Location-Independent Packet Scheduling for Wireless EPON

by

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ABSTRACT

Ethernet Passive Optical Network (EPON) is the latest access technology, based on IEEE standards, utilizing an optical network architecture optimized for simple, economical delivery of voice, data, and video over native Ethernet to residential, business, and enterprise customers. With the recent explosion of wireless enterprise Local Area Networks (LANs) and the widespread implementation of Wi-Fi networks in public hot-spots the need for convergence of wired access networks (such as EPON) and *wireless enterprise LANs* is rapidly evolving and expanding. This converged wired/wireless access system will provide high-speed connection, mobility, and seamless wireless handover for customers within large enterprise buildings and campuses. The convergence can be achieved by collocating the functionalities of the EPON customer premises equipment – called Optical Network Units (ONUs) – with those of the wireless base stations. This allows ONUs to forward the information to both wired and wireless End-user Terminals (ETs). Since every customer has different needs, the converged network must ensure fair and proportional distribution of upstream bandwidth between ETs. Moreover the handover between wireless coverage areas must be designed to guarantee minimum packet delay and loss.

We present novel bandwidth allocation architecture for the hybrid (converged) network that spans both the wired and wireless legs of the wireless EPON. It is aimed to support a variety of users with different needs, characteristics and bandwidth requirements. Moreover its unique design does not limit the network to the choice of wireless or wired connection, and it supports mobility management and handover between wireless LANs. Part of the proposed architecture is a packet scheduling scheme, referred to as "Location-Independent Packet Scheduling Scheme" further mentioned as LIPS. It distributes bandwidth between ETs in the Optical Line Terminal (OLT). The OLT-based central control ensures global fairness between different Classes of Service (CoS) on a given ET, as well as with CoSs of other ETs. To meet the design goals, the OLT employs a credit pooling scheme so that the combined per-CoS traffic from all ETs will add up to the level agreed upon during the process of bandwidth negotiation. The OLT uses a weighted-share policy to determine how much of the ONU's bandwidth to allocate to each ET-CoS pair. The OLT also controls traffic from every ET in order to place a limit on the number of packets transmitted during each transmission interval. We show how the ET mobility can be easily handled by implementing a dedicated buffer to each ET at the ONUs and by extending/modifying the Standard Multi-Point Control Protocol. Dedicated buffers allow the data transmitted from the ET to be stored in the ONU, thus allowing an ONU to act on behalf of the ETs connected to it. The modifications to the MPCP Gate and Report message provide the OLT with the ability to control every ET. Thus an ET does not trigger a bandwidth re-negotiation after roaming to the new ONU.

Using simulation methodology, we evaluate the performance of LIPS and compare it to the existing dynamic bandwidth allocation algorithms. Through the simulation we prove that the LIPS scheme is capable of maintaining throughput/time-share fairness in asymmetric situations where the mobile users require different levels of service, while still offering the same level of delay performance.

We also present an improvement over the proposed architecture aimed to increase the maximum number of ETs that can be concurrently attached to a given ONU referred to as "ONU Shelving". Then we present a shadow buffer system that nullifies the handover packet loss. We compare the packet loss of the shadow buffer system to those of the original design. We conclude that ONU shelving and "Shadowing" can increase the maximum number of ETs per ONU and minimize the handover packet loss.

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TABLE OF ABBREVIATIONS USED

ADSL	Asymmetric Digital Subscriber Line
AF	Assured Forwarding
BE	Best effort
CaDAR	*Capacity and Delay Aware Routing*
CBR	Constant Bit Rate
CO	Central Office
COPS	Class-of-service Oriented Packet Scheduling
CoS	Class of service
DARA	Delay Aware Routing algorithm
DBA	Dynamic bandwidth allocation
DOCSIS	Data Over Cable Service Interface Specification
EF	Expedited Forwarding
EPON	Either Passive Optical network
ET	End-user Terminal
FTTB	Fiber-to-the-building
FTTC	Fiber-to-the-curb
FTTPC	Fiber-to-the-PC
GPON	Gigabit Passive Optical Network
IPACT (IPLS)	Interleaved Polling with Adaptive Cycle Time
MAC	Media Access Control

OLT	Optical Line Terminal
ONU	Optical Line Unit
PON	Passive Optical Network
QoS	Quality Of Service
RTT	Round Trip time
TDM	Time Division Multiplexing
WDM	Wavelength Division Multiplexing
WEPON	Wireless Ethernet Passive Optical Network
WESO	Weighted ET scheduling Order
WRED	Weighted Random early Detect
WRPON	Wavelength Routing PON
WRR	Weighted Round Robin

NOTATIONS USED

μ_{kl}	Weight that insures fair distribution of CoSs bandwidth between ETs
a	Pareto shape parameter
b	Pareto location parameter
B_k	Total number of scheduled CoSk bytes
b_{lk}	Bytes scheduled for CoS k of ET l
B_{tot}	Total number of bytes in re-arranger after round 1 of LIPS
Cp	Capacity of the EPON trunk
Dj^{mn}	Data of the ET n connected to ONU m
F	Full buffer length in bytes
g	Guard time
$Gj^{(m)}$	Gate transmission time
K	Number of CoSs supported by the system
L	Total number of ETs in the system
L_{req}	Is the size of the Request Message
L_{tot}	Total number of request in the single cycle
M	Number of ONUs in the System
N	Number of ETs per ONU
P	Packet drop probability
Pd	Packet maximum drop probability
Qa	The average size of the CoS queue in WRED Buffer
$Qmax$	The maximum queue threshold
$Qmin$	The minimum queue threshold
r_{cbr}	Rate of the CBR traffic
R_k	Rate of credit arrivals to the CoS k credit pool
r_l	Rate of credit arrivals to the ET l credit pool
$RTT^{(m)}$	Round trip time to ONU m
S	Number of shelves in the ONU
$Tmax$	The maximum cycle time
T_{sch}	LIPS scheduling time
V_k	Volume of the CoS k credit pool
v_l	Volume of the ET l credit pool

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CHAPTER 1 INTRODUCTION

In general, transport networks can be divided into three separate categories. The first one supports the information exchange between cities (long haul). The second supports the communication within an area of the city (metro). The third is the access network that supports the communication between each customer and the central office (CO). Since we consider the service of the end user, the access network is the foundation of this thesis.

We will first look at the medium on which access networks have been built. Copper wires were the original medium of information transmission [1][2]. Asymmetric Digital Subscriber Line (ADSL) and Data Over Cable Services Interface Specification (DOCSIS) are two of the most well-known methods to transmit information over copper wires in the access side. Several kinds of modulation techniques, like quadrature-amplitude modulation or discrete multi-tone, have been proposed to increase the bandwidth of the copper cable. The bit-rate-distance product of the twisted-pair copper cables is limited to around 10 Mb/s-km. Cable modems use a coaxial copper cable designed for the cable TV, and can provide greater data rates than the twisted-pair copper wire, but when many customers share its bandwidth the data transfer degrades.

Contemporary research into both ADSL and DOCSIS maximizes the utilization of copper-based lines. Thus the true bottleneck of both systems lies in the copper wire itself. Fortunately, an optical fiber can be used as the transmission medium to replace the copper wire. It provides higher bandwidth than ADSL and DOCSIS, because the bit-rate-distance product of optical fiber is at least several thousand times greater than that of copper wire. Therefore, the future of access systems is in optical fiber.

1.1 Passive Optical Network

Passive Optical Network (PON) is one technology recently proposed for fiber-based access networks. It uses centralized equipment called Optical Line Terminals (OLT) and customer's premises-based equipment called Optical Networking Units (ONUs) (Figures 1.1 and 1.2). "Passive" means that the devices connecting OLT and ONUs such as splitters are basic optical devices without any active circuitry. PON design minimizes fiber deployment due to a single fiber connecting the OLT and the splitter [3]. Moreover the PON system is ideally suited for downstream video, due to their underlying Point-To-Multipoint (P2M) structure. Passive splitters provide complete path transparency both in terms of data rate and modulation formats. Hence multiple wavelength overlay channels can be added to a PON without any modifications to the ONU logic [4][5].

Common fiber optic layouts are fiber-to-the-PC(FTTPC), fiber-to-the-building (FTTB) and fiber-to-the-curb (FTTC). Although FTTPC solutions have optimal results, since fiber is reaching all the way to individual customers, FTTB and FTTC seem to be the most economical solutions today, at least in the cases where the customer population is dispersed [5].

In the downstream direction (from OLT to ONUs), a PON is a point-to-multipoint network (Figure 1.1). The OLT typically broadcasts all the information to the network and it is up to the individual ONU to determine which data is intended for it. In the upstream direction, a PON is a multipoint-to-point network: multiple ONUs can potentially simultaneously transmit data to the OLT (Figure 1.2). The directional property of the passive splitter is such that an ONU's transmission cannot be detected by the other ONUs. Hence, simultaneous transmission from different ONUs will collide. In the upstream direction (from user to network), a medium access control mechanism must be used to avoid collisions and fairly share the trunk fiber channel capacity and resources.

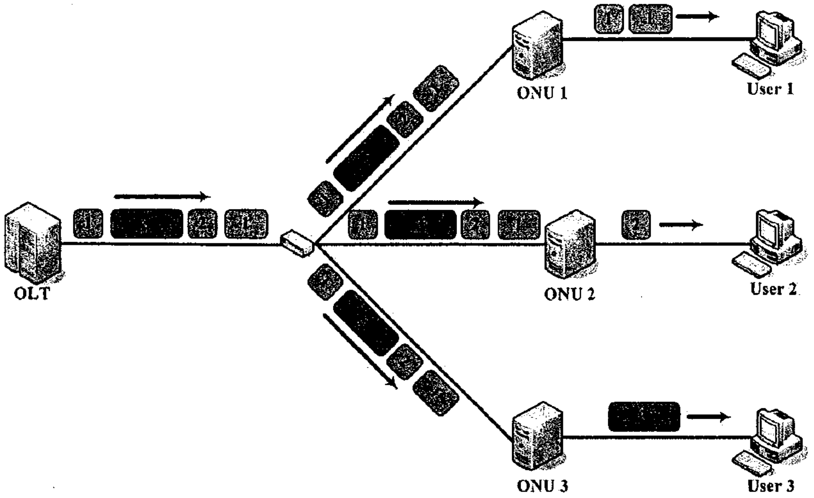


Figure 1.1 Downstream direction in PON

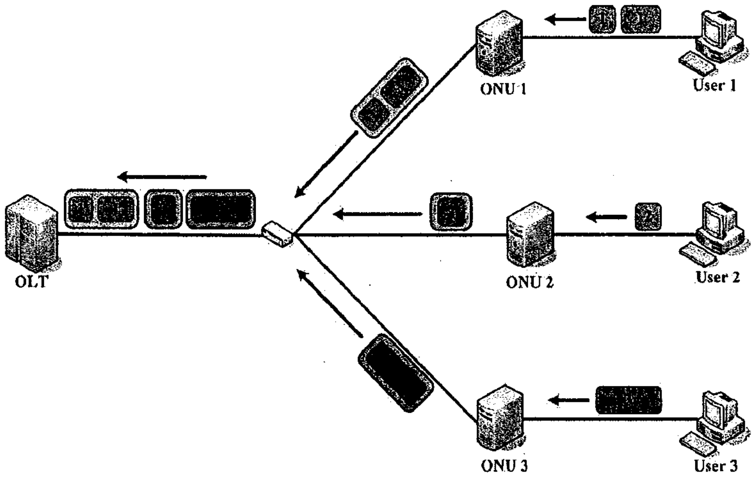


Figure 1.2 Upstream direction in PON

1.2 Wavelength Division Multiplexing (WDM) vs. Time Division Multiplexing (TDM)

There are two main methods of separating the ONU's upstream channels: Wavelength Division Multiplexing and Time Division Multiplexing [5][6][7]. Wavelength division multiplexing (WDM), in which each ONU operates on a different wavelength, is in theory a simple solution. However in practice the WDM solution would require either a tuneable receiver, or a receiver array at the OLT to receive multiple channels. Moreover the transmission will still be limited by power attenuation and splitters. An even more serious problem for network operators would be wavelength-specific ONU inventory: instead of having just one type of ONU there would be multiple types of ONUs based on their laser wavelength. It would also be more problematic for an unqualified user to replace a defective ONU, because a unit with wrong wavelength may interfere with some other ONU in the PON. Using tuneable lasers in ONUs may solve the inventory problem, but this is very cost ineffective solution at the current level of technology.

Several alternative solutions based on WDM have been proposed, including wavelength-routed PON (WRPON). A WRPON uses an arrayed waveguide grating (AWG) instead of a wavelength-independent optical splitter/combiner. This solves the problem of splitter signal attenuation. Furthermore it allows for dedicated service for each user. The major drawback of this system is an even greater cost of more complex equipment [8]. The overwhelming price tag of any WDM-based system renders it as an unattractive solution in today's PON.

In a TDM, simultaneous transmissions from several ONUs will collide when reaching the splitter. In order to prevent data collisions, each ONU must transmit in its own transmission window (timeslot). One of the major advantages of a TDM-based PON is that all ONUs can operate on the same wavelength and be absolutely identical component-wise. The OLT will also need only a single receiver. A transceiver in an ONU must operate at the full line rate, even though the bandwidth available to the ONU is lower. However, this property allows the OLT to efficiently change the bandwidth allocated to each ONU by changing the assigned timeslot size.

In future, the costs of the wave division technology will decrease. Modern TDM systems already employ WDM in order to separate upstream flow from downstream. We believe that the PON system will never fully switch to WDM, and the future lies in the co-existence of TDM and WDM systems. Further development in TDM can provide a better fusion of both systems. Today, TDM is a preferred method for optical channel sharing in an access network, and is the one used in this thesis.

1.3 EPON vs GPON

There are currently two standardization activities to build PON: Ethernet PON (EPON) and Gigabit PON (GPON). IEEE 802.3ah group is currently in charge of EPON development, while GPON is regulated by ITU-T G.984 [11][12]. Both systems operate in the upstream channel in burst mode. Burst-mode transmission slots from the two different ONUs are TDM-multiplexed and have a guard band between them. The guard time is needed to allow for turning the laser off and on between subsequent transmissions for both EPON and GPON.

EPON systems relay the Ethernet-encapsulated data arriving at the LAN subscriber port of the ONU or at the WAN Ethernet port connected to the OLT. EPON is a natural extension of the LAN system. It bridges the gap between the LAN and Ethernet-based WAN structures [8][9].

GPON on the other hand uses ATM encapsulation mechanisms to relay any data streams which are delivered at the OLT [10]. GPON strips the incoming Ethernet frames from preambles and encapsulates them in a new format, in which they are delivered to the OLT module for further processing. The incoming data frames are non-fragmentable in EPON systems, while GPON has the ability to fragment and assemble frame fragments.

In terms of the data rate, standard EPON operates at a theoretical rate of 1.25 Gbit/s (1.0 Gbit/s in practice) in accordance with the IEEE 802.3ah standard [11]. Newly developed and deployed EPON Turbo operates at an even higher rate of 2.5 Gbit/s. GPON supports downstream rates of 1.25 Gbit/s and 2.5 Gbit/s (from OLT to ONU) and upstream rates of 155 Mbit/s, 622 Mbit/s, 1.5 Gbit/s and 2.5 Gbit/s (from ONU to OLT) [12]. 10GEPON (10 Gbit/s) is also a system considered by the IEEE, however it is at its early stages of development [14].

In terms of hardware, EPON parameters are very relaxed and thus low-grade optical and electronic components can be applied. GPON requires shorter guard bands and faster electronics, which significantly increase the technical challenge level and the equipment cost. EPON systems, due to their lowered hardware requirements, are more cost-effective [9].

Another main consideration is that GPON deployments are mainly in the trial phase, while EPON has had approximately 8 million subscriber ports and a CO capacity of 16 million ports deployed, mainly in the Asian market (Japan, Korea, China, etc.) by 2007 [13]. Hence our system is based on EPON infrastructure.

1.4 Wireless Ethernet Passive Optical Network (WEPON)

Wireless EPON is a system discussed in this thesis, although similar models have recently been used elsewhere as well [15][16][17]. WEPON consists of two parts: the wireless front-end network and the optical back-end network. The wireless front-end contains mobile ETs, such as IP phones, wireless handsets, PDAs, personal computers, and notebooks. A star topology is used to connect ETs in a service area to the ONU/base station that serves that area. The ONU acts as a gateway between the wireless front-end and the optical fiber back-end by forwarding the aggregated traffic collected from ETs. The optical fiber back-end is a standard EPON network consisting of an OLT located in the operator's CO, and a plurality of ONUs, each located in a designated service area (or floor).

We have considered three different layouts for WEPON. The first layout is designed specifically for an enterprise environment, depicted in Figure 1.3. In this case, the fiber is running all the way to the building (FTTB). ONUs, co-allocated with a wireless base station, are situated on each floors to provide wireless or wired access to the ETs. This way each floor has a separate wireless coverage area. In the second layout fiber is reaching to the curb of a campus, where each ONU is responsible for its wireless zone. In this environment a user can have a wired connection to the ONU. (Figure 1.4). In these layouts configurations the customer can be wired or wirelessly connected to any ONU in the covered area. Moreover the customer can also seamlessly move from one coverage area to another.

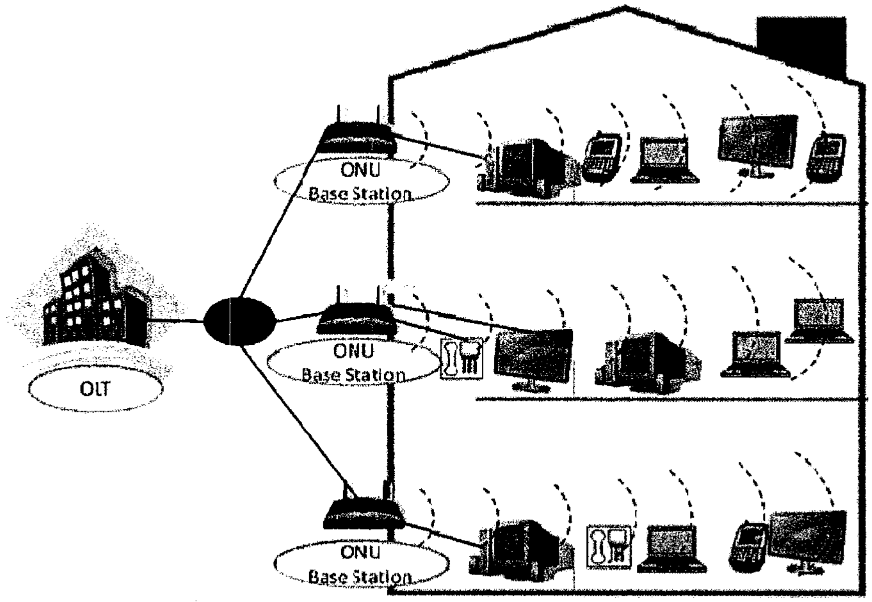


Figure 1.3: Enterprise layout

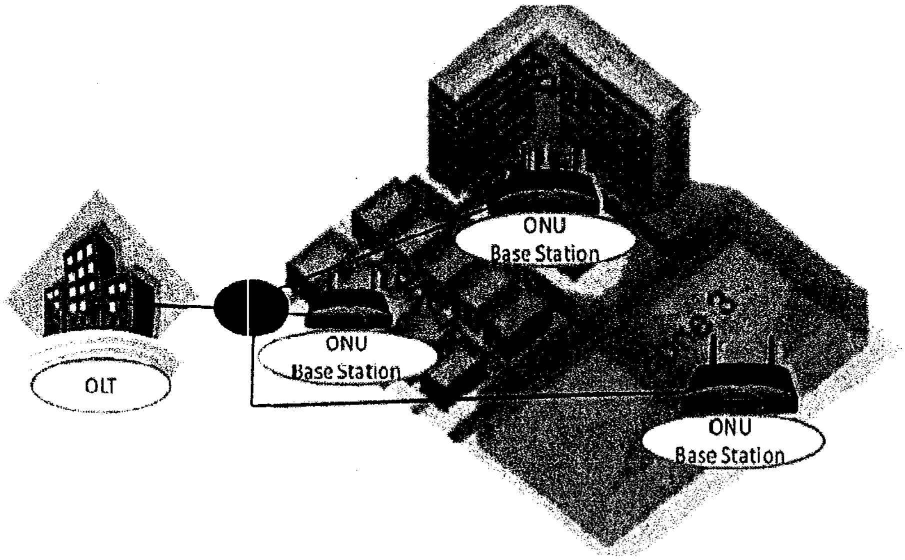


Figure 1.4: Campus layout

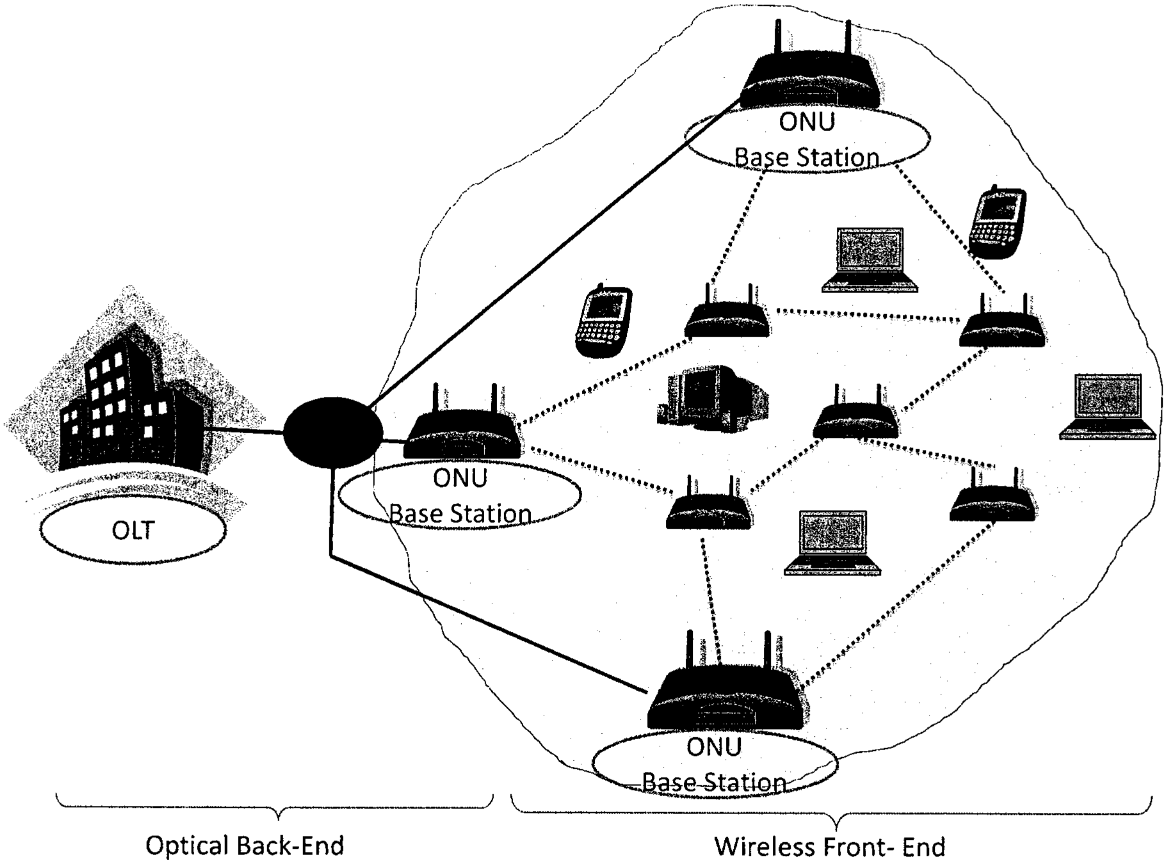


Figure 1.5: Only wireless access layout

The third layout is shown in Figure 1.5. ONUs are located in different areas and it is up to the wireless front end to relay the information from the end user to the ONUs. This layout is targeted towards Europe, as communications lines are buried below pavement and houses. Since the lines are not accessible, transition from existing copper-based lines or their repairs is not possible. Replacing copper wires with a wireless front end will eliminate the need for extensive excavation. This will result in the most economical and cost-effective approach. All three of the layouts still share the overall configuration where ONU acts as a gateway between wireless front and optics back-end and connect to the OLT through a passive optical splitter.

To the best of our knowledge modern research on the integration of wireless and fiber optics hybrid systems concentrates on the wireless front end. *Capacity and*

Delay Aware Routing (CaDAR) developed at the University of California is one such example [13]. CaDAR is a routing algorithm for the wireless front end of a WEPON. Its goal is to minimize the network delay within the wireless front-end by assigning weights to different wireless links. Two wireless nodes have a link between them if their distance is less than their respective transmission range. CaDAR is designed mostly for the third configuration shown in Figure 1.5. Therefore it also minimizes the average number of intermediate wireless nodes needed to transmit information from an ONU to an ET.

M. Kiese, E.Georgieva, D.Schupke et al, in one of their latest papers on wireless optical access networks investigated the influence of different routing strategies on bandwidth availability [16]. Their conclusion was that the wireless bandwidth availability is much lower than that of the fiber optics. Hence the main consideration was given to the wireless front end. The main conclusion reached by the authors is that at wireless front-end the bandwidth availability is increased when the multi-path routing is considered versus single-path.

Delay-Aware Routing Algorithm (DARA) was another research work on WEPON [17]. The study was focused on wireless front-end connectivity and routing properties. Their proposed design was proven to minimize the average packet delay in the wireless front-end of a WEPON. As in previous designs, the EPON backend was not considered in their simulation.

EPON-WiMAX integration was another proposition with the setup similar to the one of WEPON [18]. The main goal of their paper was efficient deployment of network, meaning positioning of the ONUs in the network and a probability of the units failure. The authors provided a case study to demonstrate their infrastructure cost and performance gain in spectra efficiency. They concluded that both of these factors can be improved through carefully considering the ONU locations and the probability of the unit failure.

Although the above studies shared our proposed infrastructure (WEPON), we cannot provide a comparison between their study and ours. The main reason is that this thesis concentrates on the back-end optical network while the designs and algorithms previously discussed concentrate on the front-end wireless part of WEPON.

1.5 Challenges in Fiber Optical Back End of WEPON

When designing a system for the back end of WEPON there are three issues to be considered:

1) One of the major challenges to be faced when designing a bandwidth allocation algorithm for WEPON is the arbitration of access to the common medium by ONUs in the upstream direction (to the OLT).

2) Another design challenge is the need to support a broad spectrum of applications with *Quality of Service* (QoS) requirements. End-user terminals today are mobilized to generate competing multimedia traffic of different service categories, such as data, voice, video, and online gaming. Indeed, many wireless handsets (such as iPOD) are presently used as a handheld computer, phone, and video/gaming player by employees within the offices and by ordinary users in public areas. The mobilization of a single end-user terminal to access multimedia applications and the integration of these services to a converged network require a networking infrastructure that can ensure the quality of coverage and can handle each traffic type according to its unique QoS requirement. In today's IP-based networks, QoS is provided by classifying packets into "classes" at the boundary of the network, and by servicing packets in each *class of service* (CoS) according to some queuing discipline (such as priority queuing).

3) A further challenge is to provide a seamless handover of ETs between ONU regions. The key to this challenge is to maintain global control over every ET. Since an ONU has no global cover, it cannot be aware of the location of ETs in the WEPON system. Therefore, OLT control is the most logical and efficient method. Moreover an OLT can specify to each ONU whether the newly arrived ET is a recently moved one or completely new to the system.

1.6 Review of Existing Bandwidth Allocation Schemes for EPON

One of the earlier solutions was a basic static TDM-based MAC. This was one of the first solutions proposed for EPON [19]. In this research the authors decided to assign a fixed-size transmission slot to every ONU. The transmission slots from all ONUs form a transmission cycle. Although this solution is very simple but has the following primary limitations: it requires network-wide synchronization of ONUs with the OLT; and it is not adaptive to the variation of traffic, which can result in a substantial underutilization of the network.

The second proposition was a slight upgrade over the first solution [19]. In it the ONUs are divided into two disjoint sets: bandwidth-guaranteed and best-effort. Within each transmission cycle, one or more timeslots are allocated to bandwidth-guaranteed ONUs, while the remaining unused timeslots are used toward the best-effort ONUs.

A more dynamic solution called Interleaved Polling with Adaptive Cycle Time (IPACT) has also been proposed for EPON [21]. IPACT assigns transmission slots to ONUs dynamically. It uses a polling mechanism with Request and Grant messaging between the OLT and ONUs. A Grant message is used by the OLT to assign a variable-size transmission slot to an ONU. A Request message is used by the ONU to convey its local buffer occupancy to the OLT. The ONU transmits Ethernet frames stored in its buffer during its transmission slot.

The same authors have combined the basic IPACT scheme (inter-ONU scheduling) with strict priority queuing (intra-ONU scheduling) in order to support different CoSs [20]. Every ONU maintains a separate queue for each CoS in its buffer. The OLT issues "colorless" grants to ONUs, which means that the OLT does not dictate how many bytes from a particular queue an ONU must transmit. Instead, the ONU uses a strict priority policy to determine the order in which the queues are processed. This scheme leads to a phenomenon known as light-load penalty, where the queuing delay for some traffic classes increases when the network load decreases. In order to tackle the light load penalty, a rate-based optimization scheme has been proposed; however, this scheme has been able to only eliminate the penalty for the second priority queue, not the other subsequent queues [21]. Another problem with this scheme is that it fails to distribute the upstream bandwidth fairly among the users. As an example, if two identical users access two different ONUs with identical requests, these users can receive very different qualities of service when one of the ONUs is lightly loaded and the other ONU is fully loaded.

A completely new design of dynamic bandwidth scheme, called *Class-of-service Oriented Packet Scheduling* (COPS), was given in [22]. The COPS scheme employs a credit pooling technique together with a weighted-share policy to ensure that transmission grants for the upcoming "transmission interval (cycle)" are distributed among different ONUs, based on their service level agreements. By being a OLT-based centralized DBA scheme, COPS is superior to other existing DBA schemes in terms of providing global fairness between different classes of service.

1.7 The main Contribution of This Thesis

The previous solutions discussed in Section 1.6 deal with improving the fiber optics part of the WEPON; hence we will use IPACT and COPS as baseline comparison of our algorithms. As previously mentioned IPACT and COPS can provide CoS fairness, but only to the ONU layer. As a consequence, these approaches fail to provide fair bandwidth allocation to the actual users (in our case ETs). Moreover, since each ET is owned by a different user, spatial fairness must be provided between ETs in order to guarantee deterministic access to the WEPON bandwidth. Since mobility of the ETs was not considered in COPS or IPACT there is no mobility management in both schemes.

In this thesis, however, we propose a novel dynamic bandwidth allocation scheme for the back-end (optical) part of the integrated WEPON. Our system concentrates on the end users, thus guaranteeing the bandwidth and providing wired and wireless connections to the ETs. "Location-Independent Packet Scheduling Scheme" (LIPS) is introduced as an integral part of the proposed system, is an OLT-based, thus allows an OLT to allocate bandwidth between different ETs. Since the OLT has global knowledge of the network, LIPS can provide global fairness between different CoSs on a given ET, as well as with CoSs of other ETs. The OLT employs a credit pooling scheme similar to the one of COPS, that allows ONUs to request bandwidth on behalf of their ETs so that the combined per-CoS traffic from all ETs will add up to the level agreed upon during the process of bandwidth negotiation[22]. The OLT uses a weighted-share policy to determine how much of the ONU's bandwidth to allocate to each ET-CoS pair, which is dependent upon their weights.

Using our simulation, which is written in C++, we study LIPS scheme and compare it to previously mentioned IPACT and COPS schemes. We present the results that show that LIPS scheme is remarkable in providing the fair and deterministic service all the way to the end user. Moreover we proved that throughput fairness can be guaranteed for each ET no matter the type of connection or location in the system. Even though LIPS concentrates on the throughput fairness it is still comparable to schemes presented in Section 1.6 in terms of delay statistics. After analyzing the simulation results, we propose two improvements over our original system. The first improvement, referred to as "ONU shelving", is designed to maximize the number ETs that can simultaneously be connected to a single ONU. The second improvement introduces the "shadow" buffering scheme. In this scheme ONU does not discard the packets of the ET that roamed to a new ONU. Instead the OLT keeps issuing transmission grants to the old ONU until all of the packets of interest are transmitted. By running the simulation with the "shadow" buffering scheme, we concluded that most of the packet loss due to handover seen in LIPS was minimized. Through minimizing the packet loss and maximizing the number of ETs per ONU, we improved LIPS to be an ideal solution for optical part of the integrated WEPON.

1.7 Thesis Organization

The rest of the thesis is organized as follows. In Chapter 2 we introduce the model used for the WEPON network in this thesis. In Chapter 3 we review the standard *Multi-Point Control Protocol* (MPCP) developed by IEEE to arbitrate access to the upstream transmission channels (timeslots) in EPON. Then we present the "Modified MPCP protocol" proposed in this thesis that supports bandwidth allocation in the integrated WEPON network. Chapter 4 introduces the scheduling architecture. Then gives the details of the proposed dynamic bandwidth allocation system and presents the LIPS scheduling algorithm. Chapter 5 presents the simulation setup used for the experiments and the results obtained from the simulations. Chapter 6 focuses on the improvements proposed over standard LIPS architecture. Chapter 7 concludes this thesis and offers some ideas for future work.

CHAPTER 2 SYSTEM MODEL

As we can see in Figure 2.1, the system is based on the EPON structure. It consists of an OLT connected to *M* ONUs using a tree topology. Each ONU supports up to *N* ETs connected to it wired or wirelessly. An ONU can service up to eight CoSs for each ET. CoSs may be used for delivering voice, video, and data traffic. They can also be mapped to standardized classes of service defined in Diffserv [20]. For instance, CoS1 can be mapped to Expedited Forwarding (EF), which provides low loss and delay-sensitive services; CoS2 can be mapped to Assured Forwarding (AF), which provides low loss services; and CoS3 can be mapped to Best Effort (BE), which does not require any commitment from the network.

In this chapter we will look at the ONU structure and the ET mobility protocols needed to improve the system.

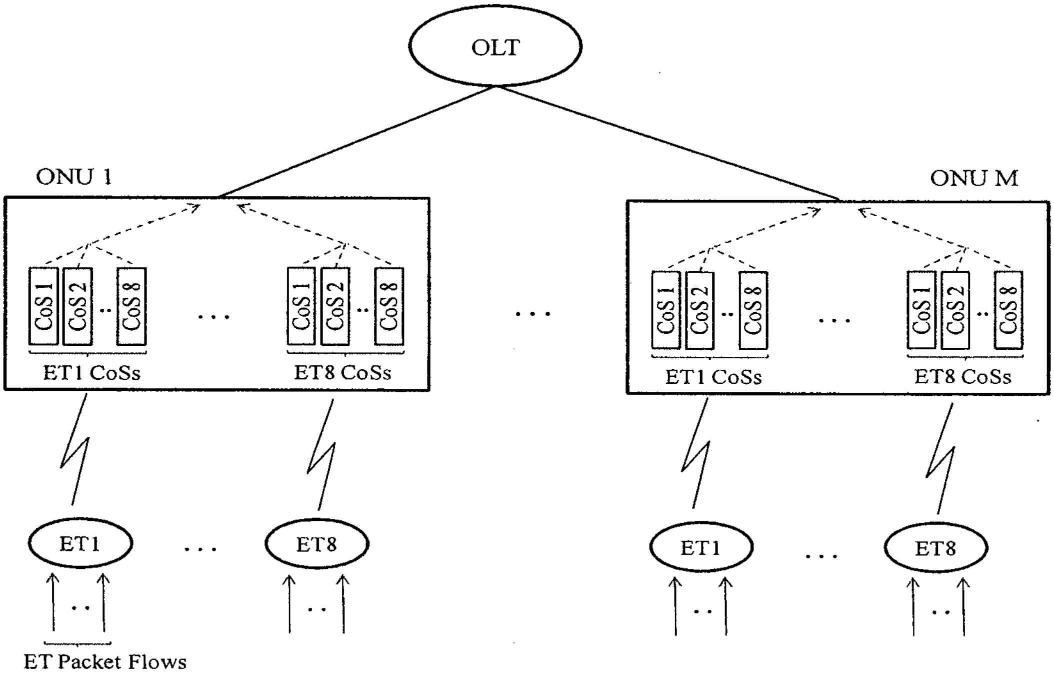


Figure 2.1 System model

2.1 ONU Structure

In our system ONUs need to act on behalf of the ETs on a regular basis; therefore information has to be stored inside the ONU. This not only takes the complexity away from ETs, making them more economically feasible; it also eliminates the need in extra hardware for end users. The structure of the ONU is reflected in Figure 2.1. This structure allows ETs to move freely within the wireless area of the same ONU. The delay between the ETs and ONUs will have no influence on the system because the ONU's behaviour is based on the status of the queues.

For an ONU to store packets from its N ETs, it employs N physically separated Weighted Random Early Detect (WRED) buffers [23]. The CoSs inside each WRED are logically separated but physically combined. The average size of the queue for each CoS stored inside a WRED buffer is denoted as Q_a. On the arrival of a CoS k packet at the ONU from a specific ET, the WRED computes a new value for Q_a using an exponential weighted moving average

$$Q_a = \mu * Q_i + (1 - \mu) * Q_a \tag{1}$$

where μ is a weight parameter, and Q_i is the current queue size for all CoSs of that particular ET.

The WRED will then decide to whether or not to drop the newly arrived packet based on the outcome of the following tests:

1) if Q_a is less than a predetermined minimum queue threshold (Q_{min}), the packet will be accepted;
2) if Q_a exceeds a predetermined maximum queue threshold (Q_{max}), the packet will be dropped;
3) Otherwise, the packet will be dropped with the probability (P) :

$$P = P_d * \frac{Q_a - Q_{min}}{Q_{max} - Q_{min}} \tag{2}$$

where P_d is a maximum drop probability.

2.2 ET Mobility

ET mobility is the ability of the system to support movement of the ETs between wireless areas of the ONUs. Although it would seem to be a logical problem for WEPON to consider, no previous research has considered the problem from the fiber optic point of view. For a smooth transition of ETs between ONUs, an OLT files information regarding the priority, bandwidth requirements and identity of each ET connected to it. The OLT must provide ETs with the time required for proper self-identification through the handover or discovery process. This will allow the OLT to identify the ET, thus assigning the ET its proper priority and bandwidth requirements.

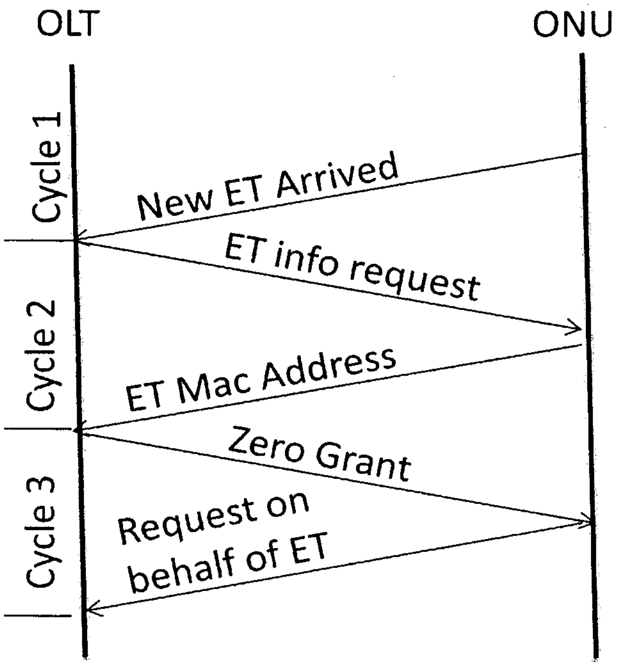


Figure 2.2 ET discovery

2.2.1 Discover Process

Discovery is the process were newly connected ETs get a chance to inform OLT of their presence. If an ONU contains space for the newly arrived ET, it automatically buffers all the data received from the ET. Then a handshake process between the ONU and the OLT occurs. This handshake procedure takes the following three cycles to complete (Figure 2.2).

1) The ONU transmits a modified Report message to inform the OLT of the newly arrived ET. The details of the modified Report Message are discussed in Section 3.2.1
2) Upon receiving the Report Message the OLT sends the "ET info request" message to the ONU via the Gate Message, to determine the identity of the ET. Within the same cycle the ONU transmits the ET's media access control (MAC) address back to the OLT through the Report message.
3) Upon receiving the ET MAC address the OLT identifies whether the ET is registered with the system, hence a valid customer. After the identification there are two possible outcomes.
   a) If ET is registered: the OLT sends a zero grant Gate message to the ONU. In response the ONU sends a Request message on the behalf of the newly connected ET.
   b) Otherwise if the ET is not registered, the OLT sends a "clear" message. Thus refusing the ET's connection to the system. Immediately after receiving the "clear" message, the ONU clears all the information stored in its buffer related to the ET.

2.2.2 Handover Process

When an ET roams from one ONU to another ONU as shown in figure 2.3, the new ONU must notify the OLT of the ET transition. Hence a handover between the two ONUs is required. This handover incorporates a three-way handshake between ONU1, ONU2 and the OLT (Figure 2.4). Where ONU1 is the ONU that previously hosted the ET and ONU2 is a new host of the ET. The three-way handshake requires the following three cycles:

1) The ONU2 transmits a modified Report message to inform the OLT of the newly arrived ET.
2) The OLT sends an "ET info request" message to ONU2 via the Gate Message, to determine the identity of the ET. Within the same cycle ONU2 responds with a Report message containing the ET's media access control (MAC) address.
3) Upon receiving the ET MAC address the OLT identifies that the ET roamed from ONU1 to ONU2. In response, the OLT sends a zero grant Gate message to ONU2, and a "clear" message to ONU1, concluding the handshake. To finalize the handover process, ONU1 clears all the information stored in its buffer related to the ET and ONU2 sends a request on behalf of the ET.

All previous research paid no attention to the ET bandwidth allocation; standard MPCP protocol message format was sufficient enough to transmit all the controlled information. In this system an OLT needs to convey more detailed information hence an expansion to the standard MPCP protocol message format is needed.

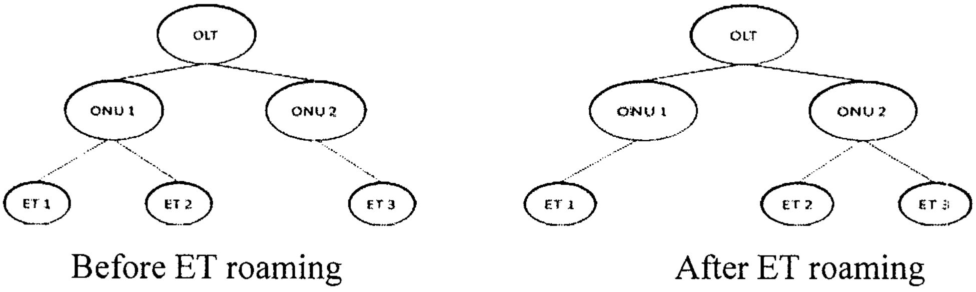


Figure 2.3 ET roaming between ONUs

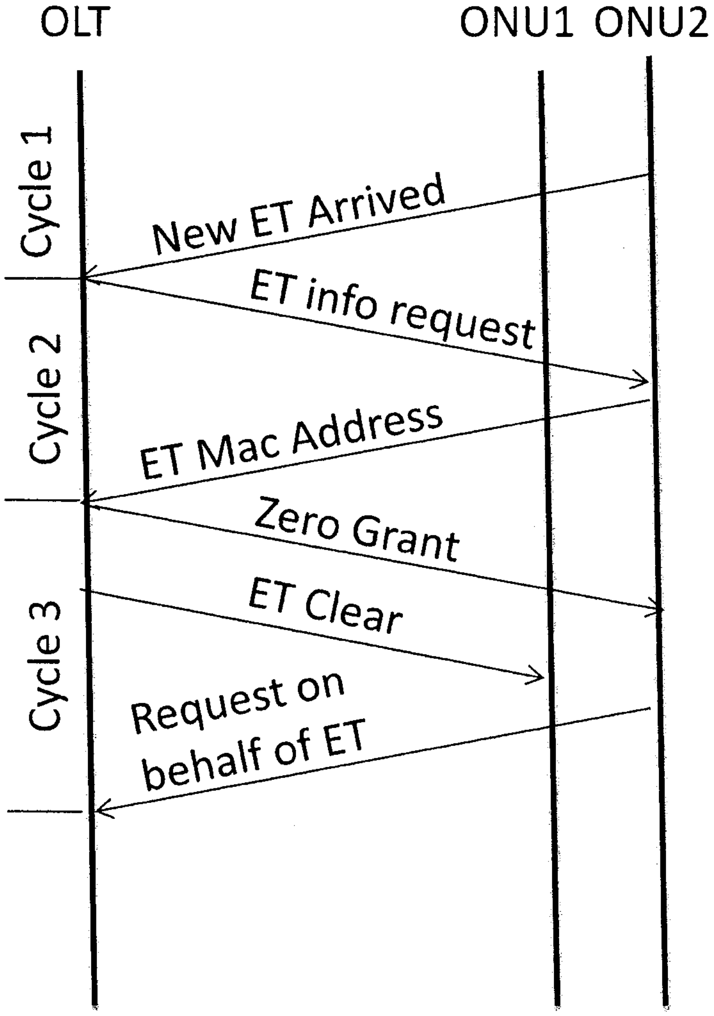


Figure 2.4 ET handover

CHAPTER 3 MPCP PROTOCOL

The IEEE 802.3ah working group is defining standards for the control and management of information transfer between OLT and ONUs in the EPON system [11]. This group has developed a *Multi-Point Control Protocol* (MPCP) that arbitrates the ONUs' transmission and supports bandwidth allocation by the OLT. MPCP defines the precise format of the request (referred to as Report) and grant (Gate) messages. For a full understanding of modifications needed we will first look at the standardized MPCP messages.

3.1 Standard MPCP Messages

Both Report and Gate messages need to have preamble, destination and source address, length type, opcode and timestamp fields (Figures 3.2, 3.4, 3.5, 3.6). The Destination and Source addresses are six-byte fields; they are used to specify where the message came from and where is it heading. The opcode specifies the type of message, for example gate or report. The Length field is used to tell how many bytes constitute the data field. Timestamp is a four-byte field; its value is equal to the clock value inside the source unit. ONUs update their clock every time they receive a valid Gate message from the OLT. The OLT on the other hand measures the round-trip time (RTT) using the timestamp as follows (Figure 3.1):

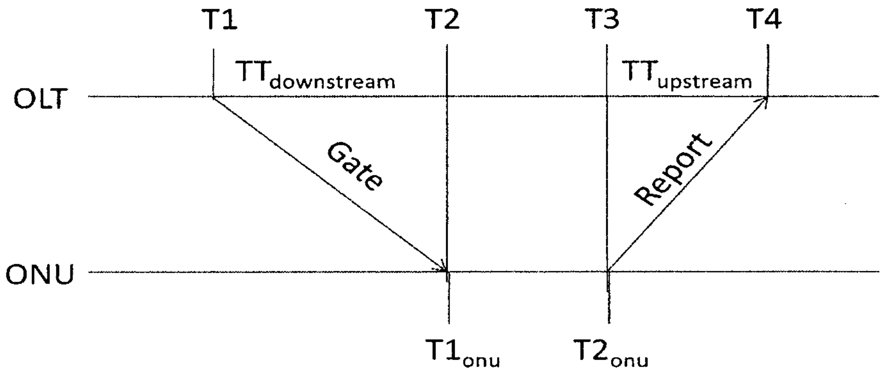


Figure 3.1 Round-Trip Time measurement

1) OLT sends Gate at $T1$
2) ONU receives Gate at $T2$, and sets its local clock to $T1_{onu}$
3) ONU sends Report at time $T3$, showing timestamp $T2_{onu}$
4) OLT receives Report with $T2_{onu}$ at $T4$. Thus the RTT is calculated as follows:

$$\begin{aligned} \boldsymbol{RTT} &= \boldsymbol{TT_{downstream}} + \boldsymbol{TT_{upstream}} = \boldsymbol{T2} - \boldsymbol{T1} + \boldsymbol{T4} - \boldsymbol{T3} \\ &= \boldsymbol{T4} - \boldsymbol{T1} - (\boldsymbol{T3} - \boldsymbol{T2}) = \boldsymbol{T4} - \boldsymbol{T1} - (\boldsymbol{T2_{onu}} - \boldsymbol{T1_{onu}}) \end{aligned} \tag{3}$$

Since the value of $T1$ and $T1_{onu}$ is the same, RTT = $T4-T2_{onu}$

3.1.1 Report Message

The Report Process has the responsibility of dealing with queue report generation. The IEEE working group has also introduced the concept of *threshold queue reporting* which is used by the ONUs to report detailed information regarding the status of the individual queues. The ONU specifies the queue occupancy based on certain fixed limits (thresholds). For each threshold (TH), the ONU sends the combined length of all packets up to that threshold. Figure 3.2 illustrates an example of threshold queuing. The ONU would report 80 bytes for TH1, 150 bytes for TH2, and 220 bytes for TH3 etc.

An ONU transmits its current bandwidth requirements to the OLT by means of a Report message (Figure 3.3). These requirements are indicated by the number of bytes waiting in each queue below each threshold. Since there are only eight Queue Report (QR) fields in the Report message and eight priority queues in the ONU, it would imply that only one threshold state per each priority queue can be specified. However, within MPCP, there can be several QRs for one priority queue in a single Report message (allowing an ONU to provide some information on the frame bounds as well). This is achieved through the "Number of Report sets" and the "Report Bitmap" fields. The "Number of Report Sets" specifies the number of report sets included in this Report message. While the "Report Bitmap" specifies which priority queues are represented in the report set. As an example, 10010001 in the Report bitmap field indicates that the queue reports for the priority 0, 3 and 7 queues follow the "Report Bitmap" field.

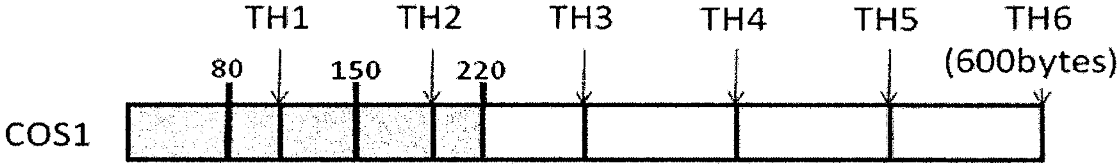
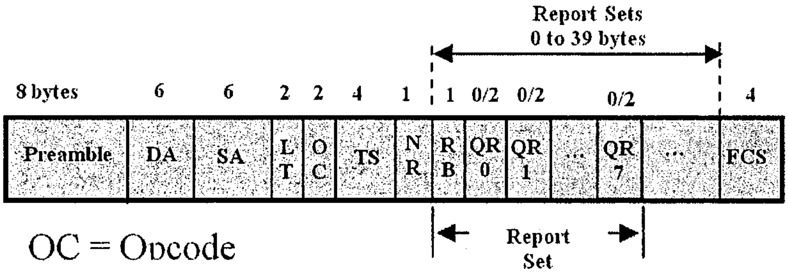


Figure 3.2 Example on threshold queuing



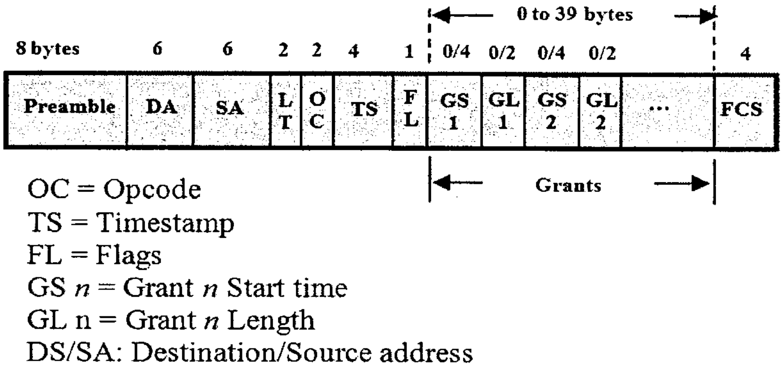
OC = Opcode
TS = Timestamp
NR = Number of Report Sets
RB = Report Bitmap
QR *n* = Queue *n* Report
DS/SA: Destination/Source Address

Figure 3.3 Standard Report message

3.1.2 Gate Message

The multiplexing of multiple transmitters is achieved through a 64-byte-long Gate message. It is needed for an OLT to specify two parameters to the addressed ONU. First the OLT specifies when the ONU is allowed to start the transmission through the "Grant Start Time" field. Second the OLT specifies the length of the ONU's transmission window through the "Grant Length" field. From Figure 3.4 we can see that there can be multiple grants within a single Gate message. The total size of grant field in the Gate message is 40 bytes as indicated on Figure 3.4. Each grant requires the "Grant start time" and "Grant length" fields adding up to six bytes. Hence there can be only six grants in a Gate message. However MPCP states that only four grants are used in a Gate message. There are three separate flags in the "Flags field". The "Number of grants" flag specifies the number of grants in this Gate message. The "Force Report" flag ask the ONU to issue a Report message related to the corresponding grant. While the "Discovery" flag indicates that the signalled grants would be used for the discovery process.

Discovery is the process whereby newly connected or off-line ONUs are provided access to the PON. The process is driven by the OLT, which periodically makes Discovery Time Windows available, when off-line ONU's are given the opportunity to make themselves known to the OLT. Although the periodicity of these windows is unspecified by the MPCP protocol, it is suggested to be greater than the maximum cycle time discussed in Chapter 4. Off-line ONUs, upon receiving this message, wait for the discovery period to begin and then transmit a Request message to the OLT. Discovery windows are unique providing that multiple ONUs can access the PON simultaneously, thus transmission overlap can occur. Measures are taken to reduce the probability of overlaps by introducing a random delay before the ONU report message.



OC = Opcode
TS = Timestamp
FL = Flags
GS *n* = Grant *n* Start time
GL n = Grant *n* Length
DS/SA: Destination/Source address

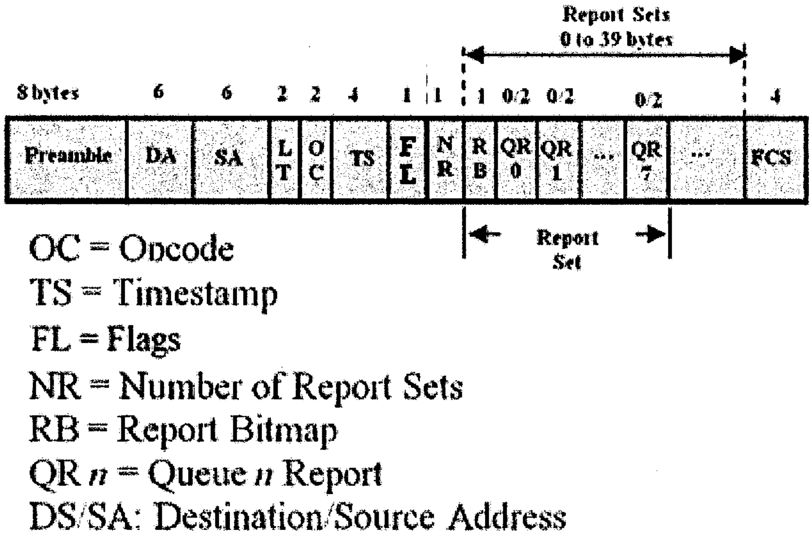
Figure 3.4 Standard Gate message

3.2 Modified MPCP Messages

In this research an OLT provides more detailed information; it specifies how many packets from each CoS-ET pair an ONU is allowed to transmit during each transmission cycle. Moreover the current report process should be improved to provide the information needed for our algorithm: "new ET". It is needed for an ONU to notify the OLT of a new ET connected to it, as described in Section 2.2

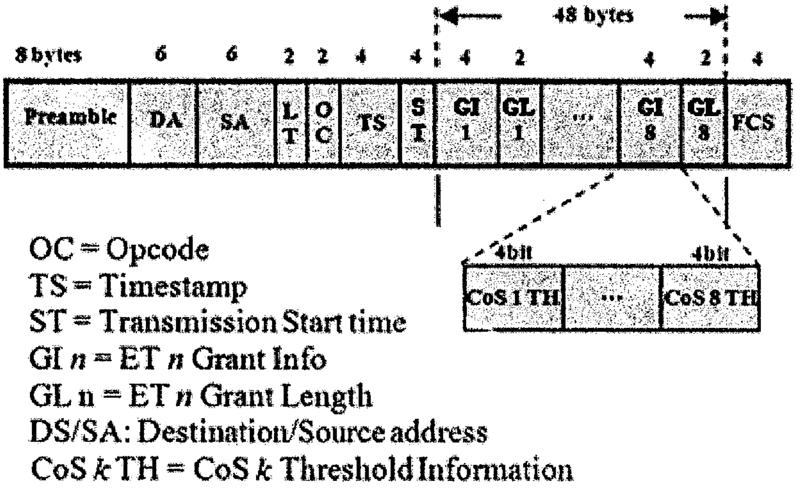
3.2.1 Modified Report Message

Consider the change in the network depicted in Figure 2.3. Although it only shows that ET2 has roamed from ONU 1 to ONU 2, with standard report message ONU2 would have no means to notify the OLT of a newly connected ET. In search for a solution first it was considered to use the discovery window time for an ONU to make the notification. Since the discovery does not occur every transmission cycle, this solution was discarded due to the delay between two consecutive discovery periods. Instead a one-byte "Flags" window was added to the Report message. The "Flags" window is proposed to be positioned between "timestamp" and "number of report sets" (Figure 3.5). Currently one bit of the flag is used for the "new ET" flag. When the "new ET" flag is set during any of the ONU reports, then a newly arrived ET is connected to the ONU. The second flag referred to as the "Shadow" flag will be added in Section 6.2 to improve the standard LIPS buffering system. The rest of the flag window can be used for future development, such as improved versatility in OLT control over ETs



OC = Opcode
TS = Timestamp
FL = Flags
NR = Number of Report Sets
RB = Report Bitmap
QR *n* = Queue *n* Report
DS/SA: Destination/Source Address

Figure 3.5 Modified Report message



OC = Opcode
TS = Timestamp
ST = Transmission Start time
GI *n* = ET *n* Grant Info
GL n = ET *n* Grant Length
DS/SA: Destination/Source address
CoS *k* TH = CoS *k* Threshold Information

Figure 3.6 Modified Gate message

3.2.2 Modified Gate Message

With the limitation of grants in the standard Gate message, the OLT can not specify the exact number of bytes granted to each ET-CoS pair. This drawback requires the OLT to send a Gate message for each ET connected to the ONU. We propose the following modification to the Gate message to eliminate its necessity for individual ETs. Thus a single Gate message can be sent to the ONU with all required specifications. First we modify the structure and interpretation of the Grants field of the Gate message. Second we use of the concept of *threshold grant reporting* which is analogous to the threshold queue reporting used in the Report message.

To use the threshold grant reporting the OLT is configured with the queue threshold values T1 to T13 used at the ONUs for each of the CoS queues. When the OLT computes a grant for a given CoS, the OLT sends the next higher threshold value, instead of sending the exact length of the computed grant. The structure of the modified Gate message is depicted in Figure 3.6. By comparing this figure with Figure 3.4 one can see that the "Transmission Start Time", "ET #*n* Grant Info", and "ET #*n* Grant Length" are newly added fields. "Transmission Start Time" is a 32-bit unsigned field. The ONU compares the "Transmission Start Time" to the local clock to determine when to begin grant transmission. Up to eight CoS queues are provisioned for each active ET in the ONU as already illustrated in Figure 2.1. The data from these queues are transmitted back-to-back to the OLT within a given burst. The logic for selecting from which specific queue the ONU must transmit data and how much data will be transmitted is provided by the "ET #n Grant Info" and "ET #n Grant Length" fields. "ET #n Grant Length" is a 2-bytes unsigned field that specifies the physical length of the ET grant. "ET #n Grant Info" is 8 hexadecimal digits long thus requiring a 4-byte field. Each hexadecimal digit corresponds to a "CoS *k* Grant threshold Information" (CoS *k*

TH) field as can be seen in the bottom of the Figure 3.6. Each "CoS *k* TH" encodes the one of the 13 grant threshold levels (T1 to T13). These levels will dictate the maximum number of bytes the ONU can transmit from the given CoS *k*-ET *n* queue. The 13 threshold values are represented as 1_{16} through D_{16} in hexadecimal. This allows the hexadecimal values of E_{16} and F_{16} to be used for control messages (Table 3.1).

Table 3.1—ET #*n* Grant Info

Field Value (hex)	*n (dec)*	Description
EFFFFFFF	1	Discovery Gate
FFFFFFFF	X	CLEAR
REST	X	Threshold Information

Discovery Gate Message: is specified by setting the field value to $EFFFFFF_{16}$ for the first ET, and it is used to state to the ONU that the current gate is a discovery gate.

CLEAR: If the hex value in the field for any ET is $FFFFFFFF_{16}$ then the current ET is to be disconnected and data is to be discarded.

If any other scenario occurred then a grant has been sent and not a control message.

This design does have a price that should be considered. An ONU does not know the exact number of bytes granted for each CoS, but only a threshold and the total amount of bytes granted for all CoSs. An ONU awards CoS1 before other CoSs since CoS1 is of the higher priority. Hence there can be a time when a newly arrived higher priority packet can pre-empt a lower priority packet out of its granted spot. Consider a case where only two priority queues exist (Figure 3.7). In this example the ET length specified that a total 440 bytes (220 of CoS1 + 220 of Cos 2) were granted. Furthermore "ET info" specified that up to TH3 should be granted for COS1. Between the request and a grant, a 70-byte packet arrived to CoS1; since it fits into TH3 it will still be granted. Therefore the packets that are transmitted are 290 bytes from CoS1 and 150 bytes from CoS2. The constant pre-emption of the lower priority packets is known as the "light-load penalty".

At high loads when the queues are full the OLT cannot grant all the data. Any data beyond the granted threshold remains in queue for transmission in the following cycle. If new data arrives after the grants are specified, it will wait in the queue until all data ahead of it has been granted. However, when the system is lightly loaded the OLT can grant all the data requested by the ET-CoS queue. If new high priority data arrives in the queue after the grants have been specified it might be able to fit under the specified threshold. Under that circumstance the higher priority packet will pre-empt lower priority data that was previously granted. Thus the "light-load penalty", as suggested by the name, can only happen at light loads. Our solution to the "light-load penalty" was to increase thresholds exponentially. By converting from the linearly increase to an exponential increase in thresholds we minimize the size between lower thresholds, the ones important at light loads. This change will reduce the chance that the packet, which arrived between the request and the grant, will fit between the queue's last packet and the threshold specified in the grant.

Figure 3.7 Light-load penalty example

An ONU can support up to eight ETs, and each ET can have up to eight CoSs; hence there are 64 ET-CoS pairs in single ONU. Even though we use four bits to specify the threshold instead of four bytes to specify the grant size, our system needs 64 of these grants versus four as previously required. With an additional eight "Grant Length" fields required and a further four bytes for the "Transmission Start time" field, the increase of overall gate message length to 74 bytes is inevitable.

Knowing what changes have been made to the standards we can now take a look at the scheduling architecture. The scheduling architecture uses modified gate messages to specify to an ONU which packets it must transmit on behalf of the ET. It also uses a modified report message to allow an ONU to request bandwidth on behalf of the ET.

CHAPTER 4 LOCATION-INDEPENDET PACKET SCHEDULING SCHEME

The OLT executes LIPS to determine how much bandwidth subtending ONUs must transmit on behalf of the ETs, and when. The scheme must also ensure that the maximum transmission cycle time would be bounded by a certain upper limit (*Tmax*). The key parts of our scheme are the scheduling architecture and the grant algorithm. In order to determine how much bandwidth our scheme can allocate we will first look at the regular transmission cycle.

4.1 Transmission Cycle

A typical transmission cycle begins with the execution LIPS in the OLT. After scheduling is completed, the OLT sends gate messages to the ONUs in sequence (Figure 4.1). A guard time is introduced between the transmission times of two

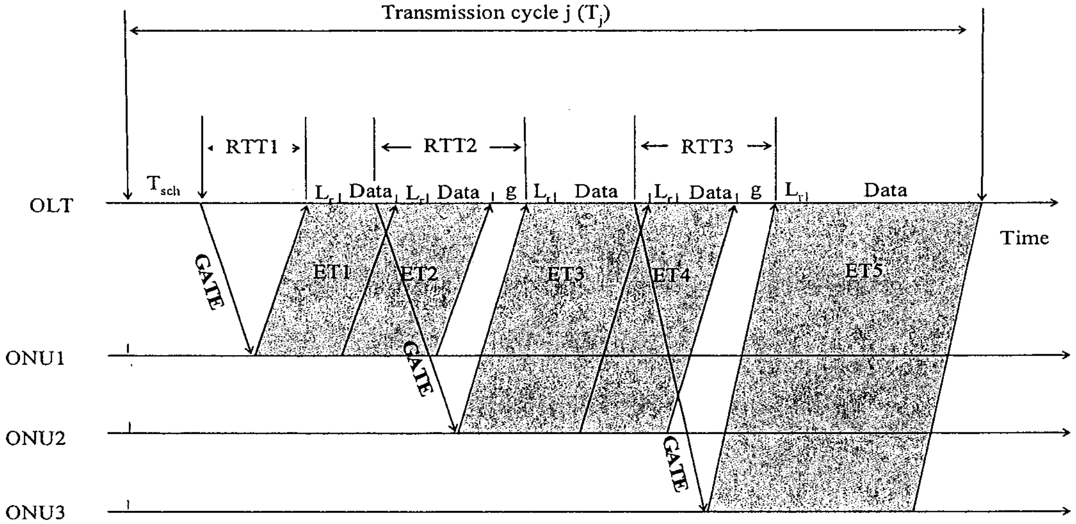


Figure 4.1 Illustrative example of transmission cycle

consecutive ONUs, for processing purposes and other optical related issues [6]. No guard time is needed between two ETs within an ONU because they are transmitted back-to-back. Every ONU must send a single modified request message to the OLT for *every* ET connected to it during its transmission slot. We are using the interleaved polling scheme to determine the gate transmission times[21]. Interleaved polling scheme allows the OLT to request the transmission from the next ONU before the full transmission from the previous one has arrived (Figure 4.1).

$$G_j^{(m+1)} = \max\begin{cases} G_j^{(m)} + RTT^{(m)} - RTT^{(m+1)} + \dfrac{\sum_{n=1}^{N}(L_{req} + D_j^{(mn)})}{C_p} + g \\ G_j^{(m)} + \dfrac{Lgrt}{C_p} \end{cases} \tag{4}$$

where $Gj^{(m)}$is the gate transmission time for ONU *m*; $RTT^{(m)}$ is the round–trip time to ONU *m*; C_p (bytes/s) is the capacity of the EPON trunk; L_{req} is the request message size of the ONU on behalf of the ET (65 bytes); and D_j^{mn} is the length data of ET *n* at ONU *m* and L_{grt} is the length of the grant message. The *jth* Gate message to ONU *m+1* is transmitted at such a point in time that the first bit from that ONU will arrive at the OLT with the guard time after the last bit from ONU *m*. This is described by the top line of (3). The bottom line says that the Gate to ONU *m+1* cannot be transmitted during the transmission time of the Gate to ONU *m*. The chance of this occurring is decreasing with every ET connected to ONU *m*. The length of transmission cycle *Tj* can be derived from Figure 4.1:

$$T_j = T_{sch} + RTT^{(1)} + \sum_{m=1}^{M}\sum_{n=1}^{N}(\frac{L_{req} + D_j^{(mn)}}{C_p}) + [(M-1)\times g] \tag{5}$$

The first two terms, scheduling time and round trip time to the first ONU, in the right side of (5) are the overheads associated with the current algorithm. The overhead can be minimized by selecting the ONU with the smallest *RTT* to be ONU1. However, this choice must not give rise to any penalty or discrimination against ETs connected to other ONUs to ensure that those ETs receive their fair share of the bandwidth. Therefore the process of allocating gates to the ONUs is independent from the order with which Gate messages are sent to them.

Using (4) we can state that the maximum number of data bytes ETs can transmit to the OLT during every transmission cycle is:

$$Bmax = [Tmax - Tsch - RTT^{(1)} - (M-1) * g] * Cp - Ltot \tag{6}$$

Where $Ltot = \sum_{m=1}^{M} \sum_{n=1}^{N} Lreq$ is the total number of requests in the cycle.

The function of the scheduling architecture in the OLT is to distribute B_{max} bytes among ETs in every transmission cycle.

4.2 Scheduling Architecture

Figure 4.2 shows the schematic diagram of the proposed Dynamic Bandwidth Allocation (DBA) system implemented in the OLT. The OLT maintains *K* credit pools, where *K* is the number of CoSs supported in the system. Each credit pool is used to enforce a long-term average rate of CoS traffic transmitted from all ETs, while still permitting short-term bursts above the allowed bandwidth. The credit pool for CoS *k* has two parameters: volume (V_k), and average rate of credit arrivals (R_k). The OLT also maintains *L* credit pools, each used to control the usage of the upstream bandwidth by an ET, where $L=M \times N$, *M* is the number of ONUs, *N* is number of ETs per ONU. The ET credit pool *l* has two parameters:

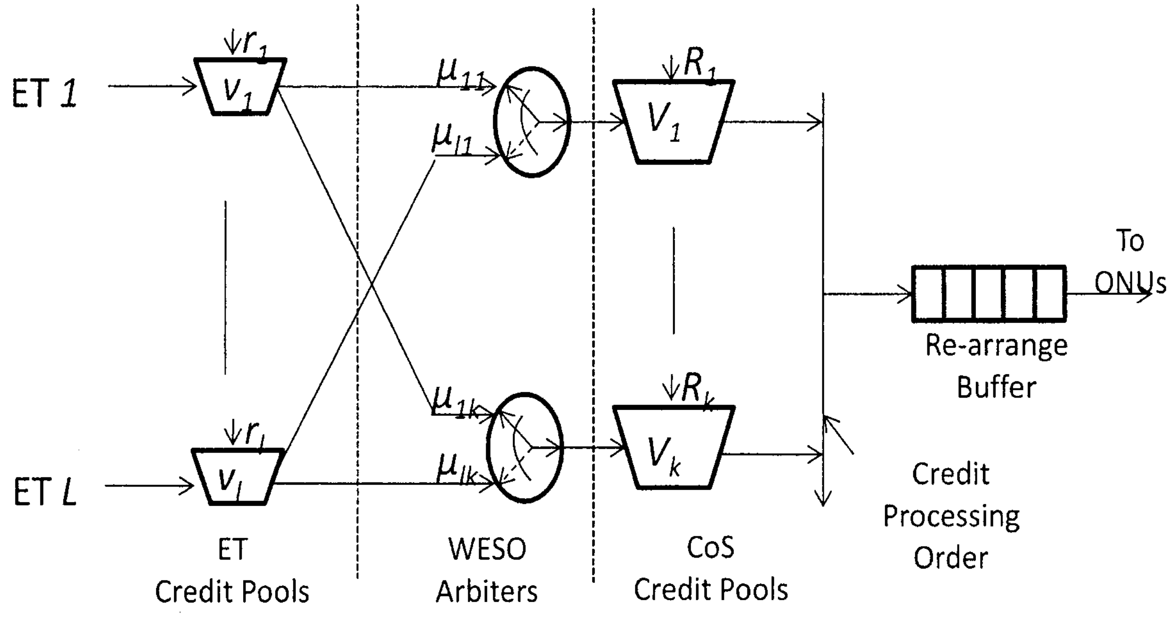


Figure 4.2 Bandwidth allocation systems

size (v_n), and average rate of credit arrivals (r_n). The v_n value determines the maximum number of bytes that ET n can transmit during any particular cycle.

At the beginning of every cycle, all the CoSs' and subtractor ETs' credit pools are initialized with their pre-configured V_k and v_n bytes for the upcoming cycle, respectively. A request from a given ET l for a given CoS k is honoured if there is sufficient credit in both the ET and CoS credit pools. A fair distribution of CoS k credits between ETs is established through the assignment of weights μ_{lk}. The number of CoS k credits (bytes) to be allocated to an ET l is determined proportionally according to its weight μ_{lk}. To enforce these weights, an arbitration mechanism Weighted Round Robin can be used, although in this system simulation we use *Weighted ET Scheduling Order* (WESO).

The WESO arbiter nl ensures that the contending ETs for CoS k will receive a fair share of the CoS credit pool, based on their weights. The arbiter uses the number of CoS k bytes scheduled so far for every ET l (b_{lk}) and the total number of CoS k bytes scheduled so far (B_k) in order to obtain a relative priority with which new requests from ETs must be processed. Ideally, the ratio of b_{lk} to B_k should be μ_{lk}. The goal of the WESO arbiter k is to achieve this optimum ratio for every ET. Consider the case where ET x and ET y are both requesting bandwidth:

$$d_k[x] = \frac{\left(\frac{b_{xk}}{B_k}\right) - \mu_{xk}}{\mu_{xk}} \leq d_k[y] = \frac{\left(\frac{b_{yk}}{B_k}\right) - \mu_{yk}}{\mu_{yk}} \tag{7}$$

If expression (7) is true then ET x will be considered for CoS k credits before ET y. The advantage of the WESO arbitration mechanism over the previously mentioned Weighted Round Robin is that it can be executed offline (i.e, during the previous cycle) in order to produce the ET orders for the current cycle.

By assigning the weights to the ETs we are taking ONUs out of the bandwidth negotiation process. It guarantees fair bandwidth usage of the ETs. It also minimizes the delay in case of ET handover, since no recalculation of the weights is needed.

The re-arranger buffer stores the credits calculated for all ETs during the scheduling time. Each entry of the re-arranger buffer contains four information elements: ONU I.D., ET I.D., CoS I.D., and the size of the credits given to that specific CoS in that ET. At the end of every scheduling time, the total size of credits stored in the re-arranger buffer will not be more than B_{max}. After the scheduling is completed, the OLT will rearrange the credits so that all ETs of the same ONU are grouped together. Starting from ONU 1, the OLT will look through all the ETs that belong to that ONU and place them first, then will move to ONU 2; then sequentially all the way to ONU M. Then the re-arranger generates modified gate messages and sends them to ONUs.

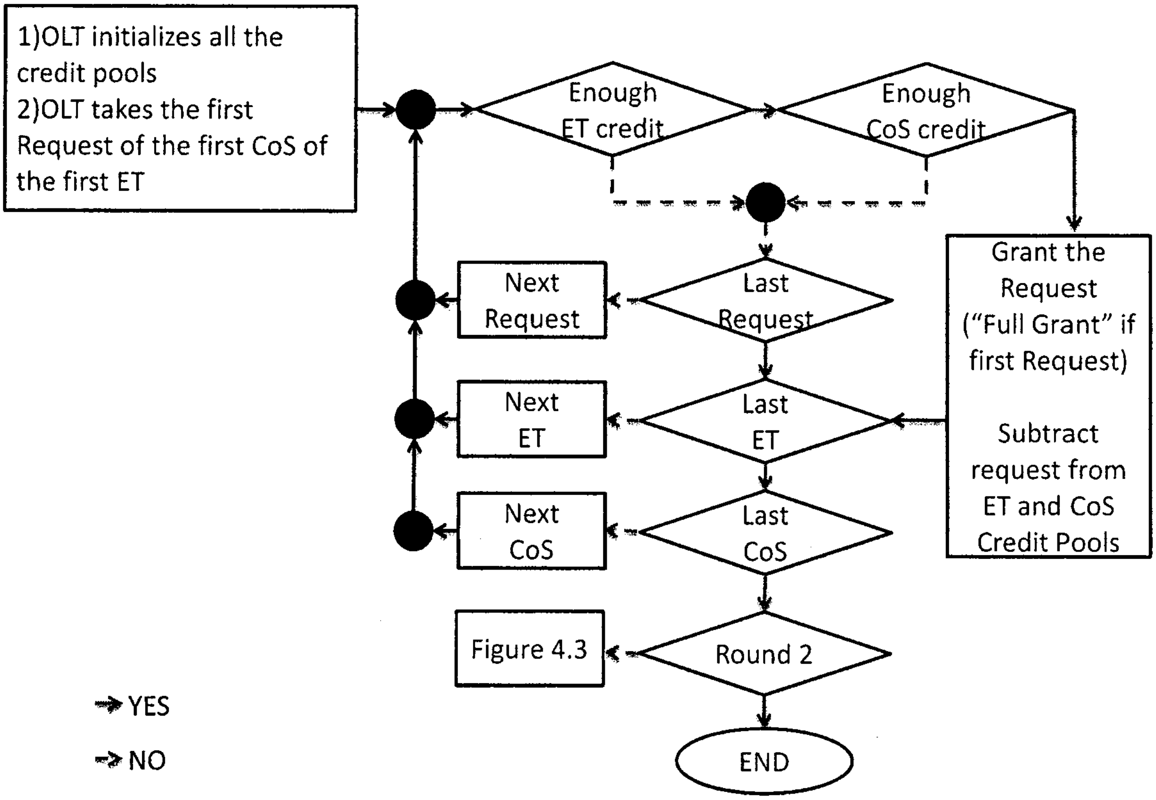


Figure 4.3 Round 1 flow chart of the Grant Algorithm

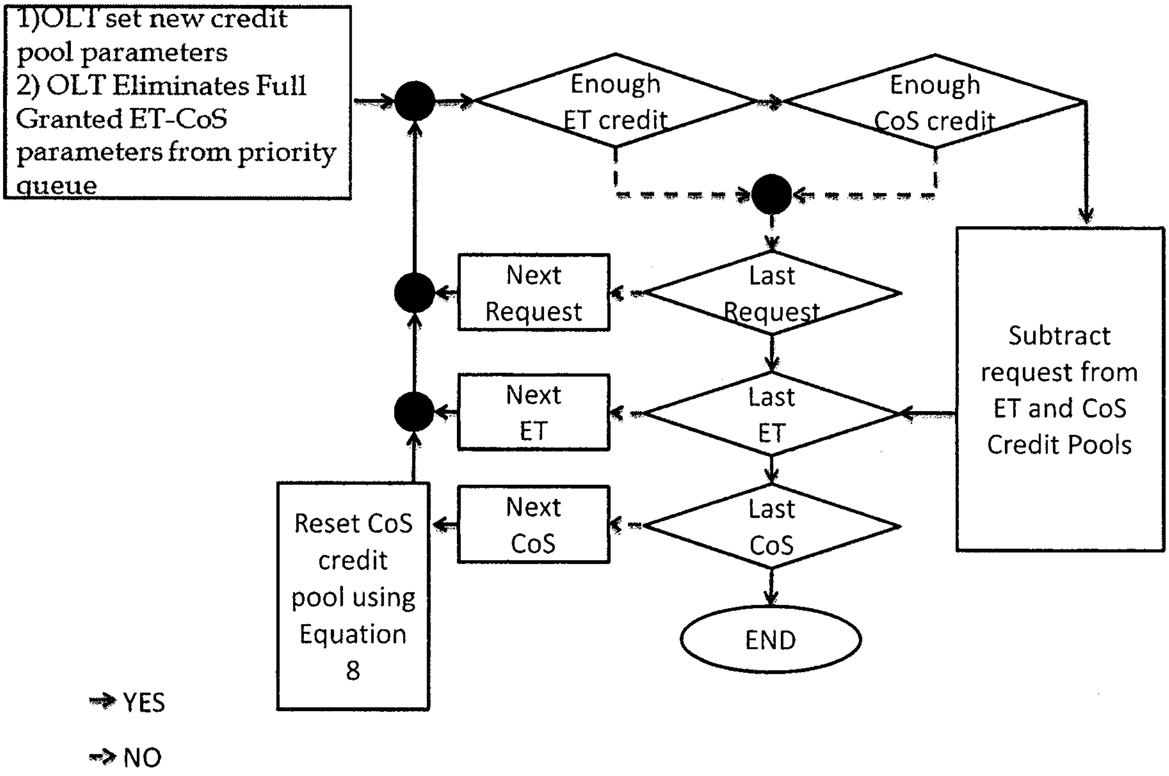


Figure 4.4 Round 2 flow chart of the Grant Algorithm

4.3 Grant Algorithm

The two rounds in the algorithm depicted on Figure 4.3 are used to calculate grants for each ET. The first round gives rise to the distribution of most of the available (Bmax) grants between ETs and is mandatory. At the end of this round, some amount of grants (credits) may still remain in the credit pools. The second round is designed to redistribute the remaining grants between ETs.

During round one the OLT will first analyze the CoS1 grants for all ETs. This implies that the OLT takes the first ET stored in the priority database, and examines the first Request message received from this ET in the previous cycle. The Request message can include 0, 1, or at most 13 queue reports for CoS, sorted in descending order of requested grant size. The OLT chooses the first report and will grant this request if there are enough credits in the CoS and ET credit pools. If neither one of the pools has enough credits, the OLT will consider the next queue report, the next highest grant size, for the current CoS until OLT can suffice with the grant. If a match is found for the first request full grant is sent, otherwise no grant or partial grant is sent. Whenever the OLT issues a grant it subtracts granted bytes from both the credit pools. The OLT then calculates a grant for the next ET in the database using the above algorithm. After all grants for CoS1 are considered, the OLT moves to CoS2 then to CoS3 then sequentially to CoS k. Round one ends when the entire CoS and ET grants are considered.

During round two, the OLT first examines the total number of credits in the re-arranger (B_{tot}). If B_{tot} is less than B_{max}, it means that there are credits still remaining in the CoS credit pools. These unused credits can be recollected and distributed between ETs that have received partial or no grant in round one. The OLT distributes these unused credits between the CoSs in proportion to their rates (R_k):

$$V_k = \begin{cases} \frac{R_k}{Rtot}(Bmax - Btot) & for\, k = 1 \\ \frac{Rk}{Rtot}(Bmax - Btot) + v_{k-1} & for\, 1 < k \leq K \end{cases} \tag{8}$$

The first term in the right side of (8) represents the fair share of the remaining credits (from round one) that will be given to each CoS in the second round. While CoS1 gets its fair share only, a lower-priority CoS *k* (*k* > 1) gets its fair share plus extra (V_{k-1}) credits. This is the total amount of credits that will remain unused in CoS *k*-1's credit pool when the processing of grants for that CoS is completed in the second round. The OLT executes the same algorithm as in round one, but this time it only considers the ETs that have not received a full grant. Like in round one, the CoS1 grants are processed first, followed by other CoSs in sequence. Once the processing of grants for CoS1 is completed, the OLT knows exactly how many credits are still remaining in the CoS1 credit pool. The OLT will then configure CoS2's credit pool with its fair share of the remaining bandwidth plus the left-over bandwidth from CoS1. In general, the exact value for CoS *k* is known once the processing of grants for CoS *k*-1 is completed.

CHAPTER 5 SIMULATION ANALYSIS

5.1 Program Information

An in-house simulation program (in C++) has been developed to test the proposed bandwidth allocation system. It will also help to compare the proposed scheme to the previously tested COPS and IPACT schemes [22]. The simulation has been carried out to measure the performance metrics such as packet delay, packet loss and throughput of the upstream bandwidth. Data is collected by tracing the packets through the system. A typical packet is generated in the ET and time-stamped. Then it is transferred to the ONU WRED buffer. If lost due to the buffer being full, the packet loss counter is incremented. Otherwise ONU generates a report message where the information for this packet is included. Following this process the packet waits in the queue for the grant from the OLT and subsequent transmission. At the arrival of the packet to the OLT the current time and the packet timestamp are compared to measure the packet delay in the network. Each simulation is run for at least 100,000 seconds of simulation time which is roughly 24 hours. Each packet delay and loss is stored, and after the simulation is complete, these are averaged out. The throughput of the system is calculated through measuring total number of bytes reaching the OLT and dividing it by the total time elapsed.

This C++ program is an event-based simulation. Everything that can happen in the system falls under specified types of events. Each type of event has a specific process associated with it. Hence first the system looks at what time the event occurred, how to respond to it and what event should be scheduled as a response. Knowing how long each process takes is a key, as it will determine when to schedule the next event. After all actions corresponding to this event are completed the system looks at the next event in the list.

Throughout all simulations 16 ONUs have been assumed to be at a randomly chosen distance between 0.5 and 20 km, and have six ETs connected to each of them. The EPON trunk capacity is set to 1000 Mbps. The size of the report message is set to 65 bytes and the gate message is set to 74 bytes as mentioned in Chapter 2. The guard time described in Chapter 3 is 5 μs. The maximum transmission (*Tmax*) cycle is set to 2 ms. The values set forth above are reasonable choices, as already reported in many other studies [21][22][24][25]. In particular, the above value for *Tmax* is chosen to meet the delay requirements of voice traffic in access networks [21][22][25]. Also both rounds of the algorithm from Section 4.3 are implemented. Scheduling time at the OLT is set to 25 μs.

In the simulations, we have classified services into three classes: CoS1, CoS2, and CoS3. These CoSs share a common buffer space of 2 Mbytes at an ONU for each ET. CoS1 represents low loss and delay-sensitive Expedited Forwarding service, which typically provides for Constant Bit Rate (CBR) voice. CoS2 represents Assured Forwarding service, which provides for non-delay-sensitive but bandwidth-guaranteed service, usually video/data. CoS3 represents Best-Effort service, which requires no bandwidth commitment from the network [21]. At each ET the traffic for CoS1 is simulated as a Constant Bit Rate (CBR) with packet rate is 1334 packets/s at 70 bytes per packet. We used an ON–OFF source to generate traffic for CoS2 and CoS3. The two sources have identical parameters. For each source, the ON and OFF (silent) intervals are drawn according to a *pareto* distribution, which has been widely used to model self-similar traffic in the Internet [26]. The pareto distribution is characterized by a shape parameter (a) and a location parameter (b).

$$F_{(x)} = 1 - \left(\frac{b}{x}\right)^{a} \tag{9}$$

This distribution has a finite mean and infinite variance when $1 < a < 2$ [27]. In our simulations, the shape parameters for the ON and OFF intervals are set to 1.4 and 1.2, respectively. These values were used to simulate Ethernet traffic in the previous studies [21][22]. b is a function of the shape parameter and the mean of the distribution. The mean time of the ON state is set to 50 ms. The mean time of the OFF state is a function of the ET's offered load. The average arrival rate of each ON-OFF source varies between 0.46 and 7.96 Mbps to achieve the desired offered load at the ETs. The length of packets generated during an ON state follows the tri-modal distribution [22]. These three modes correspond to the most frequent packet sizes of 64, 594, and 1518 bytes observed in backbone and cable networks. In our simulations, each of these packets is generated with a frequency of 62%, 10%, and 28%, respectively. These are based on the measurements taken for these packets in cable network head-ends [29].

We made the following assumptions regarding the simulation parameters. The CBR packets must always conform to the specified profile (R1, V1) for the CoS1 credit pool. This means that the value for V1 must be sufficiently large to accommodate all requests received for CoS1 grants. We obtain this value for V1 as

$$V_1 = M \times N \times \lceil r_{cbr} \times T_{max} \rceil \times (L_{cbr} + L_{IFG} + L_{preamble}) \tag{10}$$

Where:

M is the number of ONUs, *N* is the number of ETs per ONU, $\lceil r_{cbr} * T_{max} \rceil$ is the number of CBR packets arriving at each ET during a maximum cycle time, and $(L_{cbr} + L_{IFG} + L_{preable})$ is the size of a CBR packet (including IFG and preamble). The value for V1 will then be 23052 bytes, whereas that of *R1* = *V1/Tmax* will be 92.2068 Mbps. Since, in this work, CoS3 serves best-effort data, no bandwidth has been explicitly reserved for it *(V3=0)*. Thus, all the remaining bandwidth is allotted to CoS2. The best-effort packets are still able to make use of the unused bandwidth in the second round of the algorithm.

All ONUs are configured with the same queue thresholds. The lowest threshold is set to the minimum packet size of 84 bytes (including IFG and preamble) allowed in the network. The highest threshold (#13) is set to V_m bytes, where $V_m = B_{max} / (M \times N)$. Thresholds h $(1 < h < 13)$ are spaced in exponentially increasing distances, to minimize the light load penalty, as described in Chapter 2:

$$TH(h) = 84 \times e^{\frac{\ln Vm - \ln 84}{12} \times (h-1)}$$

Recall from Chapter 1 that the WRED operates based on four inputs: Weight parameter, minimum threshold, maximum threshold and maximum drop probability. For assured forwarding and best effort traffic, Cisco's default values for q_{min} in their commercial products are $\frac{13}{18}$ and $\frac{9}{18}$ of the buffer size, respectively. Cisco's default maximum drop probability is set be ten percent, and the weight parameter is set to $\frac{1}{512}$ [30]. WRED parameters in the simulations of COPS and IPACT are similar to the default WRED parameters used by Cisco [21]. Hence our WRED parameters (Q_{min}, Q_{max}, μ, P_d) are set to same value as in previous research for comparison purposes. For CoS1 the parameters are set to (F,F,1,0), for CoS2 (0.75F, F, 0.002,0.1) and for CoS3 (0.4F, 0.9F, 0.002, 0.1) where q='F' indicates the full buffer length in bytes. The Q_{min} and Q_{max} values of CoS1 are set to the total buffer space in order to ensure that the CoS1 packets will not be discarded by the WRED algorithm.

5.2 Fixed ETS Simulation

5.2.1 Simulation Setup

In order to be able to compare our results with the results in [22], we had to ensure that the ETs are all stationary. In this simulation we have all ETs connected to each ONU with handover probability set to zero. All ETs have identical traffic parameters and offered loads. The offered loads at each ET are varied from 0.016 to 0.166 in different simulations; this makes the ONU's offered load range from 0.1 to 1.0 respectively. We assumed that all ETs have an equal share of the EPON resources. For this reason, the spatial weights of each WESO arbiter are set to the same value ($\mu_{nk} = 1/MxN$). Also, we give the same profile to all ET credit pools.

5.2.2 Simulation Results and analysis

5.2.2.1 Delay

We start our first numerical analysis by investigating the delay performance of LIPS scheme and comparing it to IPACT and COPS schemes. The average and maximum packet delay as a function of the ONU offered load are shown in Figs. 5.1 through 5.4 for each of these schemes. The packet delay is defined to be the time that a packet spends in the back-end EPON from the instant that it arrives at an ONU until it is delivered to the OLT.

With IPACT, the CoS1 (shown as C1) average delay reaches 1 ms when the load is heavy, while with LIPS the delay reaches 3 ms (Figure 5.1). IPACT is superior in the CoS1 statistics but inferior in CoS2 (shown as C2) where LIPS is just over 100 ms delay, and IPACT is close to 1000 ms. As expected, COPS and LIPS schemes perform almost identical in terms of average delay, since both are based on the leaky bucket design (Figure 5.2). The CoS1 3 ms delay for COPS and LIPS can be explained as follows: when arrived, the packet has to wait on average half a cycle to be polled and then exactly one cycle to receive a grant. Hence 1.5 times cycle time would give an average of 3 ms delay. The 100 ms average delay for CoS2 can also be mathematically supported: recall that the average ON time for CoS2 is 50 ms, and the data rate at each ET is 16.7 Mbps. Thus the total number of bytes generated by the source is 104166 bytes. The amount of cycles needed to transmit this data to the OLT is found with:

$$\frac{104166}{\frac{B_{max}}{M \times N} - \frac{V1}{M \times N}} = \frac{104166}{2375 - 240.124} = 48.72\ cycles$$

Where $\frac{B_{max}}{M \times N}$ is the maximum number of bytes the ET can transmit per cycle and $\frac{V1}{M \times N}$ is number of bytes used by CoS1. 48.72 cycles is equivalent to 97.44 ms, which is close to the values we measured.

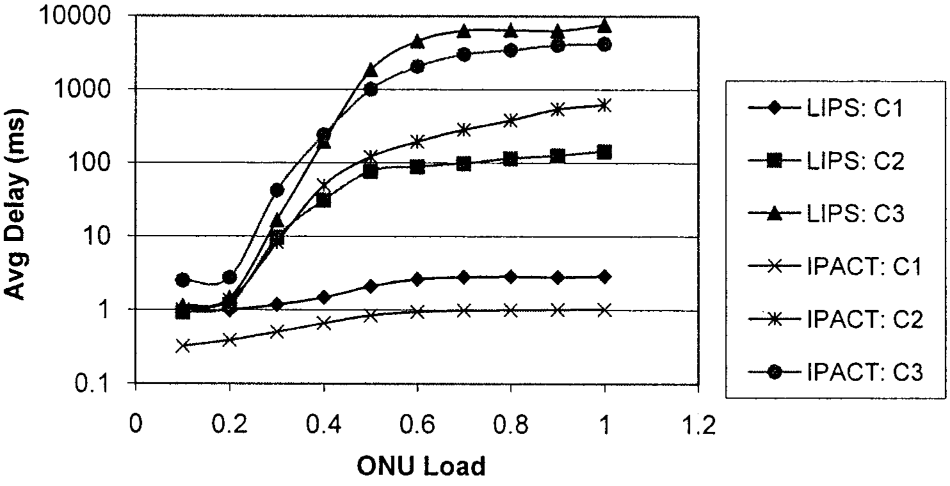


Figure 5.1 IPACT and LIPS average packet delay

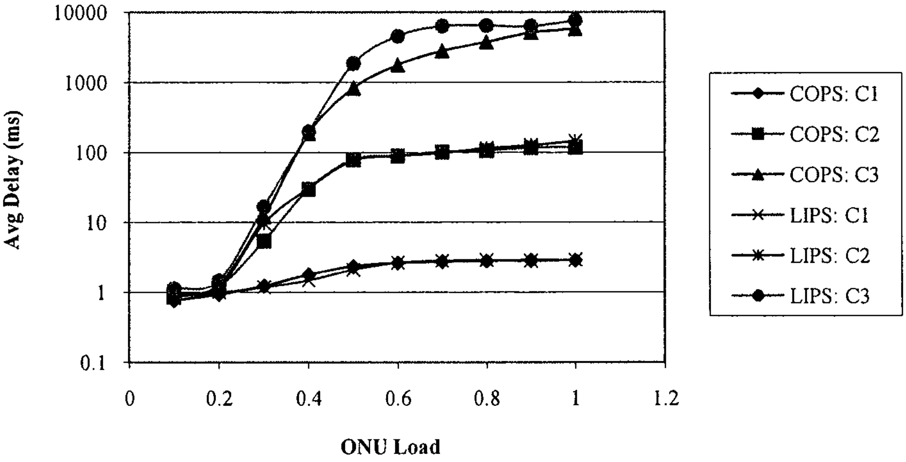


Figure 5.2 COPS and LIPS average packet delay.

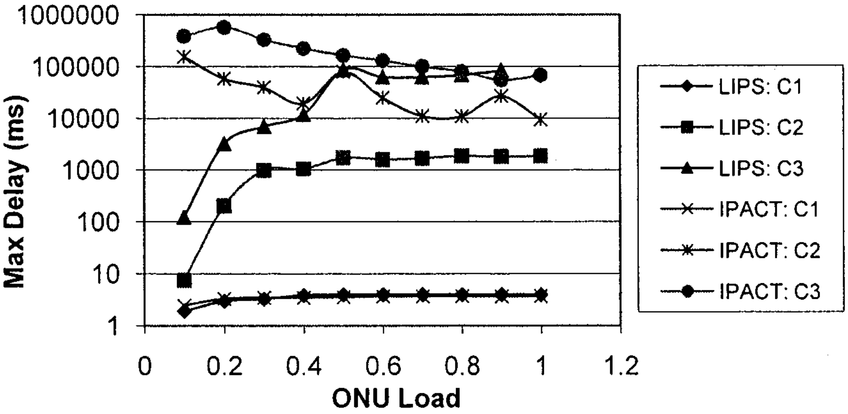


Figure 5.3 IPACT and LIPS maximum packet delay

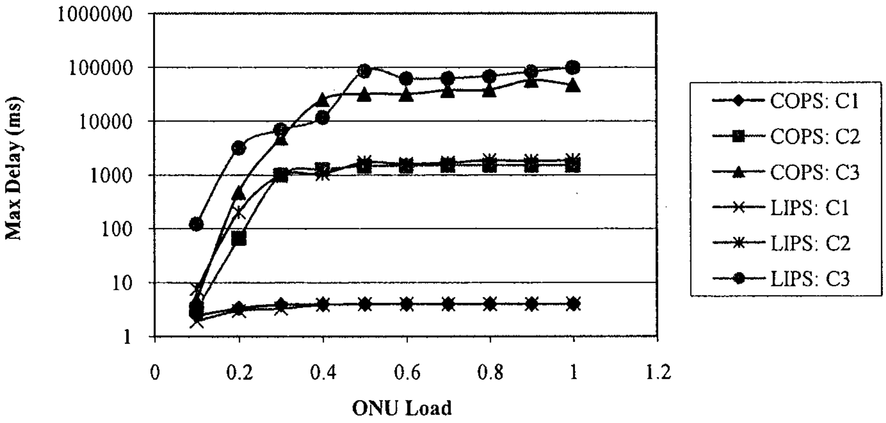


Figure 5.4 COPS and LIPS maximum packet delay

In Figures 5.3 and 5.4 we can see the maximum packet delay comparison between the LIPS, IPACT and COPS schemes. CoS 1 maximum delays for all three schemes is almost identical and it settles around 4 ms at higher loads. Maximum delay of 4ms is actually the twice the cycle time.

Figure 5.3 also shows that, in the IPACT scheme, the delays of CoS2 and CoS3 are significantly higher than the corresponding delays in the LIPS scheme. The previously discussed light load penalty can be clearly seen at loads from 0.1 to 0.2 for IPACT where the delay reaches over 100000 ms. The difference between IPACT and LIPS can be easily explained. The LIPS pre-emption is limited by the thresholds described in Section 3.2.2 while in IPACT has no such limit. Hence LIPS is not affected by the light-load penalty to the same magnitude as IPACT.

From Figure 5.4 what is the first to be noted is the maximum delays similarities of COPS and LIPS. CoS2 maximum delay of LIPS is identical to the one of COPS for CoS2 and is equal to 1500ms. The maximum CoS2 delay at high loads has a strong correlation to the buffer size. At our current 2 Mbyte buffer when the buffer is full the packet has to wait in the buffer before being transmitted. The number of cycles can be calculated through:

$$\frac{F}{\frac{B_{max}}{M \times N} - \frac{V1}{M \times N}} = \frac{2000000}{2375 - 240.125} = 936 \text{ cycles}$$

Where $\frac{B_{max}}{M \times N}$ is the maximum number of bytes an ET can transmit per cycle, $\frac{V1}{M \times N}$ is number of bytes used by CoS1, and F is the buffer size. 936 cycles is equivalent to 1876 ms of delay; this is close to what we observed in the simulation. Although not as severe as IPACT, LIPS is still susceptible to the packet pre-emption at higher CoSs. Due to this susceptibility CoS3 maximum delay for LIPS is always greater than that of COPS.

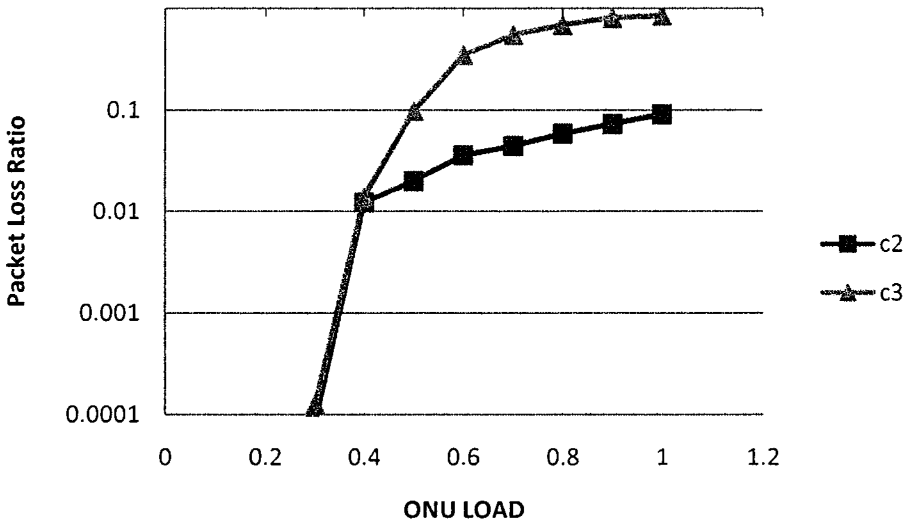


Figure 5.5 LIPS packet loss ratio

5.2.2.2 LIPS Packet Loss Ratio

Figure 5.5 shows the average packet loss ratio for CoS1, CoS2 and CoS3 as a function of ONU load. The CoS1 packet loss ratio remained 0 throughout the simulation. The biggest change in packet loss in the system is noted at 0.4 - 0.6 load. At this load the number of packets generated is much greater than the number of packets transmitted; thus the buffer is getting full on a regular basis. As we can see at high loads the packet loss remains 0% for CoS1, approaches 10% for CoS2, and approaches 85% for CoS3.

5.2.2.3 LIPS Throughput

Throughput is one of the major concerns we had with our system. The throughput measured is defined as ET data reaching OLT divided by time of the simulation. Although the values in Figure 5.6 seem small for an EPON system with a trunk capacity of 1000 Mbps, this is justifiable considering that six ETs per ONU with 16 ONUs implies 96 units using this bandwidth.

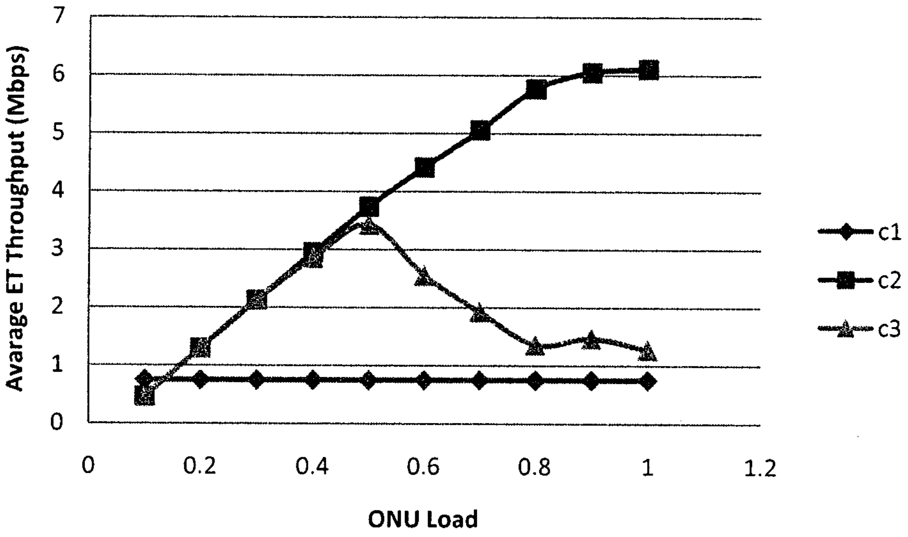


Figure 5.6 LIPS average ET throughput

Hence as an example the actual use of the trunk at 0.7 ONU load is:

$$[0.74(CoS1) + 5.05(CoS2) + 1.92(CoS3)] * 96 \approx 750\ Mbps$$

CoS1 throughput shows exactly what is expected of the CBR traffic. Since there was no change in CoS1 traffic the output remained constant during every load of a simulation. At a load of approximately 0.5, the throughputs of CoS2 and CoS3 imply more evidence that the buffer is filling up. After this critical point CoS3 has a smaller chance of getting through the access buffer, causing a constant deterioration of CoS3 throughput as the load increases. It is also important to note that CoS2 throughput is not on a constant increase. The throughput's slope ceases to be constant at moderately high loads, and reaches a steady state only at very high load.

5.2.3 *Simulation Conclusion*

In the simulations shown we compared LIPS to the previously existing and well-known IPACT and COPS. With the delay results established we can conclude that our scheme is behaving closer to COPS, although it does suffer from packet pre-emption at higher CoSs commonly seen in IPACT. Our scheme suffers no loss in CoS1 throughput and the loss of CoS2 approaching 10%.

We do believe that the choice of 2 Mbytes for the buffer size plays one of the major roles in the result of delay as well as the packet loss. The moment the buffer is full, CoS2 achieves its steady-state maximum delay. Moreover buffer size is also responsible for the throughput of CoS3 traffic, since buffer size is inversely proportional to packet loss of best-effort CoS. Although the throughput and the packet loss study on the other schemes were not done, all the stationary LIPS results provide a solid comparison base for our further simulations.

5.3 Roaming ETs Simulations

The reason for these simulations is simply to prove that as designed LIPS behaves identically in both the wired and wireless cases. In these simulations ETs would be randomly moving from one ONU to another at low and high speeds.

5.3.*1 Simulations Setup and Assumptions*

We ran two sets of simulation where the number of ONUs and ETs remained the same as in the first simulation: 16 ONUs, and 6 ETs per ONU. They can roam between different ONU domains randomly. The time an ET spends at each ONU is varied from 1500s to 15000s to simulate slow moving ETs scenario. The time frame was used to mimic the environment where a person would be rarely moving from one ONU wireless area to another. Example is a laptop use in the campus layout. 15s to 1500s to simulate fast moving ET scenario, this time frame was used to simulate users that use their Iphones and PDAs, thus moving regularly around the WEPON network. In light of the fact that the load of each ONU can now vary depending on how many ETs are connected to it, we decided to do the measurements based on the average ONU load. For example if the offered load at each ET is 0.016, with a total of 96 ETs and 16 ONUs the average ONU load is 0.1. All traffic generation for all CoSs remains the same as in simulation 5.2. The spatial weights of each WESO arbiter are set to the same value ($\mu_{nk} = 1/M \times N$).

5.3.2 *Simulation Result and Analysis*

5.3.2.1 Delay

The first things we will look at are the average and maximum delay of the roaming ET simulation. We compare them to the stationary ET simulation (Figures 5.7 – 5.10). As shown in Figure 5.7 the average delay of the slow roaming ETs and stationary ETs are identical. The handover occurrence in the slow roaming ETs scenario is so small that it has a negligible effect on the average delay. The comparison between the fast roaming ET and the stationary ET scenarios are shown in Figure 5.8. On one hand CoS1 and CoS2 results for both scenarios are analogous. On the other hand the CoS3 packet delay in the fast roaming scenario is slightly smaller. This can be explained through buffer clearing. When an ET is moving from one ONU to another all the packets in the first ONU are cleared. Since packets are being discarded, their delay is not added to the statistics. Moreover when the ET roams, it starts with an empty buffer at the new ONU, giving a chance for the CoS3 packet to be stored and granted. Hence it will further minimize CoS3 average delay.

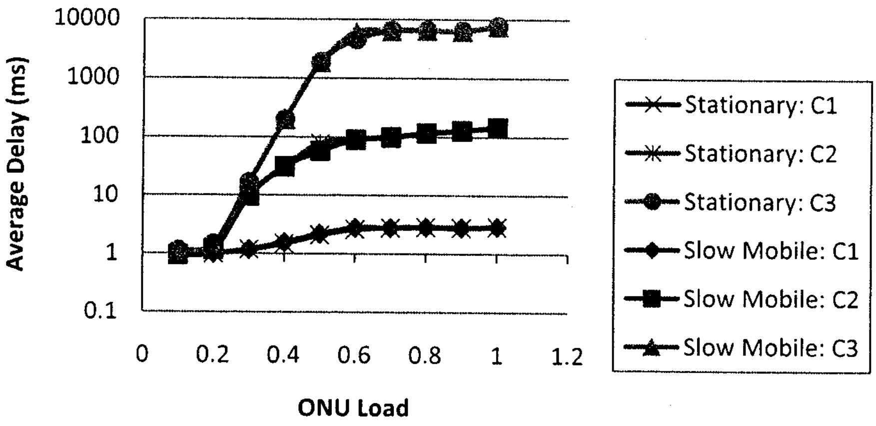


Figure 5.7 LIPS Average packet delay for stationary and slow roaming ETs

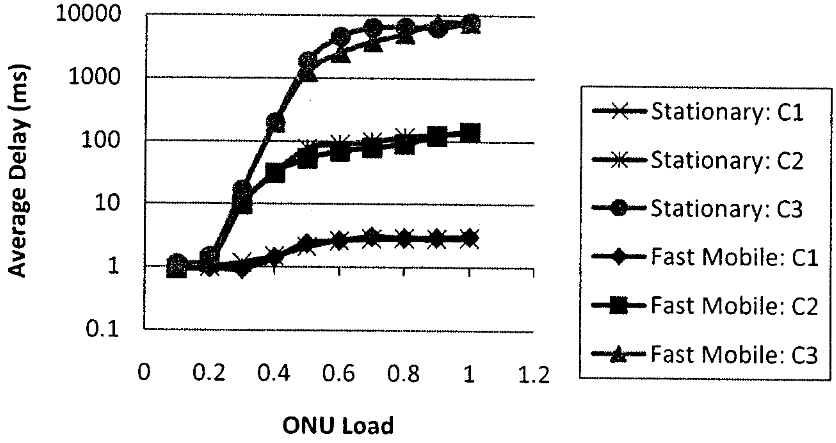


Figure 5.8 LIPS Average packet delay for stationary and fast roaming ETs

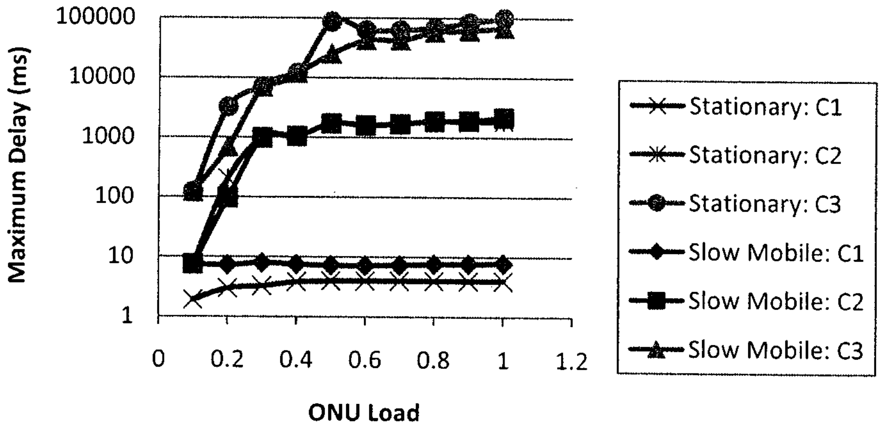


Figure 5.9 Maximum packet delay for stationary and slow roaming ETs

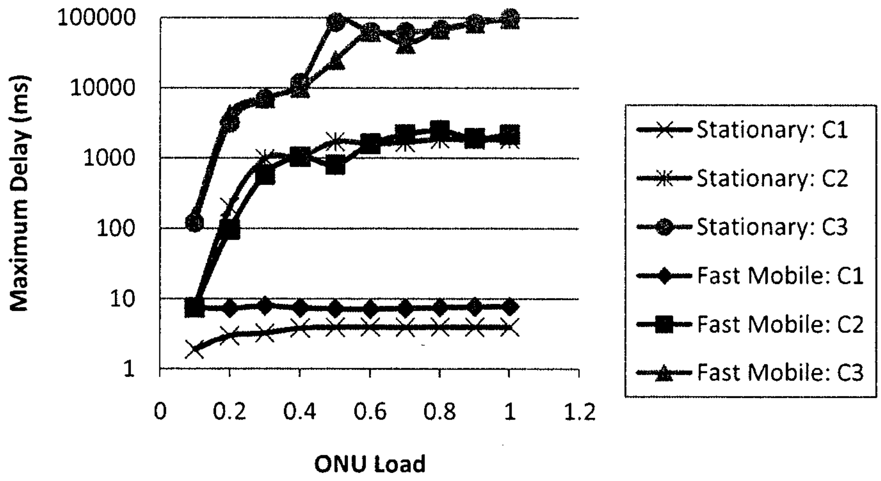


Figure 5.10 Maximum packet delay for stationary and slow roaming ETs

What instantly comes to attention in Figures 5.9 and 5.10 is the increase of our CoS1 maximum delay to 7.8ms the moment the ETs are mobile. By intuition 7.8 ms is approximately a value of 4**Tmax*. In order to explain this occurrence we first have to recall the three way handshake of the ET handover. When an ET arrives to the new ONU, the following communication will take place in the next few cycles (Figure 2.4): In the first cycle the ONU notifies the OLT of a newly arrived ET. In the second cycle the OLT sends the extra zero grant to find the ET's MAC address. In the third cycle the OLT sends a second zero grant to determine the status of the ET. After the handshake (on the forth cycle) the OLT sends a Gate message for regular transmission to occur. After carefully examining the simulation we found that such a delay only occurred when a packet is generated right after the ET roamed. Therefore the packet has to wait close to four cycles to be transmitted. Overall from Figures 5.9 and 5.10 we conclude that ET roaming does not affect the maximum packet delay for CoS2 and CoS3 *in our scenarios*. However, we do note that ET roaming can affect packet maximum delay performance. This can happen if the ETs roam between different ONUs at a greater rate than in our fast moving scenario.

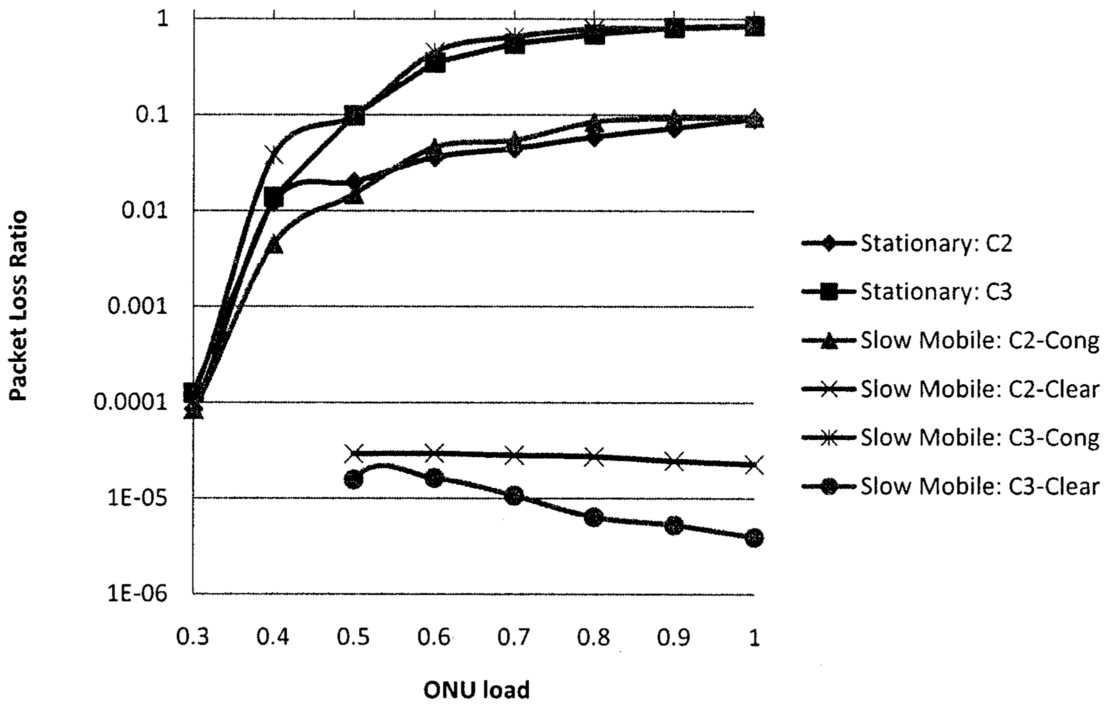


Figure 5.11 LIPS packet loss ratio for stationary and slow roaming ETs

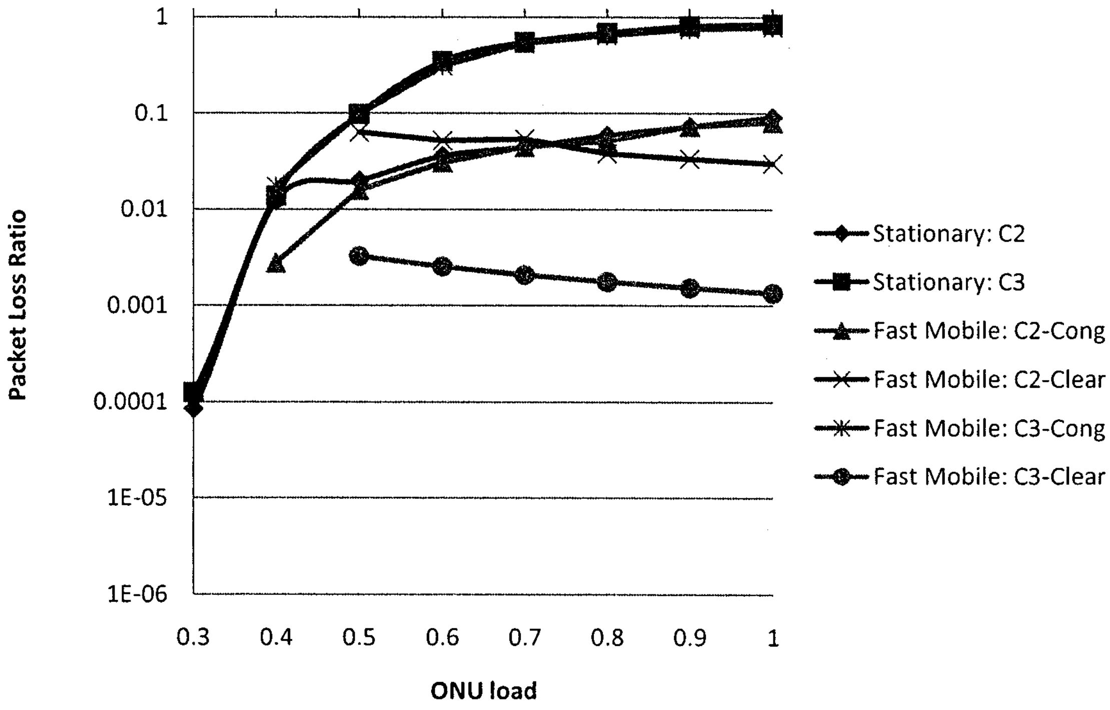


Figure 5.12 LIPS packet loss ratio for stationary and fast roaming ETs

5.3.2.2 Packet Loss Ratio

Figure 5.11 and 5.12 depicts a packet loss ratio for slow and fast roaming ETs and compares them to previously shown stationary ETs. There are two possible packet losses in the system. First one is due to the congestion and referred to as C2-Cong and C3-Cong on the figures. The second one is due to buffer clearing and referred to as C2-Clear and C3-Clear. CoS1 is not shown on any of the two figures since CoS1 packet losses have not been experienced in any of the simulations. Stationary ETs do not suffer from the ET handover packet loss, hence the only packet loss shown for stationary ETs are due to congestion (Stationary: C2 and Stationary: C3).

The packet loss due to ET handover is just a fraction of the losses due to congestion in the slow moving ETs scenario (Figure 5.11). While congestion losses remained close to the ones of the stationary ETs scenario. In the fast roaming ETs scenario reflected on Figure 5.12 the results are completely different. At higher loads the WRED buffer gives priority to CoS2 rather than CoS3 packets; hence CoS2 packet will suffer the most from buffer clearing due to handover. This is supported by our simulation since CoS2 handover loss is greater than the loss due to congestion at 0.5 through 0.7 loads. CoS3 on the other hand is not affected by the handover to the same degree. Handover packet loss for CoS3 remains only a fraction of the one due to the congestion. The other interesting fact is that the CoS2 and CoS3 handover packet loss ratio decreases as the average load increases. This is because the ET handover is not affected by load. At higher loads, the buffers reach their capacity on a regular basis. Thus during the handover the ET loses a fixed amount of data (2bytes). The only difference between the loads of 0.6 and 0.9 is that more packets are generated in the latter. Therefore the handover packet loss ratio for a load at 0.6 is higher than at 0.9. Overall the effect of the ET handover on the packet loss ratio increases as the rate of the ET roaming increases.

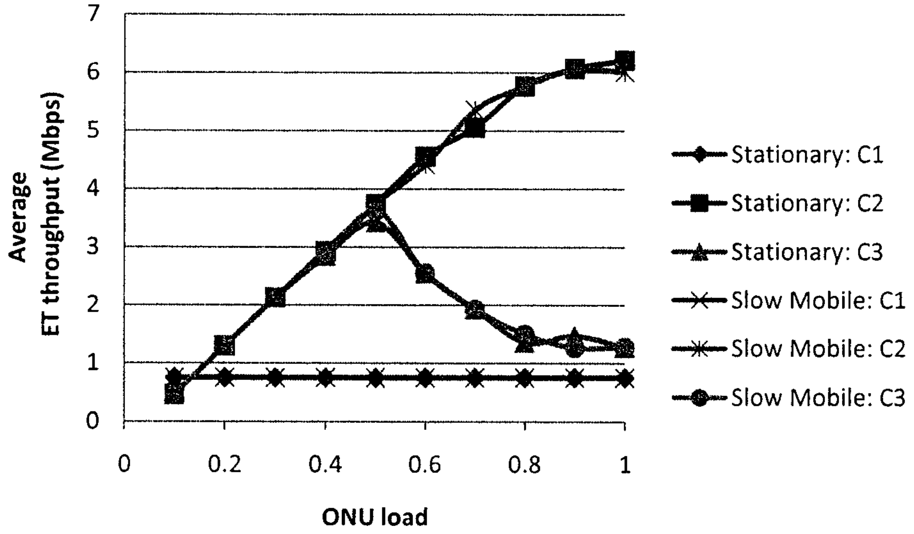


Figure 5.13 LIPS throughput with stationary and slow roaming ETs

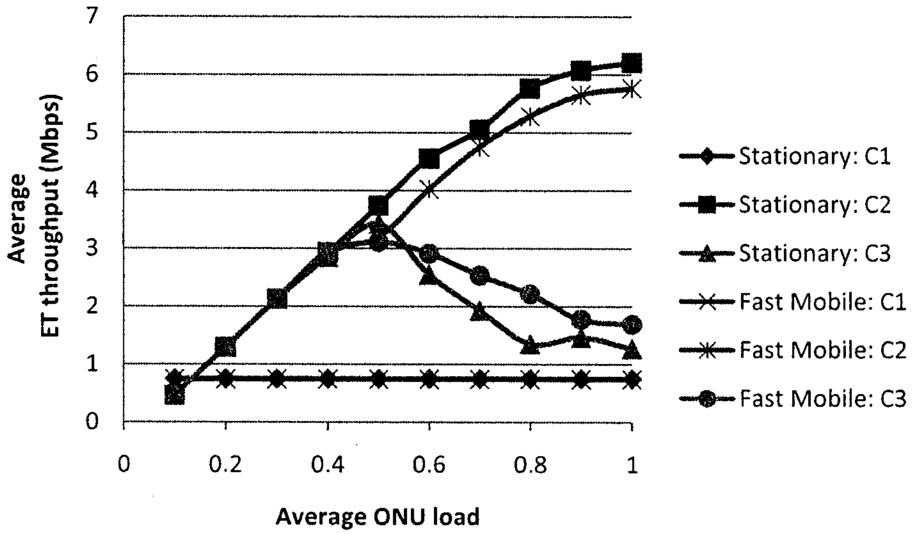


Figure 5.14 LIPS throughput with stationary and fast roaming ETs

5.3.2.3 Throughput

Overall throughput changes between the slow roaming ETs and Stationary ETs are so small, that the comparison between the two scenarios is trifling. (Figure 5.13) With the fast roaming ETs the changes to CoS2 and CoS3 are irrefutable. The CoS2 throughput decreases as the speed of roaming increases. This can be explained through additional packet losses due to handover. After the handover, an

ET is assigned to an empty buffer of the new ONU. With no CoS2 packets yet in the buffer, CoS3 packets have a higher probability to be granted by the OLT. Therefore CoS3 undergoes a slight increase in throughput as the load increases. CoS1 remains unchanged throughout the simulations due to receiving the transmission grants without any loss.

5.3.3 Simulation Conclusion

Since the interval between ET handover is marginally higher than the packet generation, the simulation results show that the system behaves closely to the stationary ET scenario. The major visible change in the system is the increase of the CoS1 maximum delay to 8 ms and the increase in packet loss due to ET handover for CoS2 and CoS3 traffic as the roaming speed increases.

5.4 Different Priority ETs

The goal of this simulation is to prove that the system in fact can control the throughput of different ETs based on their weights. The stationary ETs scenario is used in this simulation for priority tracking purposes. As opposed to previous designs, emphasis is placed on the ability to bring a variety of service levels to the user. One of the ways to try to bring variety is to have different priority users. For example in a campus environment: the highest priority can be professors – they need the best connection; medium priority would be graduate students – for some of their research that would require a faster connection, and the lowest priority would be granted to undergraduates.

5.4.1 *Simulation Setup and assumptions*

For this simulation, the ETs are generating the same traffic as in previous simulations. In order to account for different users, the weights of the ETs are varied. The weights for each ET were divided into 3 categories: low ($\mu/2$),

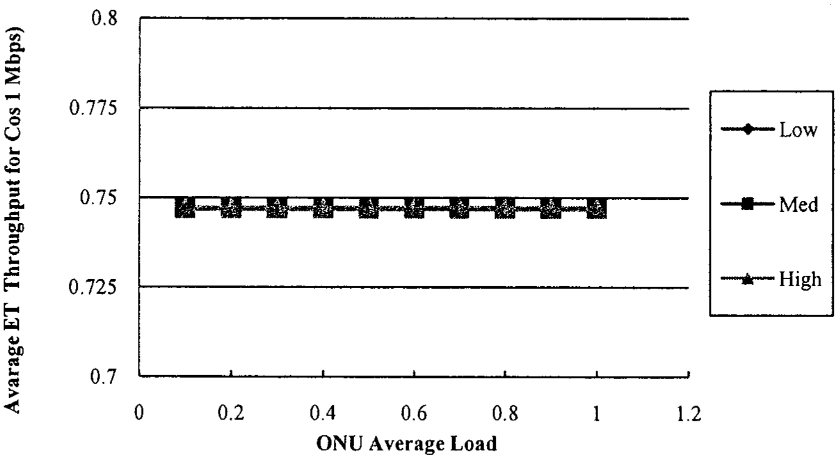


Figure 5.15 Average throughput Of CoS 1 per ONU Load

medium (μ) and high (μ*1.5) priorities. From our scheduling architecture different weights only effect the position of a particular ET in the priority queue for a particular CoS. The higher the weight of the ET-CoS pair the higher the priority. The sum of all weights of the same CoS is one, since μ = 1/ #of ETs,

5.4.2 *Simulation Result and Analysis*

In this simulation, we look at only one parameter, throughput. Throughput is an important parameter. Using throughput we can determine the rate the customer is getting, depending on high, medium or low priority.

CoS1 throughput is depicted in Figure 5.15. Although the expectation was to receive a variety of throughputs depending on the weight, this was not the case. The reason behind it is that for CoS1 the position in the priority queue does not matter: Recall from Chapter 4 that CoS1 is always the first to be considered and the CBR traffic can never exceed the ETs credit pool. Hence there are always enough credits for CoS1, no matter what the weights are.

CoS2 throughput becomes dependent on the weight of the ET only after the buffer of the ET approaches capacity, (Figure 5.16). At this point (0.5 load), the ET with the highest priority is constantly receiving more throughput. ETs with least priority are left with little capacity, as they only can use whatever is available in the CoS credit pool after the higher priority ETs. The relationship between the three priority throughputs can be seen to be relative to their weights.

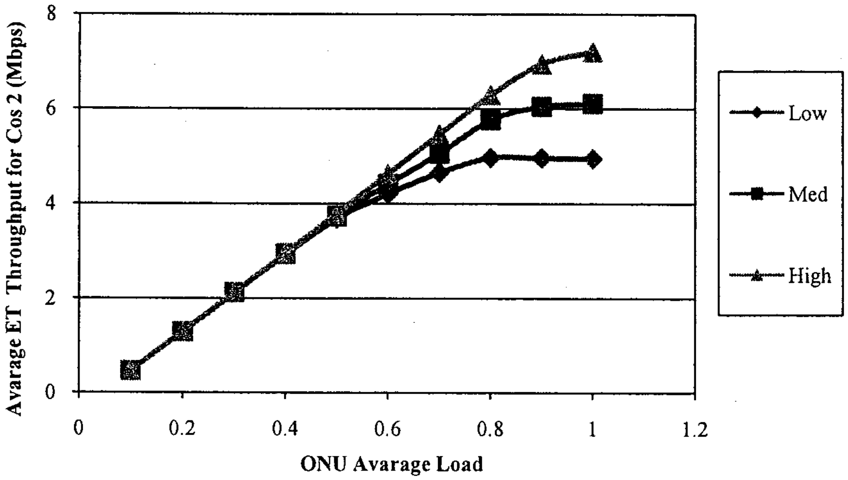


Figure 5.16 Average Throughput Of CoS 2 per ONU Load

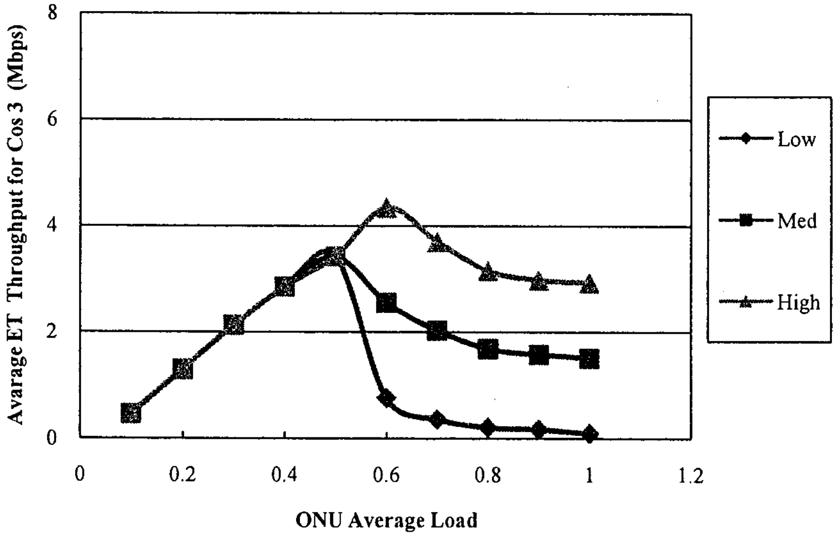


Figure 5.17 Average Throughput Of CoS 3 per ONU Load

CoS3 average throughput as expected remains unaffected by the weights up until the average ONU load reaches 0.5. (Figure 5.17) At this point the difference between the priorities is much more visible than in any other CoS. Compared to all other CoSs, dependency of CoS3 throughput on priority is the most significant. In light of the fact that CoS3 is best effort, and there is no bandwidth dedicated to it, the CoS3 bandwidth is getting distributed only during the second round of the algorithm. During this round priority plays an important role, as looking at Figure 5.13 we can clearly see that the lowest priority ETs are not capable of decent transmission at high load.

CHAPTER 6

IMPROVEMENT OF THE PROPOSED SYSTEM

While working LIPS we noticed two major drawbacks. Although they are not vital in the current environment, they might push future developers away if not addressed. We address these issues in this chapter by presenting two additions to the standard LIPS. One allows the system to exceed the limit of eight ETs per ONU, while the other one ensures no packet loss due to handover, when the ET roams.

As previously mentioned, the first drawback is the limit on the number of ETs per ONU. We modified Report and Gate message formats in order to be able to support eight ETs. The problem is maximizing the ONU capacity at only eight. *Shelving* is a key concept that would eliminate the problem. Moreover it would provide a variety of ONU unit designs, hence giving a choice to the provider.

The second drawback is handover packet loss when the buffer in the old ONU is "cleared" after an ET roams away from that ONU. The simulation result showed that the packet loss undergoes a great significant increase in fast-roaming ETs scenarios. This increase of the packet loss is seen in simulations depicted in Section 5.3.2.2. The future of hybrid access networks lies in integrating wireless mesh networks with fiber optics. This merging would require faster switching capabilities with much smaller packet loss.

6.1 ONU Shelving

Having a limited number of ETs per ONU is an obvious and intuitive problem. Since any protocol or networking changes would require a further modification of our modified gate message format, we realized that the only viable solution lies within the hardware. ONU shelving is a technique that would allow an ONU to have multiple sets of eight ETs.

Figure 6.1 shows the basic concept of ONU shelving proposed in this thesis. With ONU shelving concept, an ONU can have multiple shelves. Each shelf can house eight ETs. In order to interact with each shelf an OLT sends a Gate message specifically targeted to that shelf of the ONU. The first Gate message in the cycle would always correspond to Shelf1, the second Gate to Shelf2, etc. The Gate messages remained the same as it up to Round Robin within the ONU to switch between shelves of the ONU as depicted on Figure 6.1

The limitation of the shelving system is that under the MPCP protocol an OLT shall not issue more than one Gate message every 16.4 μs to a single ONU [11]. Hence using equation (4) we calculated that the Gate to *shelf+1* after transmitting Gate to *shelf* of the same ONU has to wait for:

$$\max\begin{cases}\dfrac{\sum_{n=1}^{N^{shelf}}(L_{req} + D^{nshelf})}{Cp} + g \\ 16.4\mu s\end{cases}$$

Where N^{shelf} is the current number of ETs in *shelf*, *Lreq* is the length of the request message, D^{nshelf} is the date transmitted by ET *n* of the *shelf*, *Cp* is the trunk capacity, and *g* is the guard time. If a scheduling architecture targets first shelf of all ONUs, then second shelf of all ONUs and so forth, the $16.4\mu s$ delay would rarely occur. Thus the overall effect on the system due to adding a single shelf to

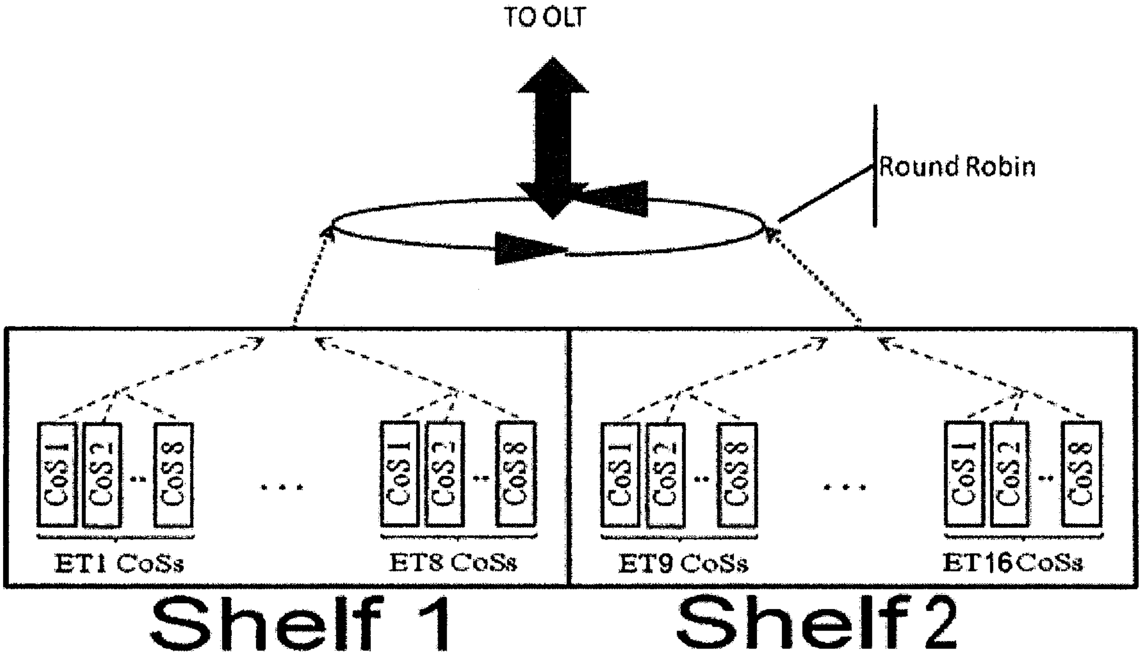


Figure 6.1 Shelving Structure of the ONU

an ONU is identical to adding a new ONU. In terms of bandwidth distribution and delay, second shelf will behave identical to a new ONU in the system. Although it terms of hardware adding extra shelf to the ONU is much cheaper than getting a new ONU unit.

6.2 Shadow Buffer Scheme

The shadow buffering scheme is an addition proposed for the OLT design. As we learned, every time an ET roams between ONUs, it leaves packets in the buffer of the old ONU. With the current buffer management mechanisms proposed in Section 2.1, the old ONU erases all the packets in its buffer immediately after it receives a "clear" signal from the OLT. The clear signal is issued on the third cycle after the ET roams away from the old ONU. Unfortunately, at heavy load conditions, the handover packet loss can be large, and hence, can have serious negative effects on the QoS and network throughput. In order to avoid handover packet loss, we propose the concept of "ET shadowing" in this section.

After the ET roamed to new ON, during the last cycle of three-way handshake described at Section 2.2.2 the OLT sends Gate message to the old ONU. With the shadow buffer design, this message indicates that the mobile ET is no longer connected to that ONU, but the packets that are still waiting in the ONU's buffer for that ET must be kept (rather than erased). For this reason, we refer to the current ET buffer in the old ONU as "shadow buffer". (see Figure 6.3). In the subsequent cycle, the old ONU responds by a modified Report message for the mobile ET with however a flag bit set. This flag bit is located in the "flags" field of the modified report message referred to as "shadow" flag (Figure 3.5). The shadow flag is meant to indicate to the OLT that this6 report message is associated with a shadow of the mobile ET, who is currently physically connected to another ONU. This helps the OLT to distinguish between the report message sent from the old ONU and the report message possibly sent from the new ONU for the same ET. The packets in the shadow buffer must be handled with priority over the requests for grant from the new ONU for the same ET, in order to avoid out of sequence packet transmission. The OLT is responsible for this priority mechanism.

During the scheduling time, after receiving all the report messages from the previous cycle, the OLT knows all the ETs and their shadow information. Since the MAC addresses of the ETs are sent to the OLT during the handshake process, the OLT knows which shadow buffer belongs to which ET. The OLT will then use modified bandwidth allocation architecture depicted in Figure 6.2. In the first step of the proposed architecture the ET's requests (*ET l*) and the ET's shadow requests (ET l_s) are combined. This makes the ET and its shadow to be amalgamated for the LIPS algorithm. The OLT then runs two rounds of the LIPS algorithm, described in detail in Section 4.3, to allocate the proper bandwidth to each ET-CoS pair. After the Grant algorithm is complete, the total bytes granted to ET-CoS pair and the shadow buffer request to the same ET-CoS pair are send to the "Request Subtractor". The subtractor grants shadow bandwidth requests first, thus avoiding out of sequence packet transmission. Then subtracts granted bytes from the total grant issued, and grants the rest to the regular ET. The OLT then uses the re-arranger to queue shadow grants and roamed grants to their respective ONUs; this process is identical to that of a standard (non-shadow) system. Overall the modification to the bandwidth allocation architecture insures that every ET is still entitled to its original bandwidth throughput. Even though its data is residing in two different ONUs.

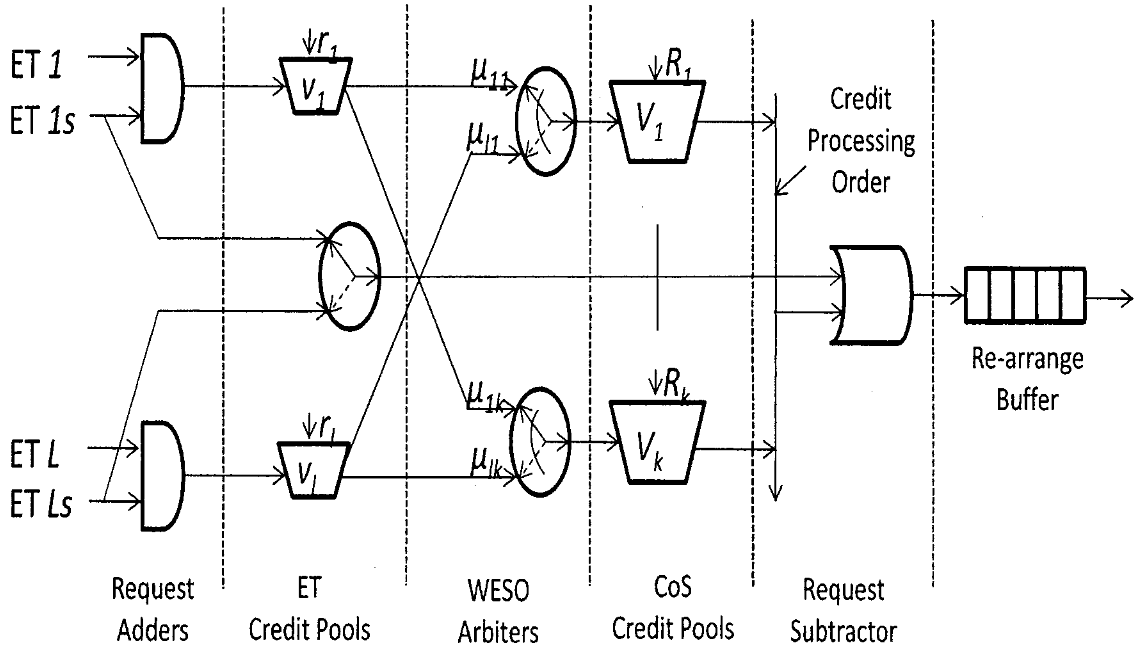
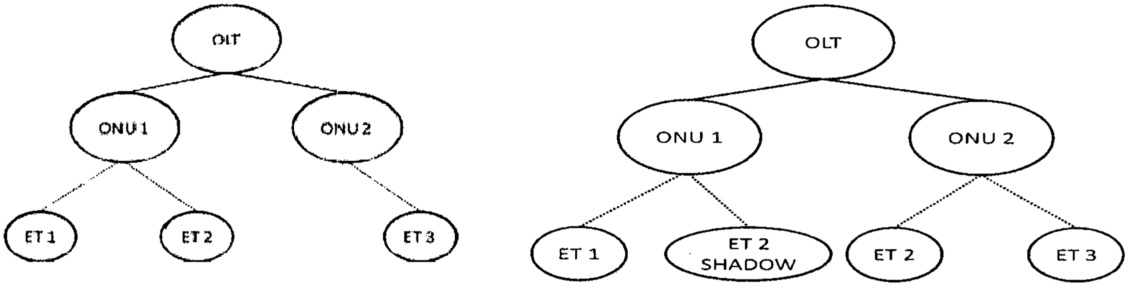


Figure 6.2 Modified Bandwidth allocation System



Before handoff
After handoff

Figure 6.3 ET handover between ONUs with Shadow

To ease the understanding of the proposed shadow buffer system we put forward the following example. Consider the following requests to the OLT and respective bandwidth grants to the ETs regular and shadow buffers in the Table 6.1. As previously stated during the bandwidth allocation the ET and its shadow bandwidth requests are combined. After OLT finalises total bandwidth granted to the respective ET-CoS pairs. OLT grants shadow requests first, leaving the remaining grant for each ET-CoS pair to the ET's new buffer. For this reason after awarding the represented ET with 70 bytes for CoS1 1000 bytes for CoS2 and 50 bytes for CoS3. OLT grants the old ONU with 0 bytes for CoS1, 500 bytes for CoS2 and 50 bytes for CoS3 for the shadow buffer, while also granting the new ONU with 70 bytes to CoS1, 500 bytes to CoS2 and 0 bytes to CoS3 for the roamed ET.

TABLE: 6.1 Shadow modification example

	Old ONU (Shadow)		New ONU		
COS	Request send (bytes)	Grants received (bytes)	Request send (bytes)	Grants grant (bytes)	Granted Total (bytes)
1	0	0	70	70	70
2	500	500	1200	500	1000
3	50	50	1200	0	50

If there are only CoS3 packets left in the shadow buffer or no packets at all, the ONU clears out the shadow buffer and sends no Report on its behalf. Thus, OLT grants shadow buffer unless the remaining queued traffic is best-effort. This method stops the OLT from scheduling extra time for the ONUs shadow buffer, and the ONU can use the newly available buffer space for incoming ETs.

We ran the shadow modification in the fast roaming ETs scenario (Section 5.3). As before, there are 16 ONUs, and 6 ETs per ONU. They can roam between different ONU domains randomly. The time an ET spends at each ONU is varied from 15s to 1500s. All traffic generation for all CoSs remains the same. The spatial weights of each WESO arbiter are set to the same value ($\mu_{nk} = 1/M \times N$).

The result of the shadow and non-shadow system can be seen in Figure 6.4 CoS1 packet loss and CoS2 handover packet loss ratio not shown since they are 0. CoS3 (C2/C3-Clear) handover packet loss becomes minimal in the shadow system and can be considered negligible. Shadow system and non-shadow system result in terms of congestion packet loss for CoS2 and CoS3 (C2/C3-Cong) are comparable. When ET roams in the shadow system the new ONU have to wait for old ONU buffers to be depleted before receiving the grants from the OLT, thus shadow congestion packet loss is always higher than the one of the non-shadow system.

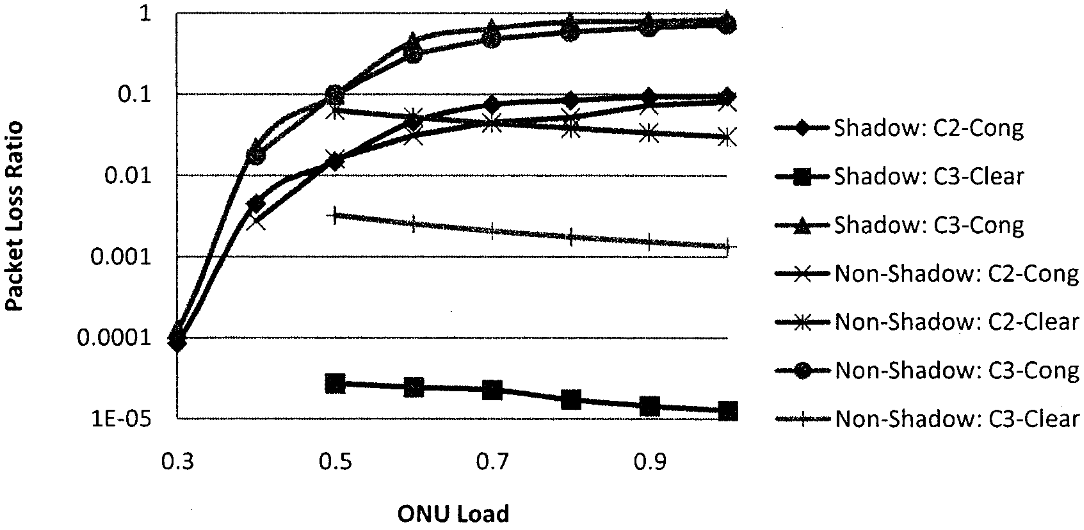


Figure 6.4 Packet Loss Ratio for shadow system

Overall the shadow system completely eliminates the packet loss in CoS2 traffic due to ET handover. It provides a good guarantee that in the future, if the ET were to roam faster, this system would be able to keep the minimal packet loss and would be less dependent on the speed of roaming.

CHAPTER 7

CONCLUSION AND FUTURE WORK

7.1 Conclusion

Throughout our research we showed that all recent studies done on WEPON networks were concentrated on improving the wireless front end. Parallel to that, the schemes previously proposed for EPON where inadequate to deal with the challenges of the WEPON infrastructure. Thus this thesis is unique in improving the optical back-end of WEPON networks to provide fair and deterministic service to the end user.

In this thesis we proposed a novel dynamic bandwidth allocation scheme for the back-end part of the integrated WEPON. Our scheme, which we have named Location-Independent Packet Scheduling (LIPS) can advantageously be used with the IEEE standardized Multi Point Control Protocol (MPCP) through the use of *modified* MPCP Gate and Report messages and an inventive grant encoding technique.

We developed an in-house simulation to investigate LIPS performance in terms of the throughput, packet loss, average delay and maximum delay. We proved that LIPS is exceptional in providing access to the users of the shared front-end wireless system while effectively balancing aggregate throughput and fairness. With the ability to maintain this fairness to the end user, LIPS remained analogous to the well known COPS and IPACT schemes in terms of average and maximum delay.

Further we proposed two advancements over our original system. Our first proposed improvement, "ONU shelving" maximized the number of ETs that can be concurrently attached to a given ONU. The “Shadow” buffering scheme was further presented as a second improvement. This scheme nullified the handover packet loss, resulting in a further improvement of the Quality of Service to the end user in the WEPON network.

7.2 Future Work

The most important next step is to integrate the wireless front-end algorithms like DARA or CaDAR with proposed optical back-end LIPS. This will further improve the service from the OLT to the wireless end users. Moreover slight modifications to either side should be expected for a smooth and productive symbioses of both algorithms in the WEPON.

Whilst this thesis has been developed, the 10G-EPON moved closer to becoming the next EPON standard, rather than a work in progress. Therefore we propose a detailed analysis on the behaviour of LIPS in the 10 Gbit/s environment.

Overall this thesis opened a lot of possibilities for future development. With this new design there are extra flag fields available in both the Gate and Report messages. Using the remaining flags, the system can be designed to maximize the number of possible ETs in the system or further improve the proposed shelving system. Based on our system even more versatile communication between the ETs and the OLT can be developed, that were not considered by the author of this thesis.

BIBLIOGRAPHY

[1] "Access Network Systems: North America – Optical Access. DLC and PON Technology and Market Report", *Report RHK-RPT-0548, RHK Telecommunication Industry Analysis*, SanFrancisco, June 2001

[2] G.Kramer, B.Mukherjee, A.Maislos, "Ethernet passive optical network" Available: http://networks.cs.ucdavis.edu/~mukherje/links/gk_wiley_bc.pdf

[3] G. Kramer and G. Pesavento, "Ethernet passive optical network (EPON): Building a next-generation optical access network," *IEEE Commun.* Mag., vol. 40, no. 2, pp. 66–73, Feb. 2002.

[4] G. Pesavento, M.Kelsey, "PONs for the broadband local loop" *Lightwave PennWell*, vol.16, no. 10 pp. 66-74

[5] G.Kramer, B. Mukherjee, and G.Pesavento, "Ethernet PON (ePON): Design and Analysis of an Optical Access Network," Photonic Network Communications, vol. 3, no. 3, pp. 307-319,July 2001.

[6] R. Ramaswami and K. N. Sivarajan, "Optical Networks, A Practical Perspective" *Morgan Kaufmann*, San Francisco, 1998.

[7] A. Dhaini, M. Chadi, A.Shami. "Dynamic Wavelength and Bandwidth Allocation in Hybrid TDM/WDM EPON Networks", Journal of lightwave technology, vol. 25, no. 1 january 2007

[8] R. Ramaswami and K. N. Sivarajan, *Optical Networks, A Practical Perspective*, Morgan Kaufmann, San Francisco, 1998.

[9] "EPON VS GPON: comparative Study" Nov 22, 2004.

[10] S.Lallukka, P.Raatikainen, "Link Utilization and Comparison of EPON and GPON Access Network Cost" *IEEE Globecom* 2005

[11] IEEE 802.3ah Task Force Homepage, [Online]. Available: http://www. ieee802.org/3/efm

[12] International Telecommunication Union G.984.1, [Online]. Available: http://www.itu.int/rec/T-REC-G.984.1-200803-I/en

[13] Teknovus Ltd., "Teknovus and Fiberxon Cooperate on "Turbo" EPON," *Teknovus Press Release*, available online at: http://findarticles.com/p/articles/mi_m0EIN/is_2007_Jan_22/ai_n17136156/.

[14] IEEE 802.3av Task Force Homepage, [Online]. Available: http://www.ieee802.org/3/av/

[15] A. Reaz, V.Ramamurthi, S.Sarkar, "CaDAR: an Efficient Routing Algorithm for Wireless-Optical Broadband Access Network", IEEE ICC 2008

[16] M.Kiese,E.Georgieva,D.Schupke "Availability Evaluation in Hybrid Wireless-OpticalBroadband Access Networks", IEEE ICCC 2009

[17] S. Sarkar, H.-H. Yen, S. Dixit, and B. Mukherjee, "A Novel Delay-Aware Routing Algorithm (DARA) for a Hybrid Wireless-Optical Broadband Access Network (WOBAN)," *IEEE Network Magazine*, vol. 22, no. 3, pp. 20–28, 2008.

[18] B.Lin, P.Ho, X.Shen, "Network Planning for Next-Generation Metropolitan-Area Broadband Access under EPON-WiMAX Integration", IEEE GLOBECOM 2008

[19] G. Kramer, B. Mukherjee, and G. Pesavento, "Ethernet PON (ePON): Design and analysis of an optical access network," *Photon. Netw. Commun.*, vol. 3, no. 3, pp. 307–319, 2001.

[20] M. Ma, Y. Zhu, and T. H. Cheng, "A bandwidth guaranteed polling MAC protocol for ethernet passive optical networks," in Proc. IEEE INFOCOM, San Francisco, CA, Mar.-Apr. 2003, vol. 1, pp. 22–31.

[21] G. Kramer, B. Mukherjee, and G. Pesavento, "Interleaved polling with adaptive cycle time (IPACT): A dynamic bandwidth distribution scheme in an optical access network," Photon. Netw. Commun., vol. 4, no. 1, pp. 89–107, Jan. 2002.

[22] H. Naser and H. T. Mouftah, "A joint-ONU interval-based dynamic scheduling algorithm for Ethernet passive optical networks," *IEEE/ACM Trans. Networking*,

[23] S. Floyd and V. Jacobson, "Random early detection gateways for congestion avoidance," *IEEE/ACM Trans. Netw.*, vol. 1, no. 4, pp. 397–413, Aug. 1993.

[24] D. Nikolova, B. Van Houdt, and C. Blondia, "Dynamic bandwidth allocation algorithms in EPON: A simulation study," in *Proc. Opticomm*, Oct. 2003, pp. 369–380.

[25] S. R. Sherif, A. Hadjiantonis, G. Ellinas, C. Assi, and M. A. Ali, "A novel decentralized ethernet-based PON access architecture for provisioning differentiated QoS," *J. Lightw. Technol.*, vol. 22, no. 11, pp. 2483–2497, Nov. 2004.

[26] V. Paxson and S. Floyd, "Wide-area traffic: The failure of Poisson modeling," *IEEE/ACM Trans. Netw.*, vol. 3, no. 3, pp. 226–244, Jun. 1995.

[27] W.Zhang, J.H," Modeling End-to-End Delay Using Pareto Distribution" IEEE ICIMP 2007

[28] G. Kramer, B. Mukherjee, S. Dixit, Y. Ye, and R. Hirth, "On supporting differentiated classes of service in EPON-based access networks," *J.Opt. Netw.*, vol. 1, no. 8/9, pp. 1–20, Aug. 2002.

[29] D. Sala and A. Gummalla, "PON functional requirements: Services and performance," in *IEEE 802.3ah Meeting*, Portland, OR, Jul. 2001 [Online]. Available: http://grouper.ieee.org/groups/802/3/efm/public/jul01/presentations/sala_1_0701.pdf

[30] Cisco Product Documentation, "Distributed Weighted Random Early Detection," Cisco Systems Inc. [Online]. Available: http://www.cisco.com/univercd/cc/td/doc/product/software/ios111/cc111/wred.htm