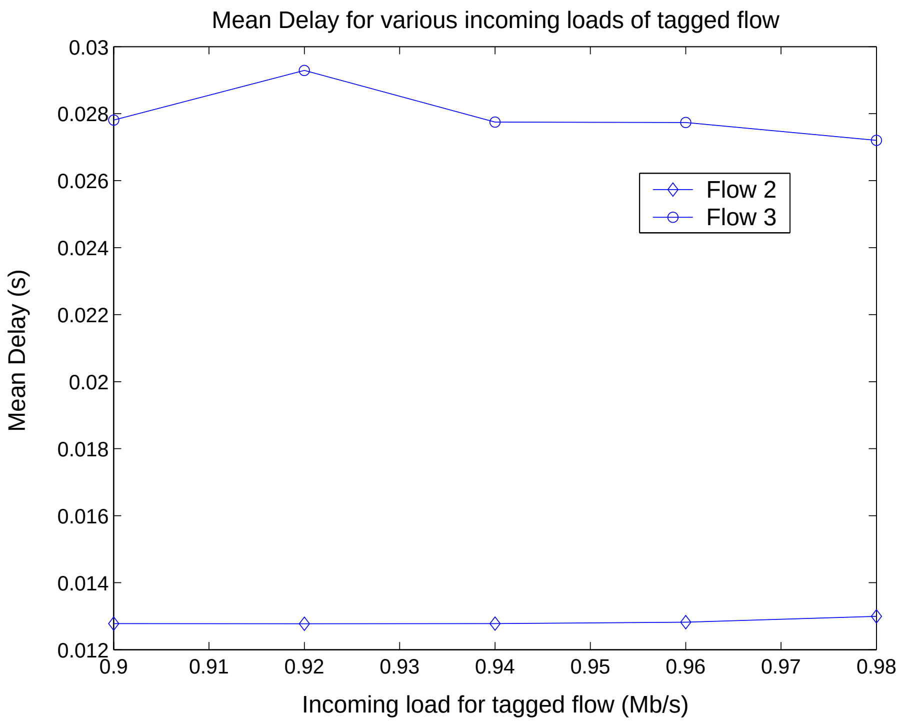


Figure 4.3.3.2.3 Mean Delay for HF^{2}Q: Exp. 3

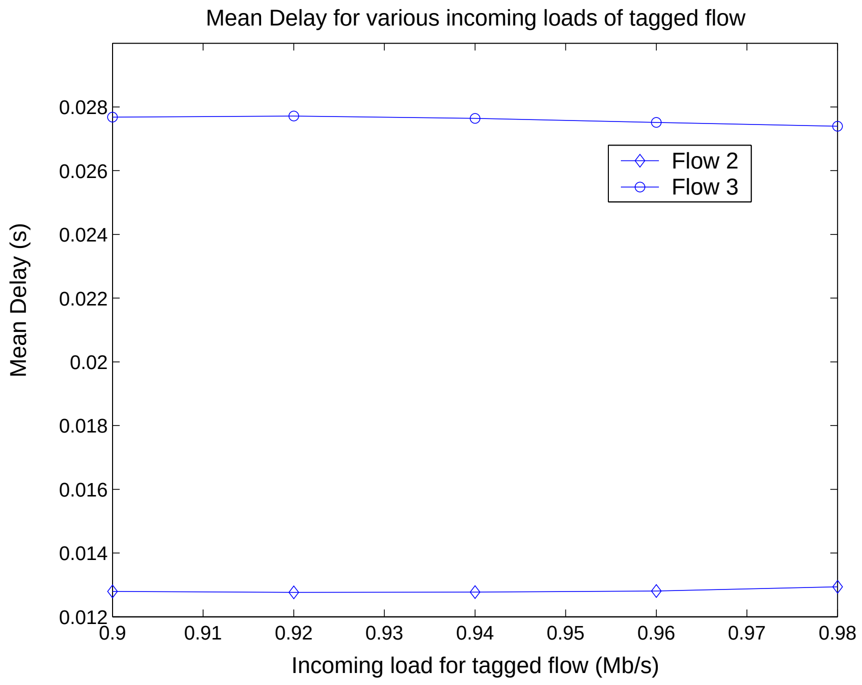


Figure 4.3.3.2.4 Mean Delay for WFQ

Analysis:

In this experiment the reservations are the same as the previous part, but both flow 2 and flow 3 have equal incoming loads of 0.09 Mbps (the incoming load of flow 2 is much less than its reservation). This scenario was basically set up to verify that performance of HF^{2}Q in all possible combination of reservation and incoming load. We again see that after the transient period at a total system load of 108%, both HF^{2}Q and WFQ exhibit the same behavior. Again in this case the delay of flow 2 is less compared to that of flow 3.

4.4 Throughput Measurements

There are a number of definitions for throughput, but here in the context of queuing systems throughput is defined as the fraction of link capacity used to carry packets. Throughput is an important performance measure of queuing systems and in turn an element of network performance within the scope of QoS since it quantifies the number of packets transmitted successfully. It is even more significant in the context of fair queuing systems since it is characterizes the amount of bandwidth actually allocated to each flow and it is interesting to see the throughput variation of each flow both when the network is underloaded and overloaded. The following sections present the throughput results for both HF^2Q and WFQ and a comparison is made with the throughput characteristics FIFO provides; case 1 presents the throughput results for HF^2Q and case 2 for WFQ and case 3 presents the corresponding FIFO results. The plots in the following sections show the throughput normalized with respect to the channel capacity. The simulation setup was same as that of the delay measurement experiments. The throughput results for both the underloaded and overloaded cases are shown in the same plot. As before the first flow in all the experiments is the tagged flow.

4.4.1 Experiment 1

The tagged flow load is varied from 0.05 Mbps (5% load) to 0.8 Mbps (112% total load). The non-tagged flows have arrival rates less than their reservations. The experiment was repeated for 21 different load values of the tagged flow. Table 4.4.1.1 summarizes the flows setup.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1(tagged)	0.5	0.05-0.8
2	0.3	0.25
3	0.2	0.15

Table 4.4.1.1 Flows setup: Exp. 1

Case 1:

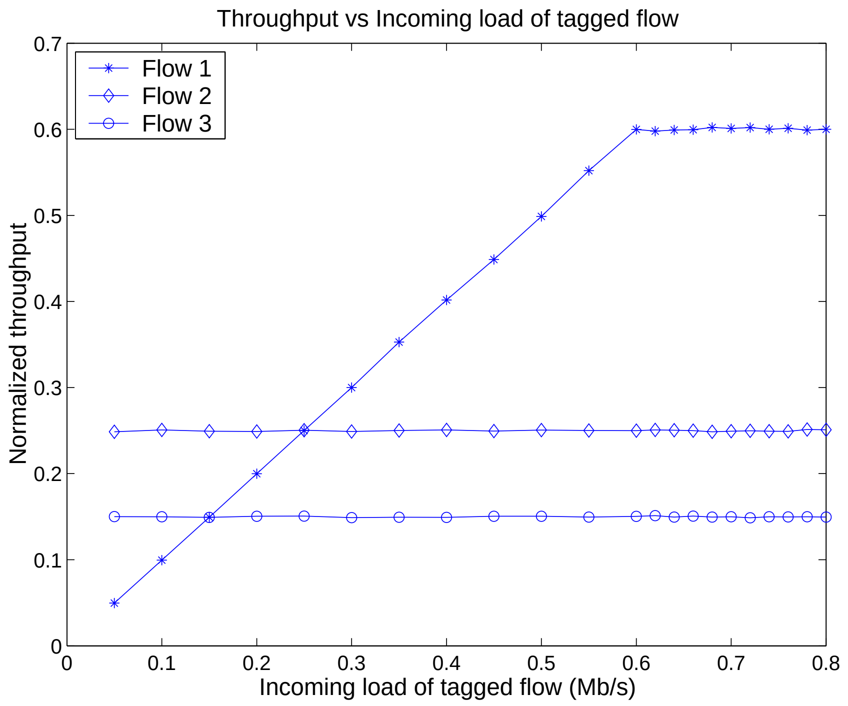


Figure 4.4.1.1 Throughput for HF^{2}Q: Exp. 1

Case 2:

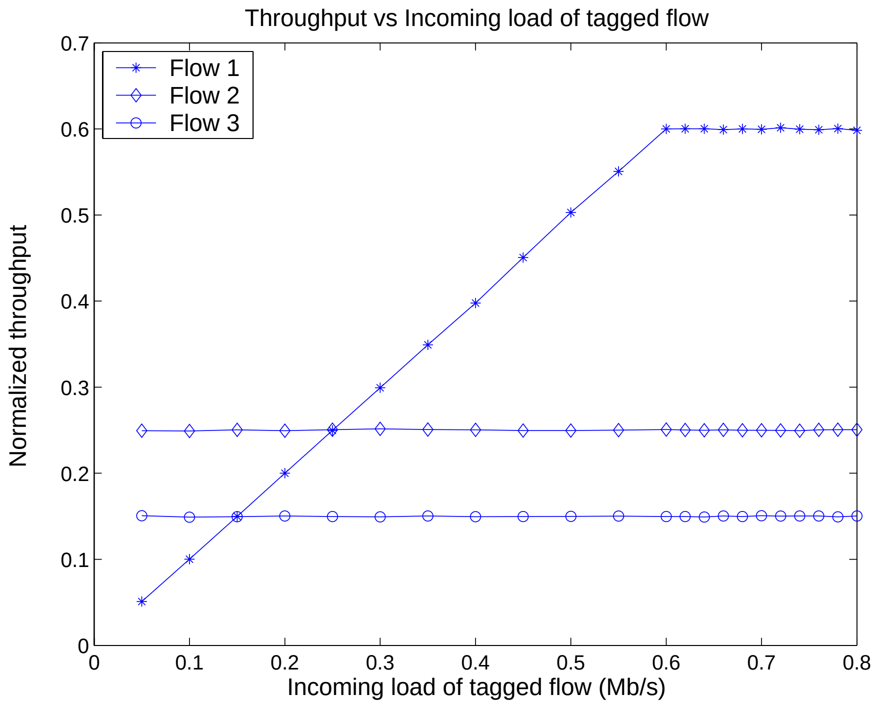


Figure 4.4.1.2 Throughput for WFQ: Exp. 1

Case 3:

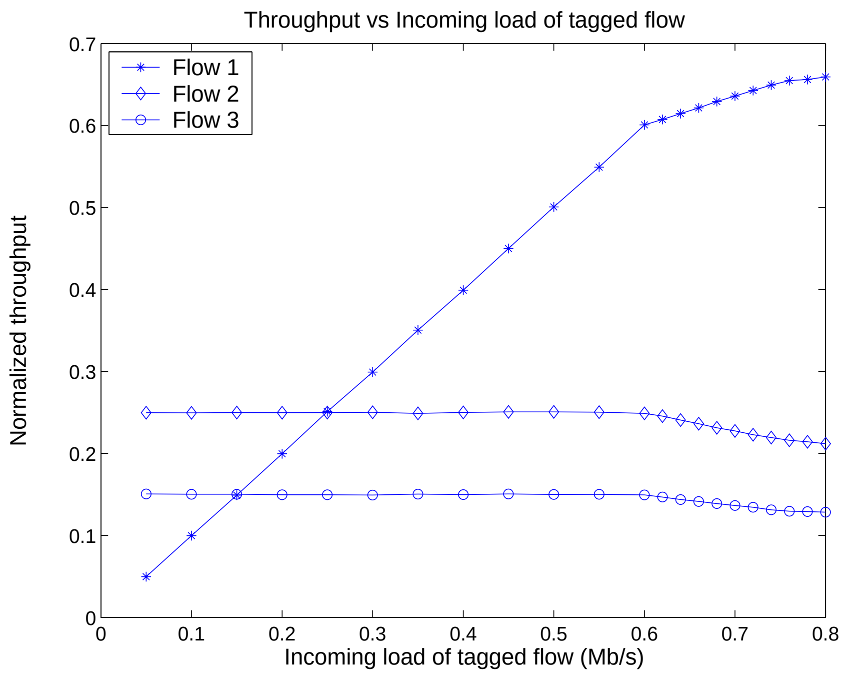


Figure 4.4.1.3 Throughput for FIFO: Exp. 1

Analysis:

It can be easily observed that when the system is underloaded, there is bandwidth available and hence with both HF^{2}Q and WFQ there is no packet loss and the normalized throughput is the same as the incoming load of the individual flows. We see that, for the tagged flow whose load is increasing, the throughput linearly increases with the offered load. Also the throughput of the other 2 flows that have constant arrival rates remains a straight line, equal to their respective incoming loads at all times. When the system is overloaded, the throughput of flows 2 and 3 remain unaffected since they are well within their reservations. The buffer length was increased sufficiently so that there is no packet loss due to buffer overflow. The throughput of the tagged flow, which eventually exceeds its reservation of 0.5 Mbps, is capped at the value of capacity unused by flows 2 and 3 (0.6 Mbps). The tagged flow thus gets penalized since its packets get dropped and thus its delay increases without bound since the buffer is always full. Comparing the HF^{2}Q and WFQ throughput characteristics with that of FIFO (figure 4.4.1.3), we see that the three service disciplines are identical when the total system load is less than 100%. This is because for any work-conserving service discipline, throughput equals offered load when the total load is less than 1. But when the system is overloaded FIFO does not offer fair treatment to the well behaving flows (flows 2 and 3), which are well within their reservation. We can observe from figure 4.4.1.3 that when the system is overloaded the throughput of flows 2 and 3 drops and the throughput for flow 1 is not linear (lesser than offered load), i.e. packets get dropped. Thus we see that FIFO penalizes flows 2 and 3 when flow 1 is misbehaving, whereas in the case of HF^{2}Q and WFQ flows 2 and 3 have throughputs equal to their offered loads.

4.4.2 Experiment 2

The tagged flow load is varied from 0.05 Mbps (5% load) to 0.7 Mbps (116% total load). In this experiment both non-tagged flows exceed their reservations. The experiment was repeated for 14 different load values of the tagged flow. Table 4.4.2.1 summarizes the flows setup.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1(tagged)	0.5	0.05-0.7
2	0.3	0.4
3	0.2	0.5

Table 4.4.2.1 Flows setup: Exp. 2

Case 1:

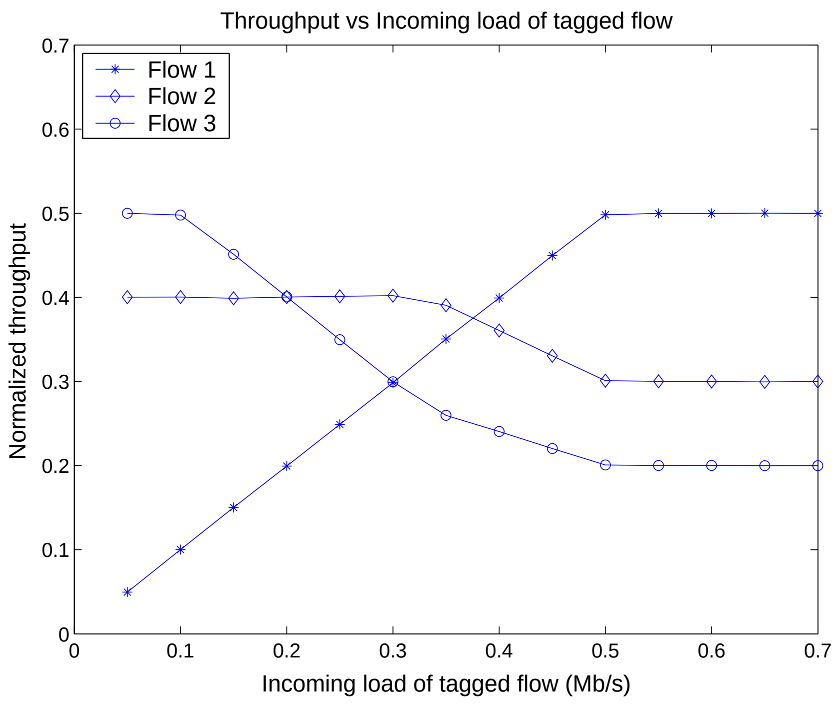


Figure 4.4.2.1 Throughput for HF^{2}Q: Exp. 2

Case 2:

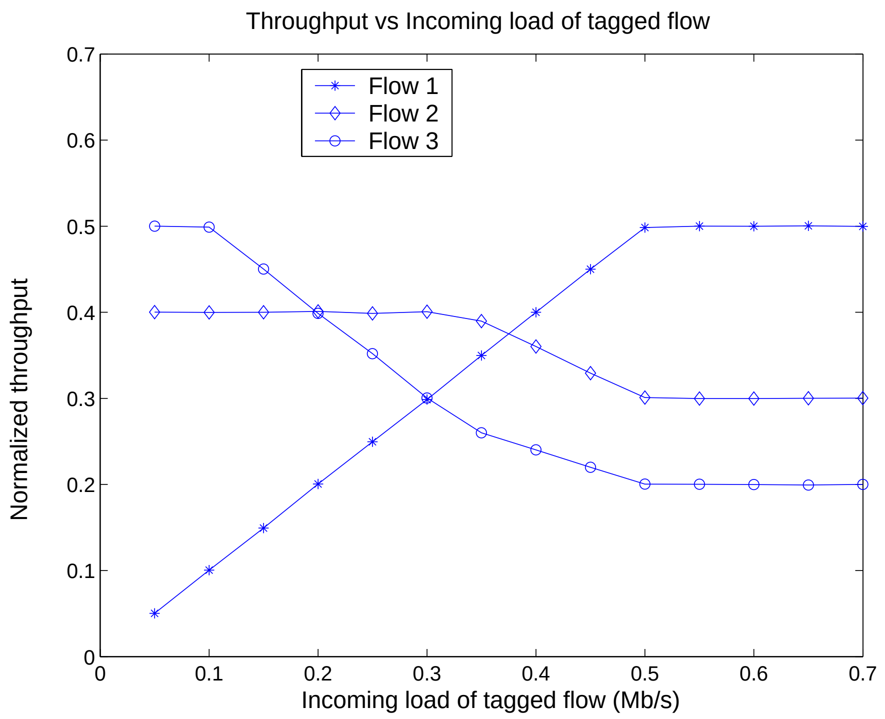


Figure 4.4.2.2 Throughput for WFQ: Exp. 2

Case 3:

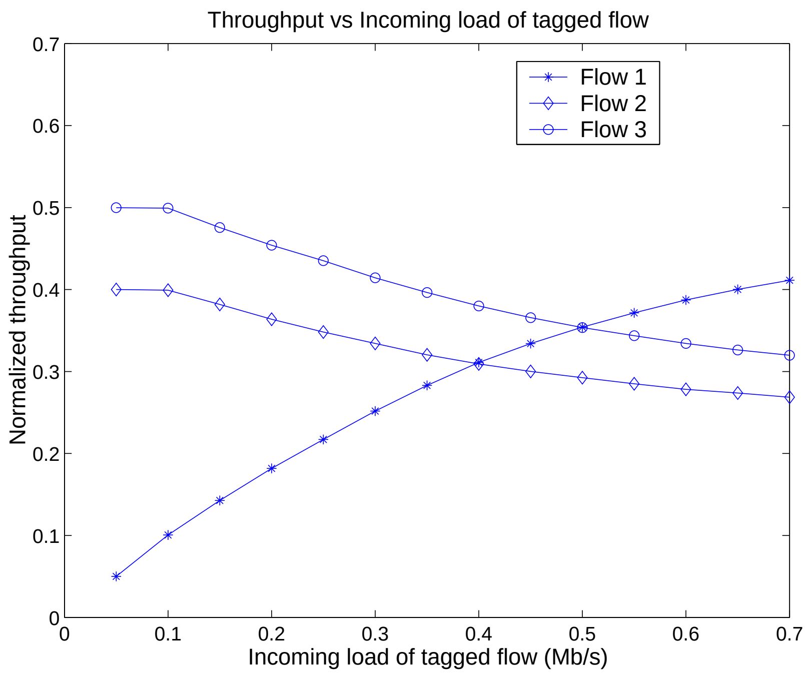


Figure 4.4.2.3 Throughput for FIFO: Exp. 2

Analysis:

This is a very interesting scenario for studying the throughput behavior for both HF^{2}Q and WFQ. When the system is overloaded HF^{2}Q and WFQ exhibit the same behavior. When the system is underloaded, bandwidth is available and both exhibit the same throughput behavior again even though HF^{2}Q exhibits the delay properties of FIFO. In this experiment both flows 2 and 3 exceed their respective reservations. We observe that till a total system load of 100% (i.e. incoming load of flow 1 is 0.1Mbps) the throughput of the three flows are equal to their respective incoming loads. But when the system is overloaded the fair bandwidth allocation is explicitly noticeable. We see that flow 3, which has the least reservation amongst all the three flows, is penalized first, i.e. its throughput starts dropping from 0.5. The throughput of flow 1 is linearly increasing as expected, so long as it is within its reservation. Also the throughput of flow 2, which has a higher reservation remains constant at 0.4 Mbps, till the incoming load of the tagged flow is 0.3 and then it starts dropping down. The throughput of both flow 2 and 3 drops so that the total throughput is 1. The throughput of flow 1 increases linearly till 0.5 and then saturates. This is ideal fair bandwidth allocation. Again comparing the results with those of FIFO we see that when the system is overloaded (after a tagged flow load of 0.1 Mbps) we see that FIFO essentially allocates bandwidth depending on demand, and hence buffer occupancy. There are no reservations with FIFO.

4.5 Conclusions drawn

From the results presented above we can definitely conclude that HF^{2}Q exhibits FIFO properties for a system whose total load is less than 1 and behaves as WFQ for a severely overloaded system. For a moderately overloaded system, however, HF^{2}Q exhibits transitional behavior that us neither FIFO nor WFQ. HF^{2}Q is "fair" in one sense since it offers equal delays to all flows as FIFO does (for system load less than 1) and is also "fair" in the bandwidth allocating sense like WFQ since it protects the well-behaving flows when the system is heavily overloaded.

We offer some intuitive explanation as to why HF^{2}Q behaves the way it does in two different scenarios, when the total system load is less than 1 and when the system load exceeds 1. As explained in chapter 3, the key in the implementation of HF^{2}Q is not updating the *lastRUpdateTime* variable when the first packet enters the system after a server idle period. It should be noted that the busy and idle periods of the reference GPS system and the packetized system coincide. But when the system is heavily overloaded, there are no idle periods and hence the buffers are never empty since the server is work conserving. Hence the scenario for updating the lastRUpdateTime never occurs, and the *roundNumber* gets updated as it would in WFQ. Hence HF^{2}Q behaves as WFQ in a heavily overloaded system.

But when the system load is less than one, after an idle period (when none of the flows are backlogged) and a packet arrives, due to the modification in the *lastRUpdateTime* the virtual time value is different than what it is in actual WFQ. Hence the ordering of packets also changes, but the most amazing and interesting aspect is that it should result in FIFO ordering. The fact that packets are actually getting serviced in ordered of arrivals was verified in the simulation by assigning packet sequence numbers upon arrival and checking if they get serviced in FIFO order. It is also worth mentioning at this point that only packets that arrive to an empty queue get the virtual time assigned as their finish times. So even in an underloaded system there is a small initial period of time when the ordering is not FIFO. This is because the FIFO ordering sets in only after a server idle period. The time for which the ordering is not FIFO has been quantified and is presented as a ratio of the total simulation time (as shown in the table 4.5.1). The total simulation time for each run for the above experiment was 6000s. After this small period of time, the ordering of packets is FIFO until a system load of 100% is reached.

LOAD (Mbps)	FRACTION OF MISORDERINGS	TIME FRACTION
0.35	0	0
0.4	0	0
0.45	5.9262e-006	1.77e-05
0.5	5.9262e-006	2.133e-05
0.55	9.6526e-006	1.9e-05
0.6	1.1099e-005	4.44e-05
0.65	0	0
0.7	7.6015e-006	4.57e-05
0.75	3.5636e-006	7.11e-06
0.8	5.9842e-005	2.83e-04
0.85	4.7061e-006	1.25e-05
0.9	5.9072e-006	1.48e-05
0.95	1.6851e-005	1.26e-04

Table 4.5.1 Quantification of transitional behavior of HF^{2}Q

It is also evident from the results that there is a transitional period when the system is slightly overloaded i.e. the system load is just above 100%, during which the HF^{2}Q system doesn't exactly behave as WFQ does. This because the arrivals are exponential and hence there may be very small amounts of time during which the server is idle. Hence when the system is heavily overloaded, there is definitely no server idle periods and WFQ behavior sets in.

5 DOCSIS Performance Evaluation

As mentioned in Chapter 1, the objective of this thesis was two-fold, one to characterize the performance of a new queuing discipline and the other to evaluate the performance of DOCSIS 1.1. Chapters 3 and 4 presented a detailed evaluation of HF^{2}Q. This chapter presents a detailed performance evaluation of DOCSIS 1.1 in the context of BWA networks.

Hybrid Fiber Coax (HFC) cable networks have been used in the past primarily to deliver broadcast-quality TV signals to homes. The wide availability of such systems and their extremely wide bandwidth allows extending their functionality to deliver high-speed broadband data signals to end-users. Data over Cable System Interface Specifications (DOCSIS), a MAC protocol elaborated under the leadership of Cable Television Laboratories, Inc., has been established as the major industry standard for two-way communications over HFC cable plants. The first specifications, referred to as DOCSIS 1.0, have been largely deployed in cable networks. The second specifications (DOCSIS 1.1) are now in the early phase of deployment. The physical layer of DOCSIS 1.1 specifications is the same as that of DOCSIS 1.0. The difference between the two sets of specifications lies on the MAC layer, which includes Quality of Service (QoS) capabilities, in DOCSIS 1.1 specifications. This has been adopted as the *de facto* standard for delivering broadband services for HFC networks. Although DOCSIS is designed for interoperability among cable modems and related products, with a few modifications it can be used in wireless Multipoint, Multichannel Distribution Service (MMDS) and

Local Multipoint Distribution Service (LMDS) systems. Such Broadband Wireless Access (BWA) networks attempt to carry high speed Internet traffic to subscriber systems, are easy to implement and can be installed without extensive infrastructure. Hence, The IEEE802.16 standard developed for BWA systems was based on two HFC MAC protocols, IEEE 802.14a and DOCSIS. This is due to the striking similarities between the cable medium and the wireless environment.

5.1 DOCSIS 1.1 Overview

5.1.1 DOCSIS 1.1 Basics

DOCSIS 1.1 is an IP-centric point-to-multipoint standard that was developed for broadband Internet access applications over cable TV networks. DOCIS specifies both the physical layer and the Medium Access Control (MAC) layer. Its major components are the Cable Modems (CM) at the customer premises and the Cable Modem Termination System (CMTS) at the head-end of the cable plant. DOCSIS 1.1 evolved from the DOCSIS 1.0 specs, but with Quality of Service (QoS) mechanisms and algorithms implemented at the MAC layer. To summarize, DOCSIS 1.1 builds upon 1.0, but also includes the following features:

- Quality of Service
- Dynamic Services
- Concatenation
- Fragmentation
- Payload Header Suppression

- IP Multicast
- CM Authentication
- SNMPv3
- CM Account Management
- Fault Management
- Secure Software

The medium between the CMTS and the different CMs is a two-way shared medium, in which the downstream channels carry signals from the head-end to users and upstream channels carry signals from users to head-end. Upstream and downstream channels are separated using Frequency Division Duplex (FDD). A CM is normally tuned to one upstream channel and the associated downstream channel. The upstream channel is an inherently shared medium while the downstream is a broadcast dedicated link from the CMTS to the CMs.

DOCSIS has a reservation-based, centralized approach for allocating bandwidth and scheduling packet transmissions on the upstream channel. The upstream is modeled as a stream of mini-lots, with a dynamic mix of contention and reservation-based transmission opportunities. CMs may use the contention mini-slots for transmitting their requests, and the CMTS allocates (schedules) transmission opportunities for the CMs in the next frame if capacity is available. Periodically, the CMTS sends a Bandwidth Allocation Map (MAP) message over the downstream channel to indicate to the CMs the specific mini-lots allocated to them. The

Allocation MAP is a MAC Management message that describes some slots as grants for particular stations to transmit data in, other slots as available for contention transmission and other slots as an opportunity for new stations to join the link. As a result of this scheduling, the CMs are guaranteed collision-free data transmission. But collisions may occur during the contention (request) period, and this is resolved using a Contention Resolution Algorithm (CRA). Specifically DOCSIS uses the Truncated Binary Exponential Backoff as its primary CRA. Figure 5.1.3.1 shows the MAP structure.

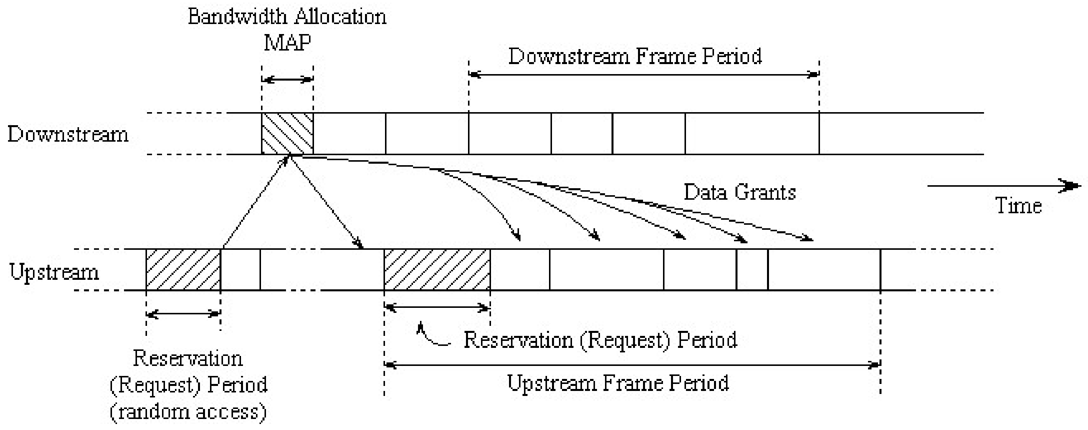


Figure 5.1.3.1 Allocation MAP Structure

Each CM has a unique MAC address. To support QoS, DOCSIS further introduces the notion of Service Flow ID (SID). Each SID is unique and defines a particular service class or flow mapping between a CM and the CMTS. The CMTS may assign one or more SIDs to a particular CM, which is essentially used for bandwidth request and allocation. ***An upstream service flow*** in DOCSIS 1.1 may be one of the following

- Unsolicited Grant Service (UGS)
- Real-time Polling Service (rtPS)
- Non-Real-time Polling Service (nrtPS)
- Best Effort (BE)
- Unsolicited Grant Service with Activity Detection (UGS-AD)

Other key enhancements made in DOCSIS 1.1 are

- Support for multiple service flows per cable modem allows a single modem to support a combination of video, voice and data packets.
- Dynamic service establishment allows MAC messages to dynamically create, modify and delete traffic flows.
- Layer 2 fragmentation allows fragmenting larger data packets
- Concatenation allows CMs to send multiple MAC frames in the same transmission

5.1.2 Upstream Bandwidth Allocation:

There are a number of ways by which the CM can obtain bandwidth for data transmission from the CMTS:

a) By making explicit requests through contention, piggybacking or unicast opportunities
b) Unsolicited grants

Contention:

Portions of the upstream bandwidth are open to all modems (contention) for requesting upstream bandwidth. The requests transmitted through contention are subject to collision, and these collisions are resolved by a ***Contention Resolution Algorithm***.

Piggybacking:

Piggyback is a request for additional bandwidth sent in a data transmission. Piggybacking obviates contention, since the requests are transmitted with the data packets.

Unicast Request Polls:

Periodic unicast request opportunities are sent as a means of real-time polls regardless of network congestion. These opportunities in the Allocation MAP (see below) may be used by the stations to transmit their request packets, avoiding contention. The allocation MAP is a varying length MAC Management message that is transmitted by the CMTS to define the transmission opportunities on the upstream. It includes a fixed-length header followed by variable number of Information Elements (IEs). Each IE defines the allowed usage for a range of minislots. Each IE consists of a 14-bit SID, a 4-bit type code and a 14-bit starting offset. The four defined types of SIDs are *broadcast* (intended for all stations), *multicast* (intended for a group of stations), *unicast* (intended for a particular station) and a *null address* (intended for no station). The various IEs that the CMTS may send in the MAP are:

- Request IE – provides intervals in which requests may be made for bandwidths for upstream data transmission
- Request/Data IE – provides intervals in which requests for bandwidth or short data packets may be transmitted
- Initial Maintenance IE – provides intervals in which new stations may join the network
- Station Maintenance IE - provides intervals in which stations are expected to perform some aspect of routine maintenance
- Short and Long Data Grant IEs - provide an opportunity for a CM to transmit one or more upstream PDUs (Protocol Data Unit which follows a MAC Header in a MAC frame)

Many different scheduling algorithms may be implemented in the CMTS by different vendors. The specification does not mandate a particular algorithm.

5.1.3 Contention Resolution Algorithm (CRA) Overview

The CMTS controls assignments on the upstream channel through the MAP and determines which minislots are subject to collision. The mandatory method of contention resolution that must be supported is based on a ***Truncated Binary Exponential Back-off,*** with the initial back-off window and the maximum back-off window controlled by the CMTS. The values are specified as a part of the MAP and represent a power-of two value. When a CM has information to send and wants to enter the contention resolution process, it sets its internal back-off window equal to the Data Back-off Start defined in the MAP currently in effect. The CM randomly selects a value within its back-off window - $[0,2^{\text{Backoff}}]$. This random value indicates the number of contention opportunities that the CM must defer before transmitting. A CM must only consider the contention transmit opportunities for which this transmission would be eligible. These are defined by either Request IEs or Request/Data IEs in the MAP. After a contention transmission, the CM waits for a Data Grant (Data Grant Pending) or Data Ack in a subsequent MAP. Once either is received, the contention resolution is complete. The CM determines that the contention transmission was lost when it finds a MAP without a Data Grant (Data Grant Pending) or Data Ack for it. The CM must now increase its back-off window by a factor of two, as long as it is less than the maximum back-off window (Back-off End). The CM now randomly selects a number within its new back-off window and repeats the deferring process described above. This re-try process continues until the maximum number of retries (16) has been reached, at which time the PDU must be discarded.

5.3 Quality of Service in DOCSIS 1.1

The DOCSIS 1.1 Specification includes several new Quality of Service (QoS) related concepts not present in DOCSIS 1.0. These include:

- Packet Classification & Flow Identification
- Service Flow QoS Scheduling
- Fragmentation and Concatenation

5.3.1 Theory of Operation

The various DOCSIS protocol mechanisms can be used to provide Quality of Service for both upstream and downstream traffic through the CM and the CMTS. The requirements for Quality of Service include:

- A configuration and registration function for pre-configuring CM based Service
- Flows and traffic parameters
- A signaling function for dynamically establishing QoS- enabled service Flows and traffic parameters
- Utilization of MAC scheduling and traffic parameters for upstream Service Flows
- Utilization of QoS traffic parameters for downstream Service Flows
- Classification of packets arriving from upper layer service interface to a specific active Service flow.

5.3.2 Service Flows

A Service Flow is a MAC-layer transport service that provides unidirectional transport of packets either to upstream packets transmitted by the CM or to downstream packets transmitted by the CMTS. A Service Flow is characterized by a set of QoS Parameters such as latency, jitter and throughput assurances. In order to standardize operation between the CM and CMTS, these attributes include details of how the CM requests upstream minislots and the expected behavior of the CMTS upstream scheduler.

The CM and CMTS provide this QoS by shaping, policing, and prioritizing traffic according to the QoS Parameter Set defined for the Service Flow. The flow in which a packet is transmitted is based on the content of the IP header fields, allowing every application to receive a different service flow. Multiple data flows (each flow corresponding to a service and identified by a service identification descriptor (SID) concurrently exist in a cable modem. A transmission request in the upstream and the corresponding grant includes the SID as the flow identifier. The cable modem and the CMTS negotiate the QoS for each flow upon allocation and dynamically as the service requirement changes. QoS is then achieved by the implementation of sophisticated scheduling mechanisms in the CMTS. A classification function is applied to every packet.

5.3.3 QoS Service Flows in DOCSIS 1.1

Scheduling services are designed to improve the efficiency of the poll/grant access. By specifying a scheduling service and its associated QoS parameters, the CMTS can anticipate the throughput and latency needs of the upstream traffic and provide polls and/or grants at the appropriate times. Each service is tailored to a specific type of data flow described. The basic services comprise:

1. *Unsolicited Grant Service (UGS)* is designed to support real-time service flows that generate fixed size data packets on a periodic basis, such as Voice Over IP (VoIP). The service offers fixed size grants on a real-time basis, which eliminate

the overhead and latency of CM requests and assure that grants will be available to meet the flow's real-time needs. The CMTS must provide fixed size data grants at periodic intervals to the Service Flow. The CM is prohibited from using any contention request and the CMTS should not provide unicast data opportunities. Piggyback requests are also prohibited. The key service parameters are:

- Unsolicited Grant Size (bytes): Used by the CMTS to compute the size of the unsolicited grant in minislot units.
- Grants Per Interval: The actual number of data grants per Nominal Grant Interval
- Nominal Grant Interval: Specifies the nominal interval between successive data grant opportunities for this Service Flow.
- Tolerated Grant Jitter: Specifies the maximum amount of time that the transmission opportunities may be delayed from the nominal periodic schedule for a particular Service Flow

2. *Real-Time Polling Service (rtPS)* is designed to support real-time service flows that generate variable size data packets on a periodic basis, such as MPEG video. The service offers real-time, periodic, unicast request opportunities, which meet the flow's real-time needs and allow the CM to specify the size of the desired grant. This service requires more request overhead than UGS, but supports variable grant sizes for optimum data transport efficiency. The CMTS provides periodic unicast request opportunities. When the source becomes inactive, the

transmission reservations are released to other flows. The CM is prohibited from using any contention request or request/data opportunities, and also from sending piggybacked requests. The CM uses only the unicast request opportunities in order to obtain upstream transmission opportunities. Key service parameters:

- Nominal Polling Interval: Specifies the nominal interval between successive unicast request opportunities for this Service Flow on the upstream channel
- Minimum Reserved Traffic Rate (bits/s): The CMTS should be able to satisfy bandwidth requests for a service flow up to its Minimum Reserved Traffic Rate.

3. *Non-Real-Time Polling Service (nrtPS)* is designed to support non-real-time flows that require variable size data grants on a regular basis, such as high bandwidth FTP. The service offers unicast polls on a regular basis, which assures that the flow receives request opportunities even during network congestion. The CMTS typically polls nrtPS SIDs on an (periodic or non-periodic) interval on the order of one second or less. The CMTS must provide timely unicast request opportunities. The CM is allowed to use the contention request opportunities as well as the unicast request opportunities. Key service parameters:

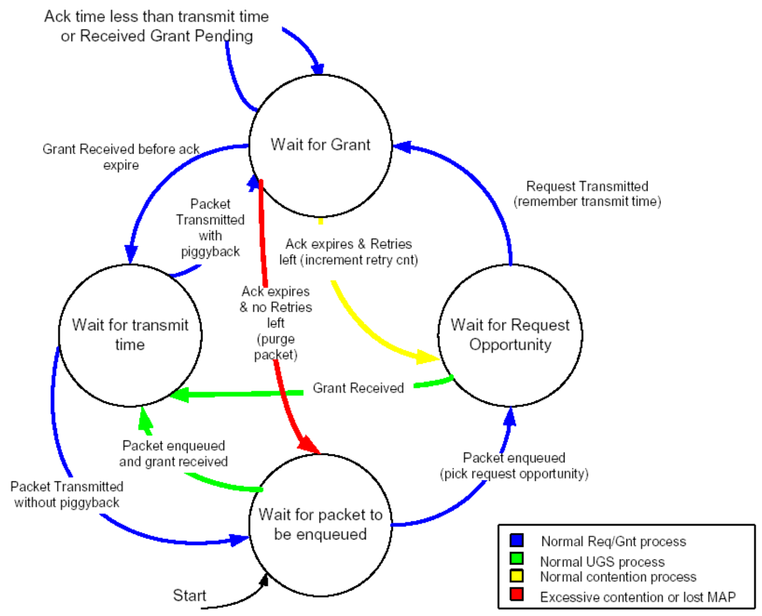
   - Nominal Polling Interval: Specifies the nominal interval between successive unicast request opportunities for this service flow on the upstream channel.
   - Traffic Priority: The value of this parameter specifies the priority assigned to a service flow (Valid range: 0-7)

4. *Best Effort (BE)* service provides efficient service to best-effort traffic. The CM is allowed to use contention as well as piggybacking of requests. This service flow has limited QoS support. The key service parameter is:
   - Traffic Priority: The value of this parameter specifies the priority assigned to a service flow (Valid range: 0-7)
5. *Unsolicited Grant Service with Activity Detection (UGS/AD)* is designed to support UGS flows that become inactive for substantial portions of time, such as VoIP with silence suppression. The service provides unsolicited grants when the flow is active and unicast polls when the flow is inactive. This combines the low overhead and low latency of UGS with the efficiency of rtPS. Though UGS/AD combines UGS and rtPS, only one scheduling service is active at a time. The CMTS must provide periodic unsolicited grants when the flow is active, but reverts to providing unicast request opportunities when the flow is inactive. It detects inactivity by unused grants. The CM is prohibited from using contention requests or request/data opportunities. Piggybacking requests are also prohibited. The key service parameters are Nominal Polling Interval, Nominal Grant Interval, and Unsolicited Grant Size.

5.3.4 Fragmentation & Concatenation

Fragmentation is sending a portion of a packet frame during a reserved slot time. Fragmentation is an upstream CM “modem capability”. The CMTS must enable or disable this capability on a per-modem basis. The per-modem basis provides compatibility with DOCSIS 1.0 CMs. Once fragmentation is enabled for a DOCSIS 1.1 modem, fragmentation is enabled on a per Service Flow basis. When enabled for a Service Flow, fragmentation is initiated by the CMTS when it grants bandwidth to a particular CM with a grant size that is smaller than the corresponding bandwidth request from the CM. This is known as a **Partial Grant.**

Concatenation refers to sending more than a frame during a transmission opportunity i.e. it allows the cable modem to make a single time slice request for multiple packets and send all packets in a single large burst on the upstream. Fragmentation and concatenation also make better use of the scarce upstream resource and improve throughput. There was no fragmentation in DOCSIS 1.0 and concatenation was optional. Figure 5.3.4.1 is a state diagram summarizing the request/grant process.



Ack time less than transmit time or Received Grant Pending
Wait for Grant
Grant Received before ack expire
Request Transmitted (remember transmit time)
Packet Transmitted with piggyback
Ack expires & Retries left (increment retry cnt)
Ack expires & no Retries left (purge packet)
Wait for transmit time
Wait for Request Opportunity
Grant Received
Packet enqueued and grant received
Packet enqueued (pick request opportunity)
Packet Transmitted without piggyback
Wait for packet to be enqueued
Start
Normal Req/Gnt process
Normal UGS process
Normal contention process
Excessive contention or lost MAP

Figure 5.3.4.1 State Diagram of Request/Grant Process

5.4 OPNET’s DOCSIS Models

OPNET's Model Library is recognized industry-wide as the most advanced suite of models of network protocols, technologies, and applications available. OPNET's DOCSIS model suite has been developed jointly by OPNET and the CableLabs' Bandwidth Modeling and Management Vendor Forum. The model is based on the DOCSIS 1.1 specification (as established by CableLabs), and includes features relevant to both DOCSIS 1.1 and 1.0 system. The model enables testing configurations and architectures before building expensive prototypes. It also allows creation of virtual representations of proposed cable modem networks so as to evaluate the capacity and quality of service (QoS) characteristics of alternative designs. A detailed description on OPNET’s DOCSIS model suite can be found in [21].

5.4.1 Bugs

In the course of performing the different simulations a few bugs were found out in OPNET’s DOCSIS 1.1 implementation. The first three have been fixed in OPNET Model Release 9. The last one is yet to be fixed.

- DOCSIS RTP process does not add unicast requests to the MAP in the required intervals .The docsis_cmts process calls the RTP process to find out whether or not a unicast request should be added to the MAP every time it creates a new MAP. The RTP process should be basically checking whether the time since the last poll is greater than the real-time polling interval or not.

If yes, a unicast request must be added to the MAP. There seems to be a logical error due to which this is not happening. As a result performance studies were not done with the rtPS QoS class.

- CMTS does not add Pending Grant IEs to the MAP. When the CMTS identifies that it can't satisfy a grant, it has to add a Pending IE to the MAP to inform the CM that it’s request has been received and that it would be satisfied in the subsequent MAP. This does not happen, due to a bug in the implementation. This results in the CM contending again which it wouldn't have done had the pending grant IE been added to the MAP. This would have a small but definite impact on the results obtained. The fact that the CMs contend again would have led to an increase in the number of contentions and hence the collision probability. But this is true only for the experiments for which concatenation and fragmentation had been disabled, because if fragmentation had been enabled, the CMTS does not deny a grant, but instead gives a partial grant. This bug may have had some effect on the simulation results.
- DOCSIS nrtPS does not add unicast requests to the MAP. DOCSIS 1.1 offers unicast request opportunities to nrtPS to avoid contention during times of network contention. But due to a logical bug in the implementation, the CMTS was not adding the unicast request polls to the MAP. This is the only feature that differentiates nrtPS from BE. Again because of this we were not able conduct performance studies with nrtPS.

- Pending grant is not issued if a DOCSIS RTP request cannot be granted. RTP stations go into collision resolution state. Since RTP is not allowed to contend, it does not get out of the collision resolution state. Hence the RTP stations stop sending traffic from that time onward.

5.5 Simulation Scenarios

The figure 5.5.1 shows an OPNET simulation scenario, with 2 CMs and the CMTS. Also the various configuration objects for configuring DOCSIS parameters and traffic parameters can be seen.

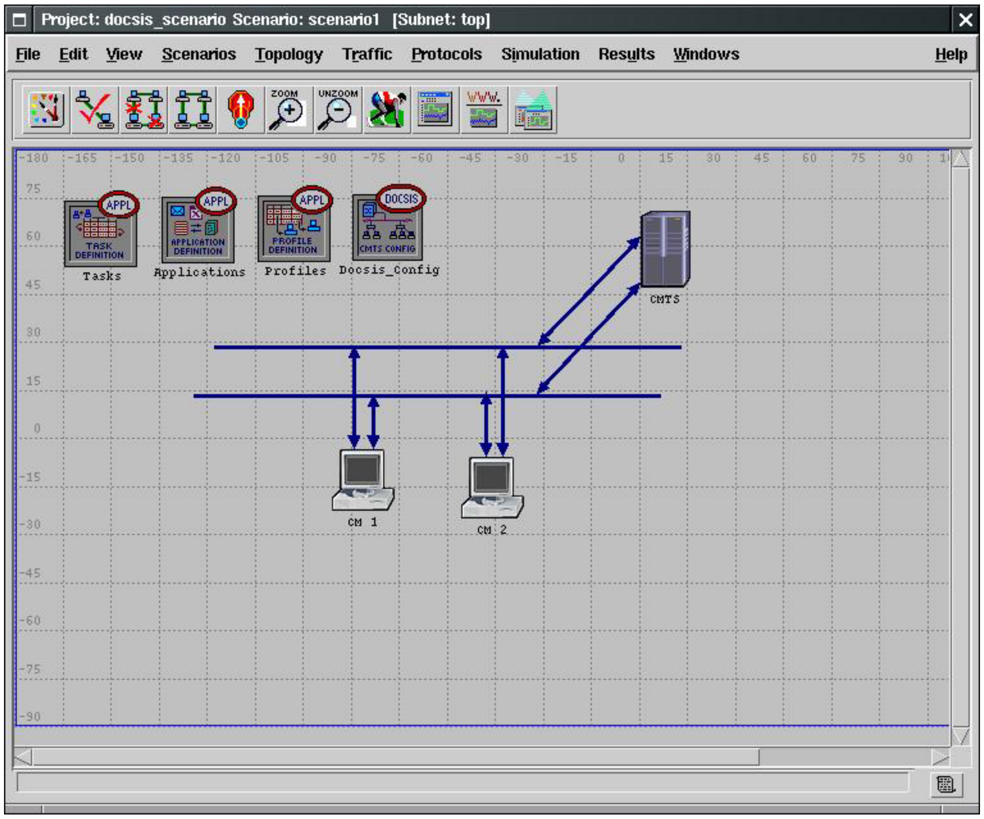


Figure 5.5.1 Simulation Scenario in OPNET

5.6 Performance Evaluation of DOCSIS 1.1 QoS

5.6.1 DOCSIS CMTS Parameters

- Physical Media Profile:

  This defines profiles that allow the user to configure the RF specifications (modulation, channel width, data rate, and center frequency) and interleave latency of downstream channels and upstream channels. Parameters used for these simulations were:

  Upstream Bandwidth: 640 Kb/s

  Downstream bandwidth: 27Mb/s

  Upstream modulation scheme: QPSK

  Downstream modulation scheme: 64 QAM

- MAP Profile:

  This attribute defines the bandwidth allocation MAP profiles that are applied to the upstream channel parameters on CMTS nodes.

  MAP inter-arrival time: 0.2s

  Short data grant limit: 128 bytes

  Long data grant limit: 10000 bytes

  Data back off start: 7

  Data back off end: 16

  Number of contention slots: 64

  Bytes per minislot: 4 bytes

5.6.2 Experiment 1

a. Comparison of Best Effort & UGS Delays
b. Effect of Piggybacking on Best Effort Delay

Configuring the UGS station:

The total upstream bandwidth is 640Kbps. 64 minislots have been allotted for contention. The remaining bandwidth can be calculated in the following way:

Duration of one minislot = 50 us

Number of minislots in MAP: MAP time / Duration of one minislot

= 0.2/0.00005

= 4000 minislots

Number of minislots available for data transmission	= 4000- (32+2) [2 slots for station maintenance]
Size of one minislot	= 4 bytes
Remaining bandwidth	= (3966 * 4 * 8) / 0.2
	= 634.56 Kbps

Of the remaining bandwidth ~ 10% has been allotted to UGS. Since UGS is generally used for Constant Bit Rate Traffic, i.e. for flows that generate fixed packets in regular intervals, the station has been configured in the following manner.

<u>UGS Station Traffic Statistics</u>:

- Request Packet size = 1000 bytes
- Packet Inter-arrival time = 0.21s
- Packet size distribution: Constant
- Inter-arrival times distribution: Constant

UGS QoS Parameters:

The Nominal Grant interval is set so that there is a grant every MAP and it is also approximately equal to the packet inter-arrival time.

- Grant Size = 1300 bytes
- Nominal Grant Interval = 0.2s

Physical overhead calculation:

The following parameters are used for the physical overhead calculation. These are configured as a part of the Physical Media Overhead within the MAP Profiles. Messages can be short or long data frames. Short data frames have a 100-byte limit while long frames can go up to 10,000 bytes. Since all the packets lengths used in the simulation exceeds 100 bytes, the long data frame physical parameters are listed below. Before transmitting a packet received from the higher layer, the CMTS divides the packet into code words. It then adds FEC error correction bytes to each code word and preamble length bits to each frame.

Preamble Length: 56 bits

FEC Error Correction (FEC Parity): 16 bytes

FEC Code Word Length: 226 bytes

Guard Time (bits): 40

Last Code Word Mode: Fixed

A sample overhead calculation for a packet size of 1300 bytes is shown below.

The MAC headers are added to the IP frame (includes 20 bytes of IP header and 20 bytes of TCP header).

Total message size before upstream physical overhead is added = Size of IP frame + DOCSIS_MSG_PDU_HDR_SIZE (20 bytes)

Message size = Payload size + TCP/IP headers + MAC Overhead

= 1300 + 40 + 20

= 1360 bytes

No of code words = (message size)/(FEC Code word length- FEC Parity)

= 1360 /(226-16)

= 6.47

= 7 (rounded off to the highest integer)

Now the final message size = number of code words x FEC code word

= 7 x 226

= 1582

Total Frame Size = Message size + (Preamble + Guard Time)/8

= 1582 + (56 + 40)/8

= 1582 + 12

= 1594 bytes.

The total bandwidth allocated to the UGS station is (1594 * 8)/0.2 =63.76 Kbps.

Configuring the BE stations:

The rest of the bandwidth was distributed among the BE stations. The stations had identical parameters and also traffic statistics. The load was increased by adding stations. Fragmentation and concatenation were disabled.

BE Stations Traffic Statistics:

- Mean packet Size = 800 bytes
- Mean packet inter-arrival time = 0.22s
- Packet size distribution: Exponential
- Inter-arrival time distribution: Exponential

Length of Simulation (min): 37

Approximate number of packets generated by each station: 10,000

Since Best Effort flows are similar to Internet traffic, exponential arrivals and exponential packet sizes were set. The inter-arrival time must be greater than the MAP inter-arrival time for the queue size to remain bounded. This is because fragmentation and concatenation were disabled (to make the scenario simple). Hence only one packet at a time can be sent in an upstream transmission interval. Hence the mean inter-arrival time should be set so that there is at most one arrival during the MAP inter-arrival time (on average).

Results:

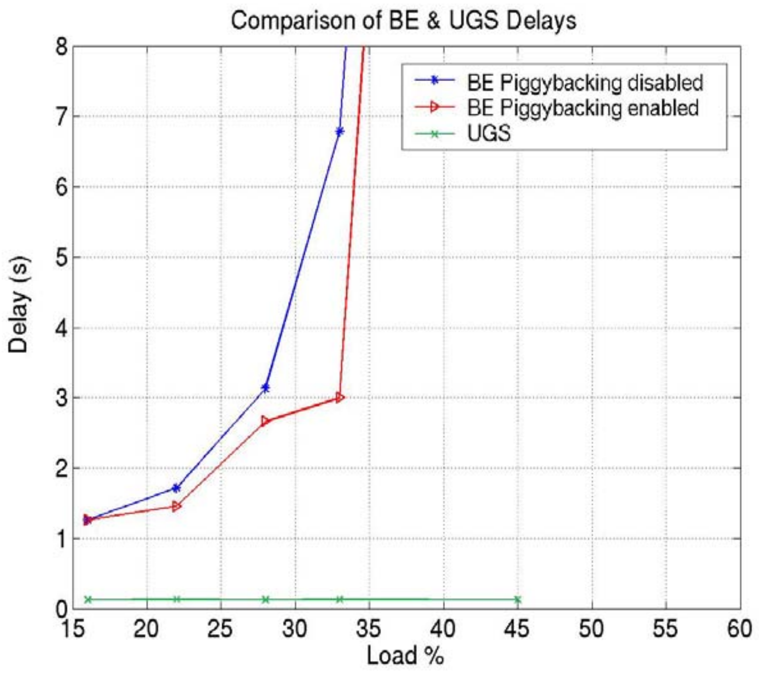


Figure 5.6.2.1 Comparison of UGS and BE Delays

Analysis:

The performance of BE traffic was compared with UGS. UGS flows are allowed to reserve a certain portion of the bandwidth. No requests for transmission are needed; hence UGS has a low bounded constant delay, since it receives a grant every MAP and thus queue size is low and bounded. BE on the other hand has "request grant, request grant" pattern. Stations have to contend for sending requests and have to wait for grants in the subsequent MAPs. Contentions may result in collision and thus there may be increased delay due to retransmissions. The effect of piggybacking on delay is also studied. With piggybacking enabled, requests are piggybacked to outgoing data and thus the frequency of contention is reduced and delay is reduced. We see that the system saturates quickly. This is because fragmentation and concatenation were disabled; hence only one frame can be sent per MAP. This leads to relatively high delay values, since the queue size builds up quickly. This is because as load increases, the CMTS would not be able to satisfy all the requests from all the stations. Also in this experiment and in all the experiments that follow, the load value is the actual load value and not the calculated value. The load value was calculated from the simulations as

Reserved slot time in MAP/ Total MAP duration. We report actual loads because with the exponential packet lengths and exponential inter-arrival times, it is very difficult to predict the load values from the mean packet size and inter-arrival time. The packet lengths are exponentially distributed and hence they would be distributed in a wide range of values. Also, as explained before, there is also some amount of complexity involved in the computation of the upstream physical overhead.

5.6.3 Experiment 2

Effect of Fragmentation & Concatenation

Configuring the UGS station:

As before ~ 10% of the total bandwidth was allotted to the UGS Station. The MAP configuration parameters were retained for all the experiments.

UGS Station Traffic Statistics:

- Request Packet size = 1000 bytes
- Packet Inter-arrival time = 0.21s
- Packet size distribution: Constant
- Inter-arrival times distribution: Constant
- Number of packets generated by each station: 10,000

UGS QoS Parameters:

- Grant Size = 1300 bytes
- Nominal Grant Interval = 0.2s

The total bandwidth allocated to the UGS station is (1594 * 8)/0.2 =63.76 Kbps.

Configuring the BE stations:

The rest of the bandwidth was distributed among the 11 BE stations. The stations had identical parameters and also traffic statistics. The load was increased by adding stations to the scenario. The experiment was performed to study the effect of fragmentation and concatenation on performance i.e., on mean delay. Hence one set of simulations was carried out with fragmentation and concatenation disabled, and another with enabled. Piggybacking was enabled and data back-off start was maintained at 7.

<u>BE Stations Traffic Statistics</u>:

- Mean packet Size = 800 bytes
- Packet size distribution: Exponential
- Inter-arrival time distribution: Exponential
- Mean packet inter-arrival time: variable
- Number of packets generated by each station: 10,000

Results:

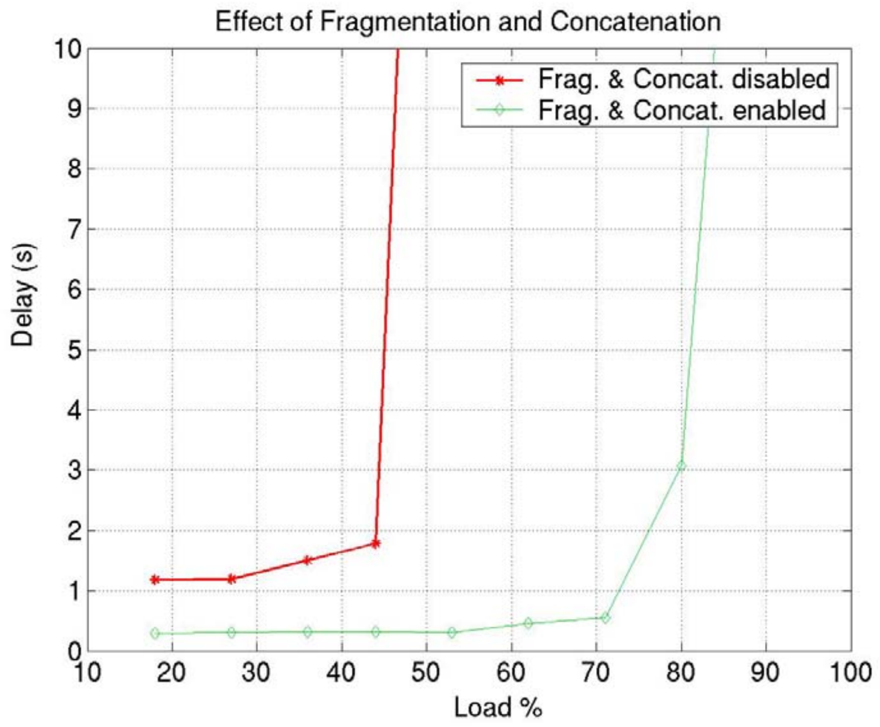


Figure 5.6.3.1 Effect of Fragmentation & Concatenation

Analysis:

DOCSIS concatenation combines multiple upstream packets into one packet to reduce packet overhead (as explained already, the physical upstream overhead is considerable) and overall latency, and to increase transmission efficiency. Using concatenation, a DOCSIS cable modem makes only one bandwidth request for multiple packets, as opposed to making a different bandwidth request for each individual packet; this technique is especially effective for bursty real-time traffic, such as voice calls. Since a single bandwidth request is made for a multiple frames, the queuing time is reduced, and this brings down the overall access delay. This also saves upstream bandwidth to an extent since sending of many small request packets is avoided. The need for frequent contention is also reduced, resulting in fewer collisions. This also contributes to an improvement in performance. When enabling concatenation, fragmentation should also be enabled; otherwise packets that are too large for transmission in a single MAP are not transmitted. That is, when concatenation is enabled the request sizes are generally big, and the CMTS does not issue a grant at all if fragmentation is disabled and there are not enough minislots in the current MAP. Thus fragmentation and concatenation together help in better utilization of the upstream bandwidth compared to the case when both were disabled. In that case we are only able to reach loads of 45%, since after that the queue builds up and delay increases indefinitely. But when fragmentation and concatenation were enabled, we are able to reach much higher load values with lesser access delays.

5.6.4 Experiment 3

Effect of Backoff Start and Piggybacking

Configuring the UGS station

As before ~ 10% of the total bandwidth was allotted to the UGS Station. The MAP configuration parameters were retained for all the experiments.

UGS Station Traffic Statistics:

- Request Packet size = 1000 bytes
- Packet Inter-arrival time = 0.21s
- Packet size distribution: Constant
- Inter-arrival times distribution: Constant
- Number of packets generated by each station: 10,000

UGS QoS Parameters:

- Grant Size = 1300 bytes
- Nominal Grant Interval = 0.2s
- The total bandwidth allocated to the UGS station is (1594 * 8)/0.2 =63.76 Kbps.

Configuring the BE stations:

The rest of the bandwidth was distributed among the BE stations. The stations had identical parameters and also traffic statistics. In this experiment the number of BE stations were maintained constant at 11 and the load was changed by changing the inter-arrival times. In order to make load values predictable and to simplify protocol operation, fragmentation and concatenation were disabled and packet lengths were made constant. This was done since with constant packet lengths, we would know for sure the exact load values since there would be no randomness involved in the packet lengths. Also disabling fragmentation and concatenation helps study the effect back-off values better. The experiment was done first disabling piggybacking, and then enabling it. Back-off end value was maintained at 16.

<u>BE Stations Traffic Statistics</u>:

- Mean packet Size = 800 bytes
- Packet size distribution: Constant
- Inter-arrival time: variable
- Inter-arrival time distribution: Exponential
- Number of packets generated by each station: 10,000

Case a: Piggybacking disabled
Results:

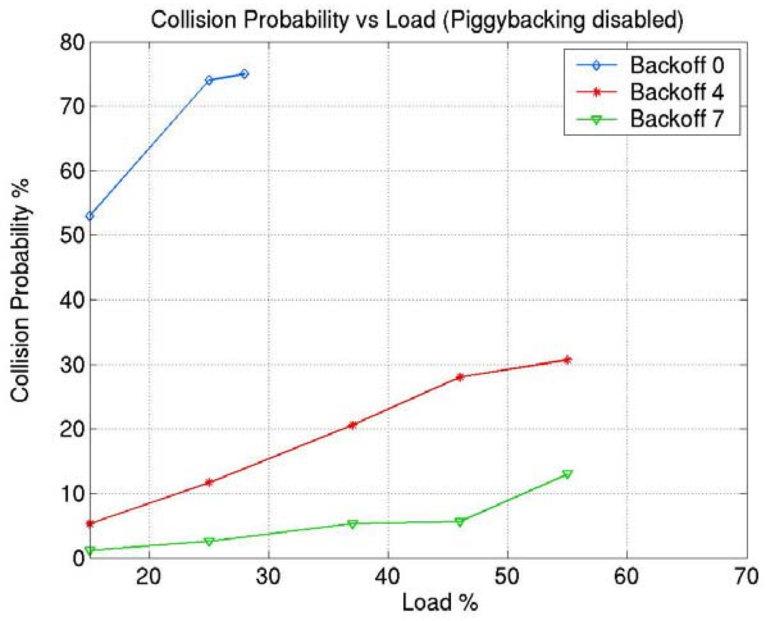


Figure 5.6.4.1 Effect of Back-off on Collision Probability (Piggybacking disabled)

Analysis:

We see the effect of the data back-off start value on collision probability. For a data back-off of zero we see that the collision probability is very high and it increases with increase in load. This is because since the initial value is zero, the back-off window is zero (no random wait). Hence the probability that two stations would pick the same minislot for contention is very high. This would result in a collision and the contention requests of both the stations would be lost. Also since piggybacking has been disabled, there are more frequent contentions as load increases. This leads to increased probability of collisions. We see that the collision probability decreases as the back-off value increases. This is because the station defers its transmission by a greater number of minislots (since its window size is big). Also we see that we are able to go up to much higher loads for back-off 4 & 7. This is because with a lower back-off value, there would be increased number of retransmissions since the collision probability is more. Hence the queue builds up quickly. But with greater back-off values, the collision probability is comparatively less and hence we are able to go up to comparatively high load values.

Case b: Piggybacking enabled

Result:

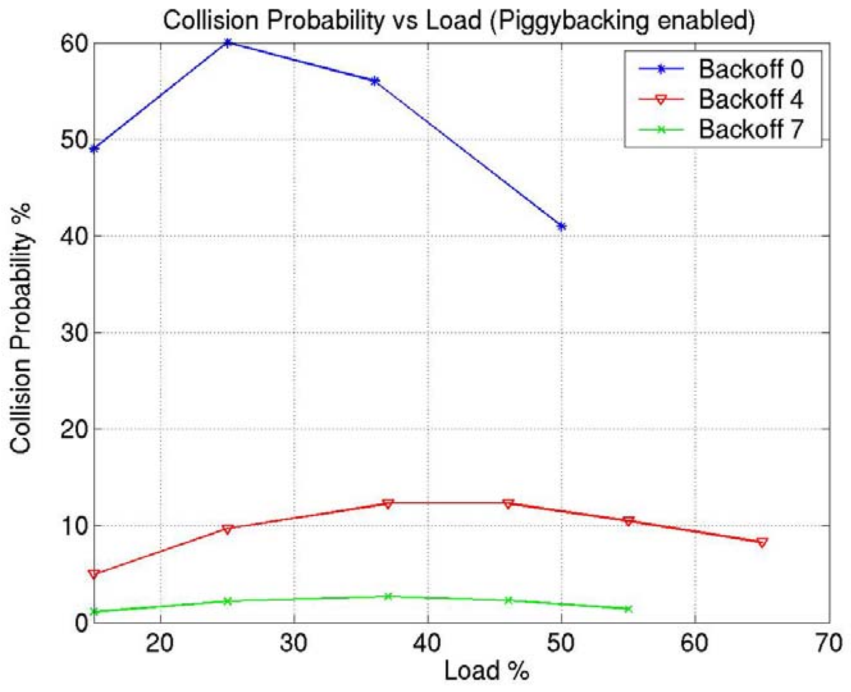


Figure 5.6.4.2 Effect of Back-off on Collision Probability (Piggybacking enabled)

Analysis:

The same experiment was repeated with piggybacking enabled. Here the collision probability increases with load for lower load values, and then decreases. This is because with higher load values, the contention load decreases, since most of the requests can by piggybacked with the data transmissions. But at lower loads, the probability that there may be a packet in the queue while data is being transmitted is less. Hence the station might be forced to use contention more often, even if piggybacking is enabled. Hence the collision probability increases. But at higher loads, the piggybacking feature may be used more effectively and hence the number of contentions and hence the collisions may be reduced. Also it can be observed that we are able to reach higher loads compared to the case when piggybacking was disabled.

Effect of Back-off on Delay:

Results:

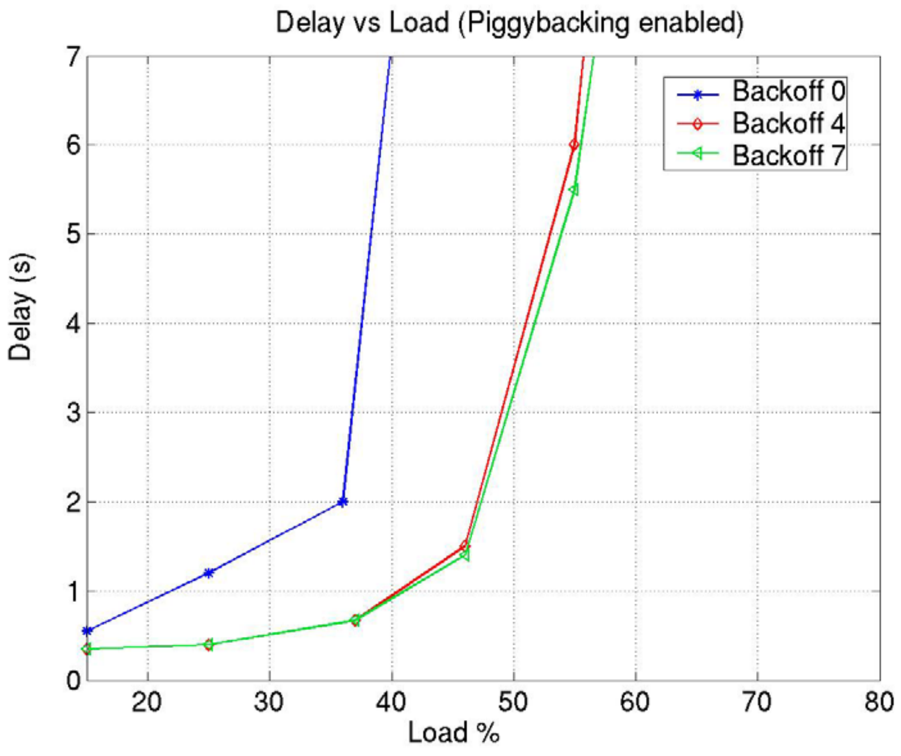


Figure 5.6.4.3 Effect of Back-off on Delay (Piggybacking enabled)

Analysis:

We see that increasing the data back-off value above 0 produces a marked improvement in access delays. This is because if the back-off value is high, the collision probability reduces and hence delay decreases. We also can observe that there is not much difference in delays for back-off 4 and 7. With back-off of 7 though the collision probability is much lesser than what it is for 4, the corresponding decrease in delay is offset by the increase due to increased delay in contention. This is because the stations have to back off by a larger number of minislots, and this increases the average waiting time to send the request packets.

5.6.5 Experiment 4

Effect of Back-off Start and Piggybacking – A more realistic scenario

Configuring the UGS station:

As before ~ 10% of the total bandwidth was allotted to the UGS Station. The MAP configuration parameters were retained for all the experiments. The rest of the attributes were same as that of the previous experiments.

Configuring the BE stations:

The rest of the bandwidth was distributed among the BE stations. The stations had identical parameters and also traffic statistics. In this experiment the number of BE stations was maintained constant at 11 and the load was changed by changing the inter-arrival times. Here fragmentation and concatenation were enabled and packet lengths were made exponentially distributed. Hence it was not possible to predict the load values ahead and they were measured from the simulation. The experiment was done first disabling piggybacking, and then enabling it.

<u>BE Stations Traffic Statistics</u>:

- Mean packet Size = 800 bytes
- Packet size distribution: Exponential
- Inter-arrival time: variable
- Inter-arrival time distribution: Exponential

- Number of packets generated by each station: 10,000

Results:

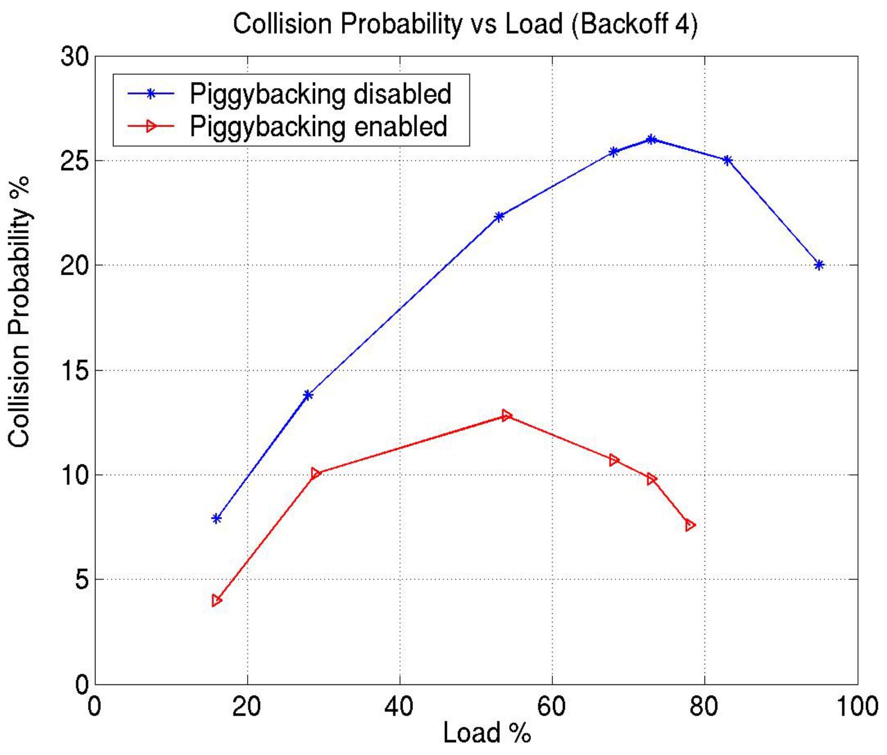


Figure 5.6.5.1 Effect of Piggybacking on Collision Probability

Analysis:

The simulations were carried out to study the effect of back-off on collision probability, for back-offs 0, 4 & 7. The results are plotted in figure 5.6.5.1 showing the effect of piggybacking for back-off 4. We see that enabling piggybacking has reduced the collision probability since the need for contention is reduced with piggybacking enabled. But we see that the trend is same whether piggybacking has been enabled or not. This is because concatenation has been enabled. So the effect of concatenation is more prominent. Since with concatenation enabled, requests for multiple frames can be made, there is less need for contention. Hence the collision probability first increases with load, for lower load values. This is because there are a large number of intervals during which there are no packet arrivals, and hence the stations are forced to use contention; with piggybacking the need for contention is a little less. But as load increases, the buffer is almost never empty and hence the contention load keeps decreasing and hence the collision probability decreases.

Effect on delay:

For the same scenario, the effect of back-off on delay is discussed for piggybacking enabled case. The behavior is the same for the piggybacking disabled case, except that the delay is slightly more.

Results:

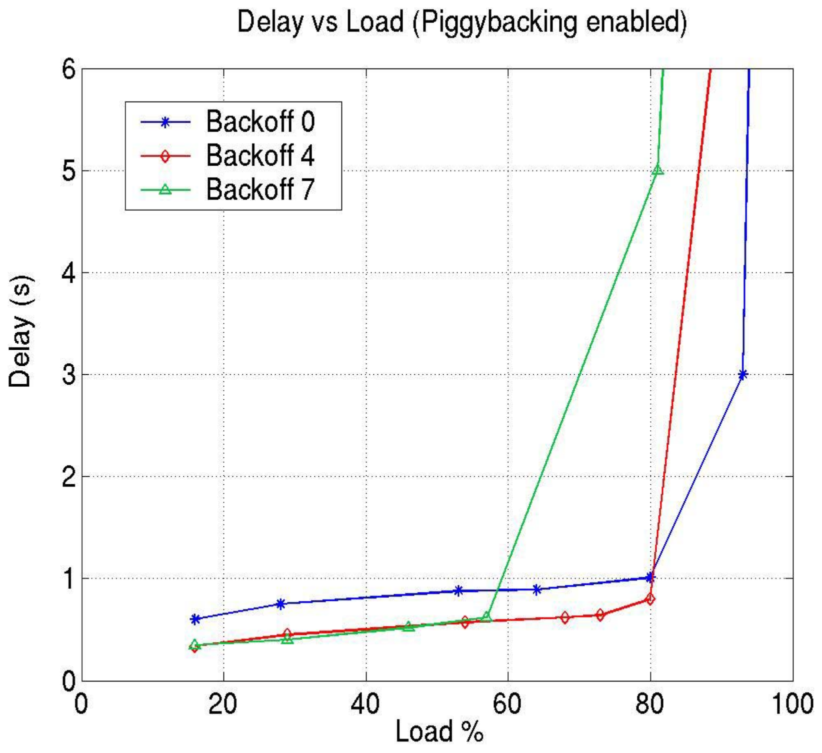


Figure 5.6.5.2 Effect of Back-off on Delay (Piggybacking enabled)

Analysis:

For low loads, increasing the back-off value reduces the collision probability and hence delay decreases significantly due to reduced retransmissions. There is not much of a difference between back-off 7 and 4 for lower loads, since the decrease in delay as a consequence of decreased collision is offset by the increased delay incurred during contention. This is because as the back-off value is more, the back-off window is larger and hence the station defers by a greater number of minislots before transmitting its request. This increases delay since for a specified value of priority, the CMTS issues grants in the order in which they were received. But it is observed that the load value at which the system saturates, i.e. the buffer builds up, decreases as the back-off value increases. This is because each station defers its transmission by a greater number of minislots, and hence it is possible that the request may not reach the CMTS before the next MAP is transmitted. Hence it may so happen that the station is forced to wait for two MAP periods. This leads to the queue building up and delay rising indefinitely. But for back-off zero, the window starts with a small value and slowly increases each time a collision occurs and hence it is more adaptive to the load, and helps utilizing the entire bandwidth.

Comparing the results illustrated in figure 5.6.5.2 with those in figure 5.6.4.3, we see that the system saturates quickly for higher back-off values in figure 5.6.5.2, as opposed to figure 5.6.4.3. This is because for the experiment of figure 5.6.5.2, fragmentation and concatenation were enabled, whereas for experiment 3 (figure 5.6.4.3) both were disabled. So with fragmentation and concatenation enabled, it is possible to achieve a much higher throughput and utilize the entire bandwidth available in one MAP period. Also if the back off is high, with the piggybacking enabled, it is possible that certain requests reach the CMTS only after the subsequent MAP is sent. This leads to an increased waiting time, and also building up of the queue. On the other hand, with the previous experiment (figure 5.6.4.3), since fragmentation and concatenation were disabled, the collisions had a prominent effect and it wouldn't be possible to saturate the entire available bandwidth. So the system saturates quickly for the lower back-off value.

5.6.6 Experiment 5

Effect of Priorities on Delay

Simulation Scenario:

The set up consisted of 2 UGS stations and 8 Best Effort stations. The packet attributes and other UGS parameters were the same as that of previous experiments. The total bandwidth allocated to the UGS station is (1594 * 8 *2)/0.2 = 127.72 Kbps. There were 8 Best Effort stations with priorities [0-3], i.e. 2 stations per priority. The number of stations was maintained constant throughout the experiment. Load was changed by changing the packet inter-arrival times. Since fragmentation and concatenation were enabled, there was no constraint on the inter-arrival times (i.e. they are no longer required to be greater than the MAP duration.). Again load values were measured from the simulation.

<u>BE Stations Traffic Statistics</u>:

- Mean packet Size = 800 bytes

- Mean packet inter-arrival time = variable
- Packet size distribution: Exponential
- Inter-arrival time distribution: Exponential
- Number of packets generated by each station: 10,000

Results:

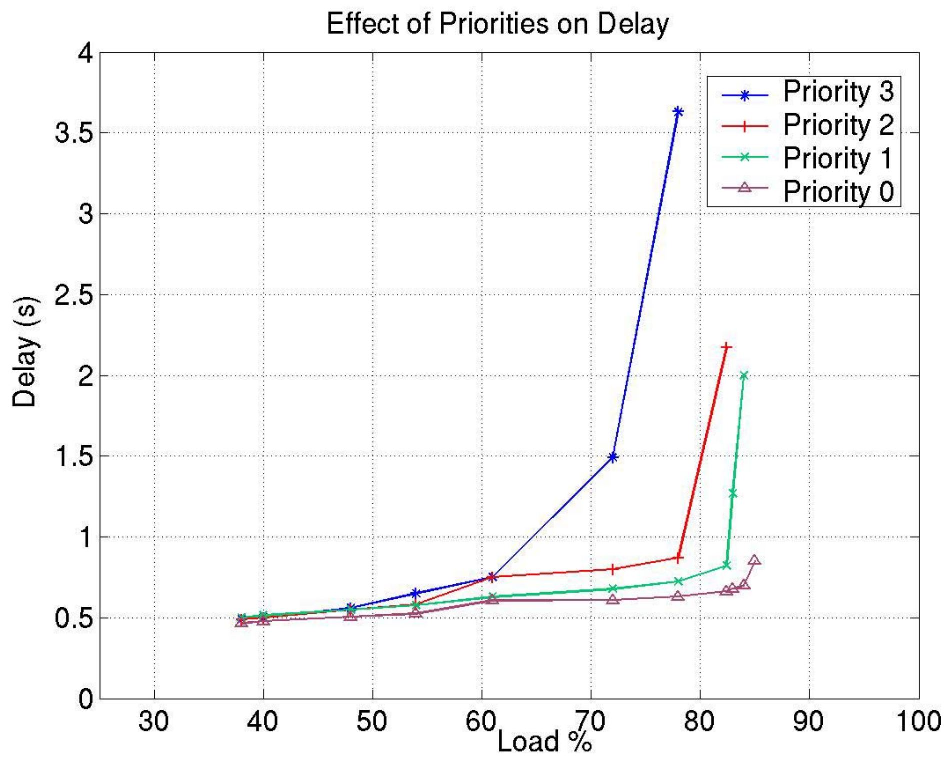


Figure 5.6.6.1 Effect of Priorities on Delay

Analysis:

The effect of Traffic Priorities on delay is illustrated in figure 5.6.6.1. The standard recommends a Traffic Priority parameter for the BE and the nrtPS flows. There is no priority for the UGS and the rtPS effort stations, since UGS flows are guaranteed bandwidth, and the rtPS flows get unicast request opportunities. Thus these flows are implicitly assigned a high priority. Only BE flows were set up in the simulation experiment for studying the effect of priorities. The CMTS uses the Traffic Priority attribute for determining precedence in grant generation. Priorities do not affect contention process, though. Traffic Priorities range from 0-7 applicable to BE and nrtPS, 0 being the highest. It is clearly seen that the higher priority stations have lower access delay, since it is given preference in transmitting data. The grants in the MAP are scheduled in order of priority, and thus a high priority station gets to transmit its packets first and thus suffers lower access delays.

6 Conclusions and Future Work

We finish up this thesis with a quick summary of the contributions of this research effort and a few pointers to future work

6.1 Summary of Contributions

- ***Characterized the performance of HF^{2}Q:*** The performance of a new queuing mechanism was characterized completely. The properties of HF^{2}Q were studied from simulations under different load conditions. The performance was evaluated by studying the delay and throughput characteristics of the queuing algorithm.
- ***Compared the performance of HF^{2}Q to WFQ:*** The performance of HF^{2}Q was compared to that of WFQ since HF^{2}Q results from a modification made to WFQ. The comparisons were made under different scenarios (different combinations of reservations and incoming load for the different flows). This helps us better understand the behavior of both the algorithms, which are extremely complicated.
- ***Evaluated the performance of QoS features of DOCSIS 1.1:*** This is very significant because a complete performance evaluation of this particular standard (standardized in 2001) has never been done before. DOCSIS is a very complex MAC protocol with a number of new QoS features introduced. Hence understanding of the operation of the protocol and evaluating the performance of the QoS features for different classes of traffic is very important for any

future work. This thesis has evaluated the QoS features for 2 classes of traffic namely UGS and Best Effort.

6.2 Summary of Conclusions

- It was determined from simulations that HF^2Q behaved as FIFO when the total system load is less than unity and behaves as WFQ when the total system load exceeds unity.
- The throughput behavior of HF^2Q was studied and comparisons were made with that of WFQ and FIFO. Since all three are work-conserving, when the total load is less than 1, they have identical throughput characteristics. When the system is overloaded, WFQ and HF^2Q behave identically and offer protection to well-behaving flows, but FIFO essentially offers throughput according to demand, as expected.
- As regards the performance evaluation of DOCSIS 1.1, it was established from simulations that the UGS traffic (Constant Bit Rate Traffic) has bounded and low delays irrespective of the load offered by the other classes. It was also determined that fragmentation and concatenation improves the delay characteristics of best effort traffic when the load is very high. Also piggybacking improves delay for best effort since it reduces the probability of collision.

6.3 Future Work

- There is much scope for future work with regard to the performance of HF^{2}Q. Understanding how the queuing algorithm behaves is very important because this could be a significant step in presenting an analytic explanation as to why HF^{2}Q behaves as FIFO in an underloaded case. This is again another key step to a possible future work as - finding a closed form expression for the main waiting time for WFQ.
- DOCSIS 1.1 mentions two other traffic classes namely RTPS and NRTPS. The evaluation of the performance of these traffic classes was not possible due the fact that there were bugs in the software implementation. Hence evaluating that would definitely be an interesting and worthwhile research. Also it would be interesting to compare the performances of existing scheduling architectures for DOCSIS 1.1 to the new scheduling architecture proposed in [19].

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Appendix 1

This Appendix presents results for experiments that are similar to experiment 1 described in chapter 4. Case 1 presents the underloaded results and case 2 presents the overloaded results. Case 2a gives the results for all load values while case 2b presents results that do not show the transitional behavior of HF^{2}Q

Experiment a:

The tagged flow load is varied from 0.05 Mbps to 0.9 Mbps. The non-tagged flows have identical reservations and incoming loads. The experiment was repeated for 10 different loads of the tagged flow.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1(tagged)	0.8	0.05-0.89
2	0.1	0.05
3	0.1	0.05

Table A1.1 Flows setup: Exp. a

Case 1:

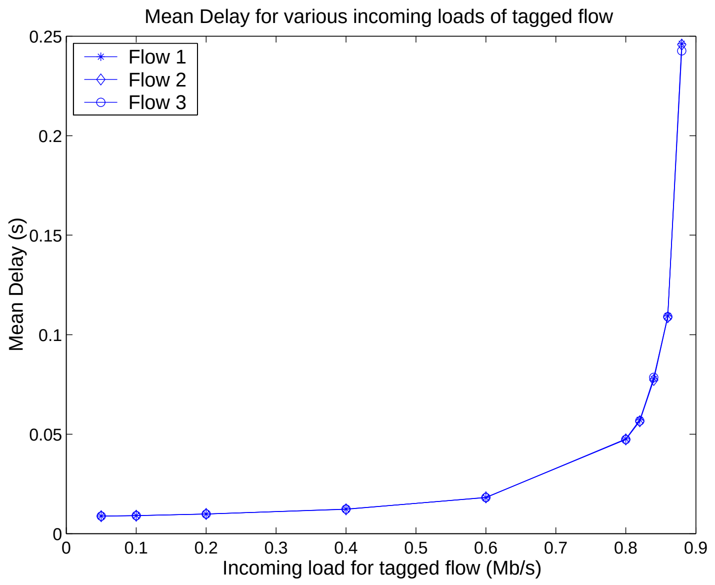

Figure A1.1 Mean Delay for HF^{2}Q: Exp. a

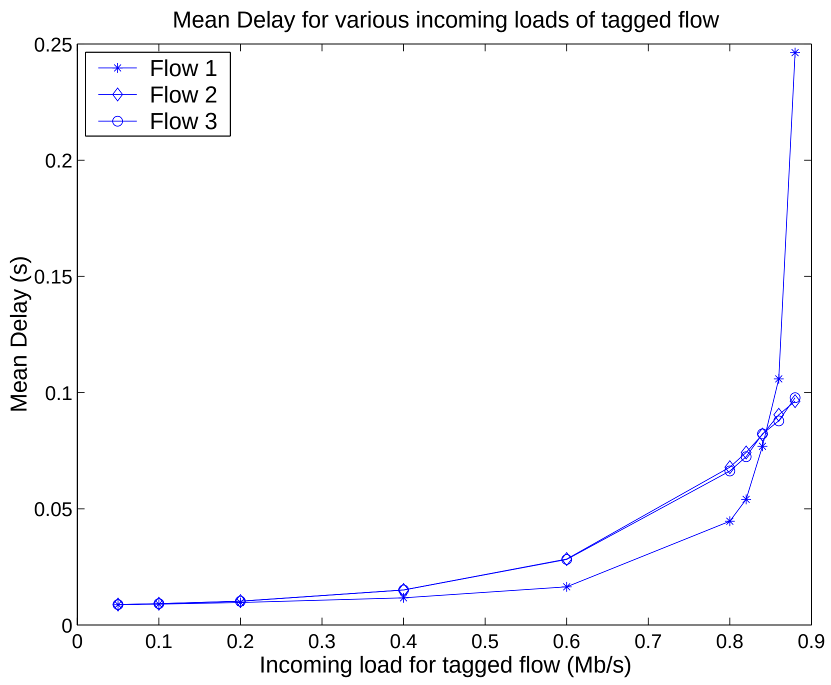

Figure A1.2 Mean Delay for WFQ: Exp. a

Case 2a:

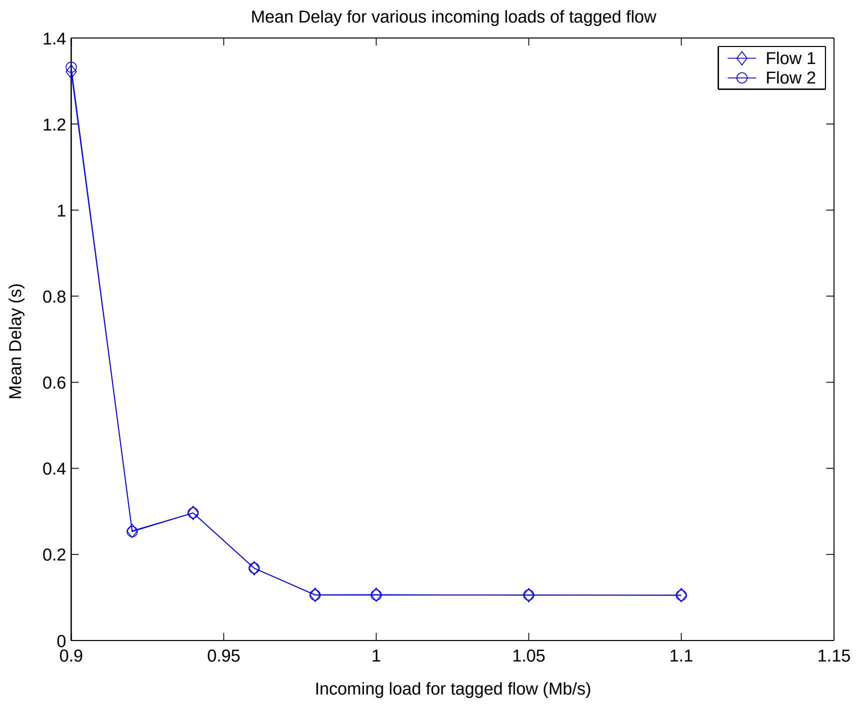

Figure A1.3 Mean Delay for HF^{2}Q: Exp. a

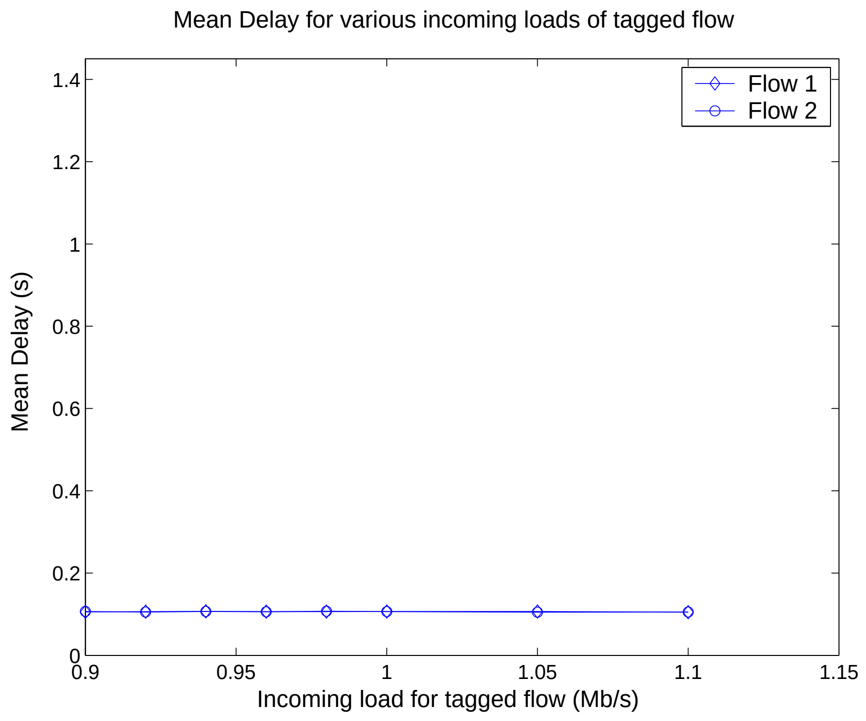

Figure A1.4 Mean Delay for WFQ: Exp. a

Case 2b:

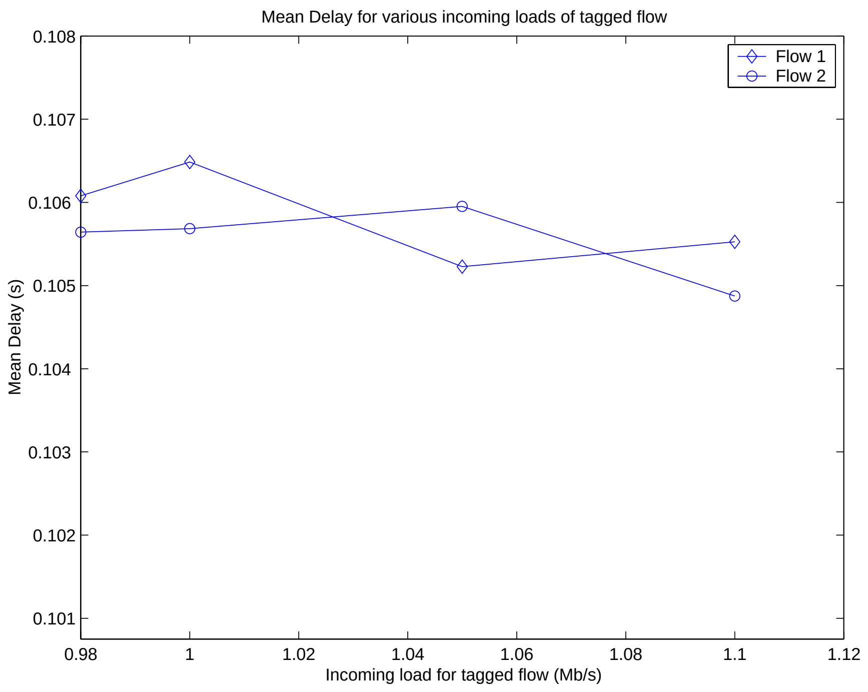

Figure A1.5 Mean Delay for HF^{2}Q: Exp. a

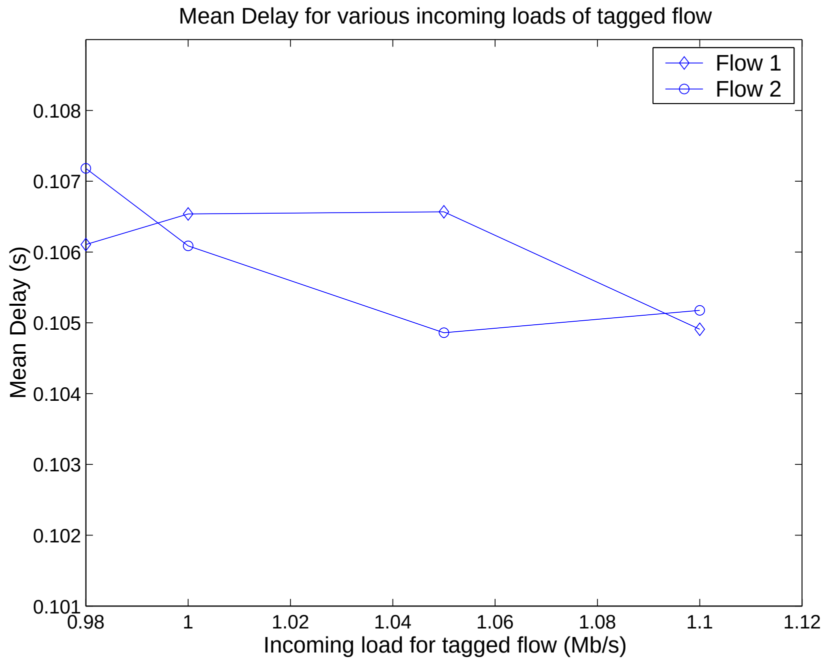

Figure A1.6 Mean Delay for WFQ: Exp. a

Experiment b:

In this experiment the load of the tagged flow varies from 0.05 Mbps to 0.25Mbps. Flows 2 and 3 have identical reservations and incoming loads. The experiment was repeated for 5 different loads of the tagged flow.

FLOW	RESERVATION (Mbps)	LOAD (Mbps)
1 (tagged)	0.2	0.05-0.25
2	0.4	0.35
3	0.4	0.35

Table A1.2 Flows setup: Exp. b

Case 1:

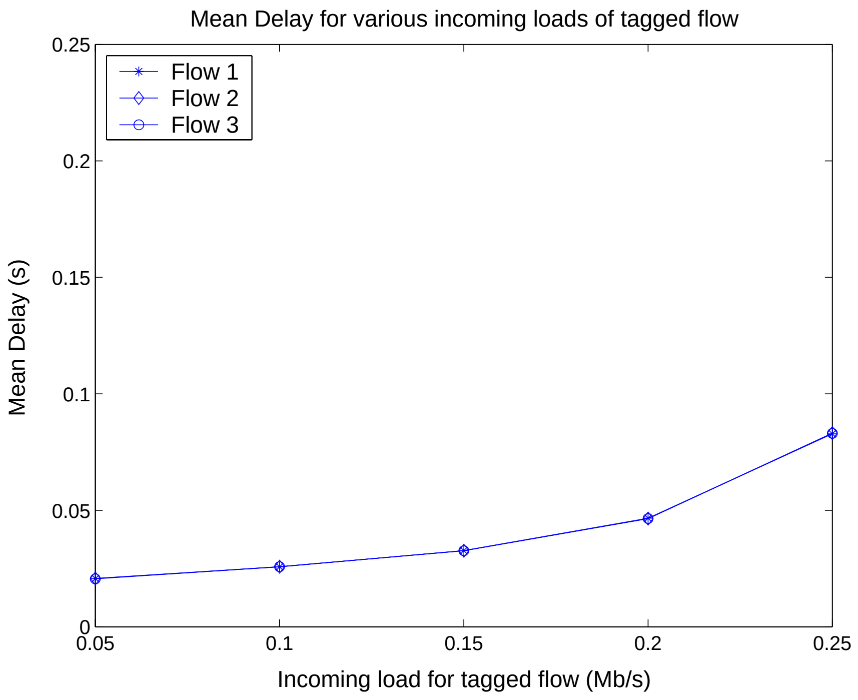

Figure A1.7. Mean Delay for HF^{2}Q: Exp. b

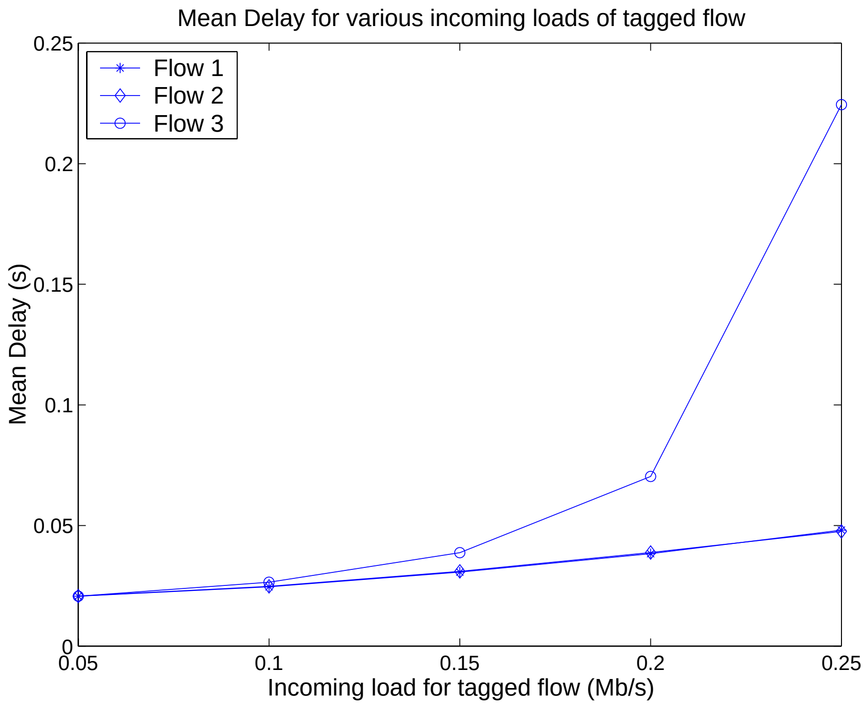

Figure A1.8 Mean Delay for WFQ: Exp. b

Case 2a:

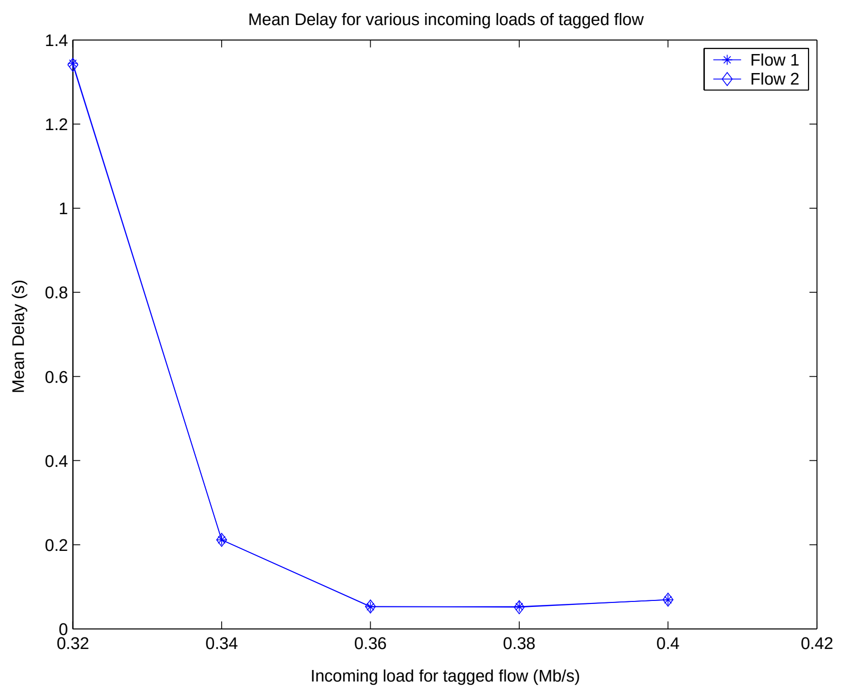

Figure A1.9 Mean Delay for HF^{2}Q: Exp. b

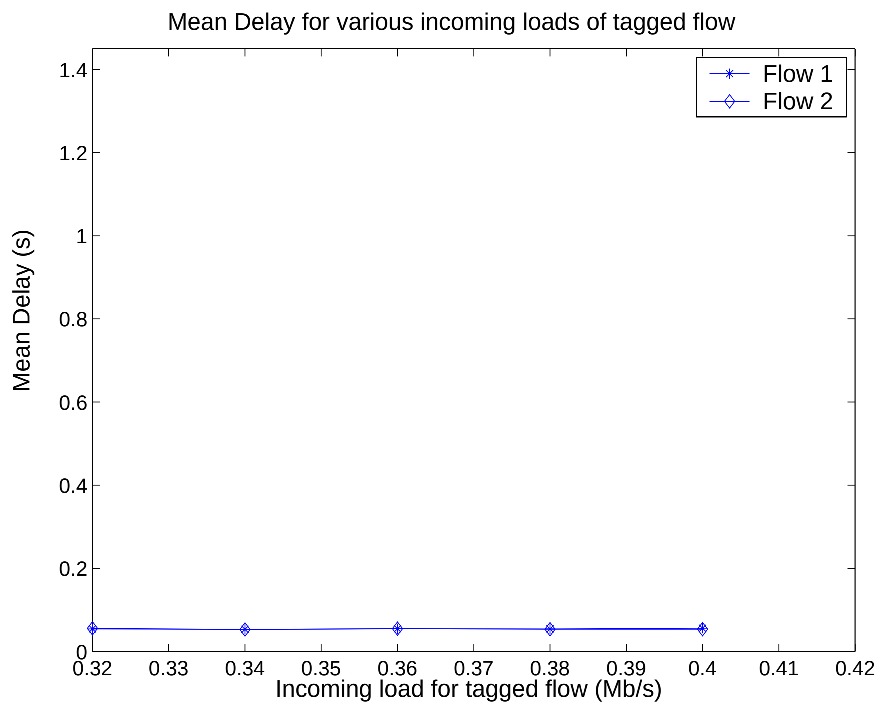

Figure A1.10 Mean Delay for WFQ: Exp. b

Case 2b:

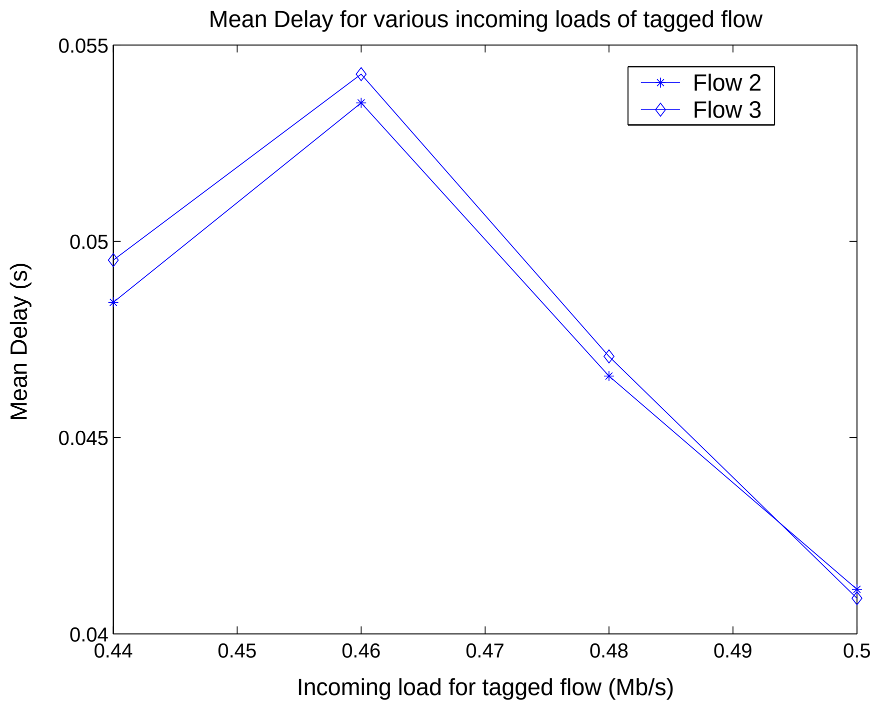

Figure A1.11 Mean Delay for HF^{2}Q: Exp. b

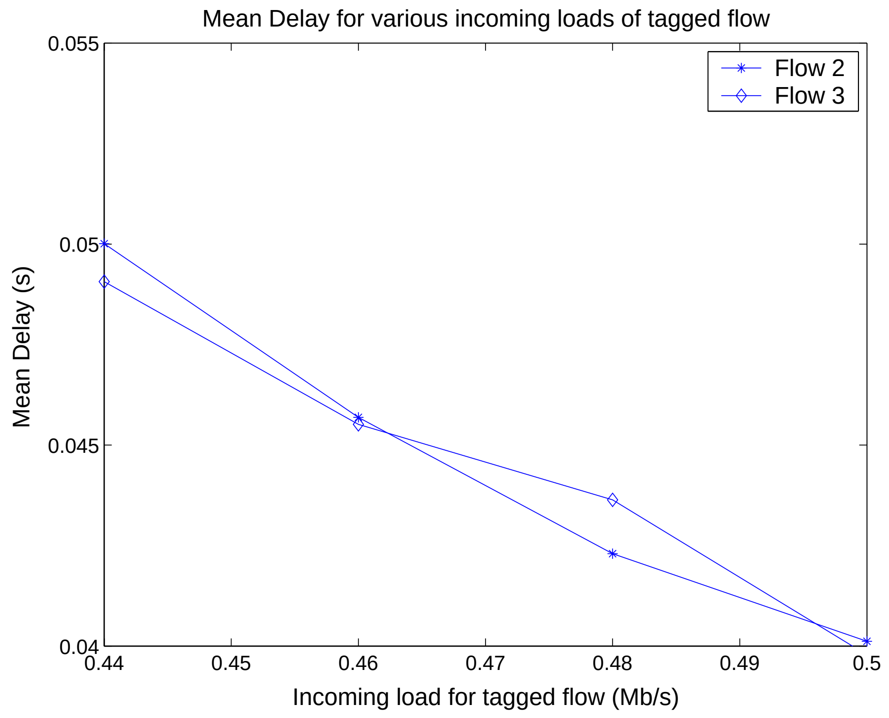

Figure A1.12 Mean Delay for HF^{2}Q: Exp. b