國 立 交 通 大 學

電機學院　電信學程

碩 士 論 文

乙太被動光纖網路中利用 PRNN 預估器作品質服務提升之動態頻寬配置研究

PRNN-based Predictive QoS-promoted Dynamic Bandwidth Allocation for Ethernet Passive Optical Networks

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中文摘要

乙太被動光纖網路（EPON）因其成本低廉之乙太網路設備及簡單、易維護之被動光纖架構，被認為是最佳之接取網路解決方案。隨著寬頻的普及與傳輸速率的提升，由語音、視訊與數據等結合的三合一服務（Triple Play）成為新趨勢，而乙太被動光纖網路整合既有基礎光纖網路與成熟之乙太網路技術，使得它成為提供三合一服務的極佳選擇。而為了提供服務使用者更快速以及更穩定的傳輸品質，有效地配置頻寬以保證服務品質（Quality of Service）便成為它的一個重要之研究課題。

在本篇論文中，我們提出了一個於乙太被動光纖網路中利用 PRNN 預估器作品質服務提升之動態頻寬配置方法。我們利用 PRNN 預估器收斂快速及預估準確之特性，針對及時性服務（例如語音服務、視訊服務）及非及時性服務（例如網際網路資料傳輸）於局端設備（OLT）作未及時回報之資料流量預測，而利用此預估結果作為頻寬分配之依據。模擬的結果顯示，我們所提出的利用 PRNN 預估器輔助作動態頻寬分配之方法確實能夠讓語音及視訊服務封包的平均延遲較 Q-DBA 方法分別改善 26%及 29%，對於資料封包而言，其平均延遲改善約 34%，而系統效能則改善約 2%。

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ABSTRACT

The Ethernet passive optical network (EPON), which represents the combinations of low-cost Ethernet equipment and simple fiber infrastructure, is considered to be the best solution to the next generation access network. Owing to the pervasion of broadband and the requirement of high-speed transmission, triple play services (combines voice, video, and data services) have become the current trend. With the integration of existing fiber network and mature Ethernet technology, EPON is considered to be the best candidate to support the triple play services. In order to provide faster and more stable transmission, an adaptive dynamic bandwidth allocation for QoS (quality of service) guaranteed has become a crucial research topic.

In this thesis, we propose a PRNN-based predictive QoS-promoted dynamic bandwidth allocation for EPONs. With the characteristics of fast convergence and accurate prediction, the PRNN-based predictor is selected to predict the late-reported traffic (real-time and non-real-time traffic) and the result of prediction is a basis for the dynamic bandwidth allocation at OLT. Simulation results show that the proposed predictive Q-DBA method improves the average voice and video delay time by an amount of 26% and 29% than Q-DBA method, respectively. The predictive Q-DBA method also improves the average data delay time about 34% and the system utilization by an amount of 2%.

誌　謝

利用工作之餘進修的確需要專注力及意志力方能竟全功，這次修讀碩士專班給了我很好的機會驗證學術與實務間的異同，對我日後工作相信會有很大之助益。碩士論文能夠順利完成，首先由衷感謝張仲儒教授耐心而仔細的教導，不論是在學術上或是做人處世方面都給予我最大的幫助。感謝正文學長在這幾年來對我的幫忙，不論是在問題的討論、各項資料蒐集及提供都對我助益良多。也感謝家慶學長、文祥學長不吝提供相關參考資料，使我能順利完成研究。

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吳星毅 謹誌

民國九十六年

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Chapter 1

Introduction

The “access network” (also referred to as “last mile” or “local loop”) has been a popular topic recently. Due to the rapid growth of the internet traffic in the past few years, a larger bandwidth requirement of the access network is needed urgently. Nowadays, the most widely deployment configurations of broadband access network are digital subscriber loops (DSL) and CATV networks. In Taiwan, up to the end of year 2006, the subscribers of xDSL and cable modem have exceeded 3.3 millions [1]. DSL uses the same copper twisted pair as telephone line where the data rate transmits in the range of 64 Kbps to 100 Mbps. Table 1.1 lists the data rate and available distance for xDSL (families of DSL) and cable modem services. As presented in Table 1.1, either xDSL or cable modem has its limitation in the data rate and available distance, which are the bottlenecks for the development of traditional access networks.

Optical fiber, with its characteristics of unlimited bandwidth, low signal loss, long distance transmission, non-conductive, and low price, has been widely applied in the core network for a long time. Due to the increasing demands for bandwidth in access network and the maturity of the optics technology, the deployment of optical fiber in the access network is an inevitable trend. As a matter of fact, optical fiber is absolutely a good solution to the bottlenecks of xDSL and cable modem, it has the capability of delivering integrated, plesio-infinite broadband voice, data, and video services at the distance beyond 20 km in the access network and has the ability to fulfill the multi-demands of broadband services.

xDSL	Downstream Data Rate	Upstream Data Rate	Transmission Distance (km)
IDSL (ISDN-like DSL)	128 Kbps	128 Kbps	5.5
HDSL (High bit rate DSL)	T1/E1	T1/E1	3.5/2.7
SDSL (Symmetric DSL)	1.5~2 Mbps	1.5~2 Mbps	4~3
UDSL (Universal DSL)	2 Mbps	2 Mbps	5.5
ADSL (Asymmetric DSL)	1.5~8 Mbps	64~1000 Kbps	5.5~3.7
ADSL2+(Asymmetric DSL)	24 Mbps max.	1 Mbps max	6
RADSL (Rate-Adaptive DSL)	12 Mbps	1 Mbps	3.7
VDSL (Very high bit rate DSL)	13~52 Mbps	1.5~2.3 Mbps	1.4~0.3
VDSL2 (Very high bit rate DSL2)	1~100 Mbps	1~100 Mbps	5~0.2
Cable modem	27~36 Mbps	0.768~10 Mbps	＞100

Table 1.1: Data rate and transmission distance of xDSL and cable modem

The Ethernet passive optical network (EPON), which represents the combinations of low-cost Ethernet equipment and low-cost fiber infrastructure, is considered to be the best solution to the next generation access network. As shown in Figure 1.1, the EPON is a point to multi-point optical network with no active elements in the signal's path from source to destination. The only interior elements used in EPON are passive optical components, such as optical fiber, couplers, combiners and splitters. All transmissions in an EPON are performed between an optical line terminal (OLT) and optical network units (ONUs). The OLT resides in the local exchange (central office) and connects the optical access network to an Internet protocol (IP), asynchronous transfer mode (ATM), or synchronous optical network (SONET) backbone. The ONU is located either at the end-user location (fiber-to-the-home (FTTH) and fiber-to-the-business (FTTB) solutions), or fiber-to-the-curb (FTTC solution) and provides broadband voice, data, and video services. Several topologies are suitable for an EPON (Figure 1.2), such as tree, ring, bus or combinations of these three topologies. In the access network, tree topology is more common used for the easy deployment and the economical use of optical fibers. Thus, for the rest of our thesis, we will focus our attention on the tree topology, as shown in Figure 1.1.

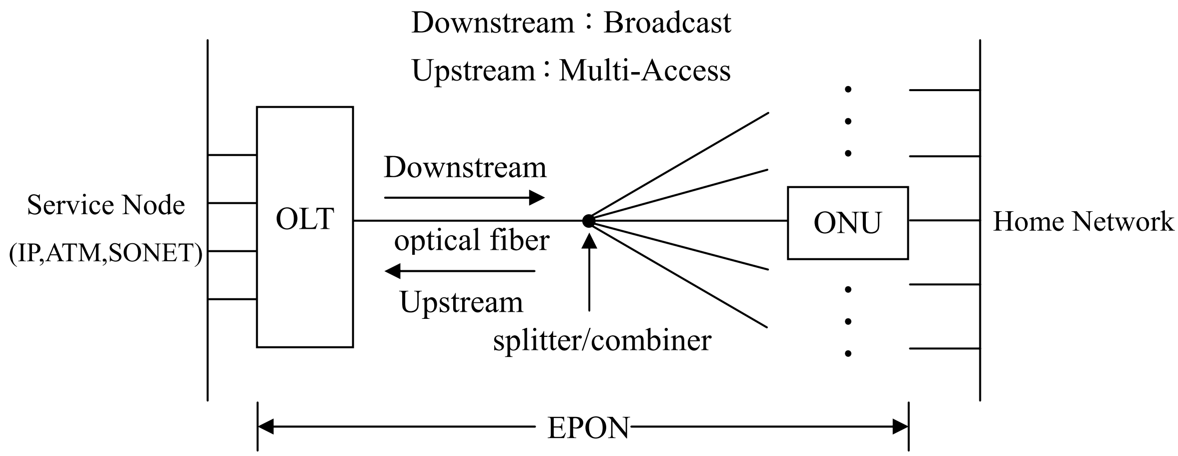


Figure 1.1: EPON-tree network

In the downstream direction (defined as frames traveling direction from the OLT to the ONU(s)), EPON is a broadcasting media (point to multi-point network). Ethernet packets transmitted by the OLT pass through a 1:N passive splitter/combiner and reach each ONU. An ONU discards packets that are not destined to it before passing the rest of them to a user. In the upstream direction (defined as frames traveling direction from the ONU(s) to the OLT), unlike downstream direction, EPON uses multi-access (multi-point to point network). Since its multi-access characteristics, the upstream multiple access scheme of EPON is the most challenging and interesting issue. The ONU needs to employ some arbitration mechanism to avoid data collisions and fairly share the fiber-channel capacity. This is achieved by allocation of a transmission window (timeslot) to each ONU. Each timeslot is capable of carrying several Ethernet packets. The ONU should buffer packets received from a subscriber until its timeslot arrives. When its timeslot arrives, the ONU should "burst" all stored packets at full channel speed. The timeslot size may be fixed or variable, for example, it may depend on the amount of data stored in the waiting queues of the ONU.

Due to its tremendous capacity of transmission, EPON is capable of transmitting various sorts of communication services, including voice, video, and data. To support the multitude of application of EPON, we classify the traffic into classes of service (CoS) and provide differentiated treatment to each class [2]. In this thesis, traffic is categorized into three classes: voice, video and data.

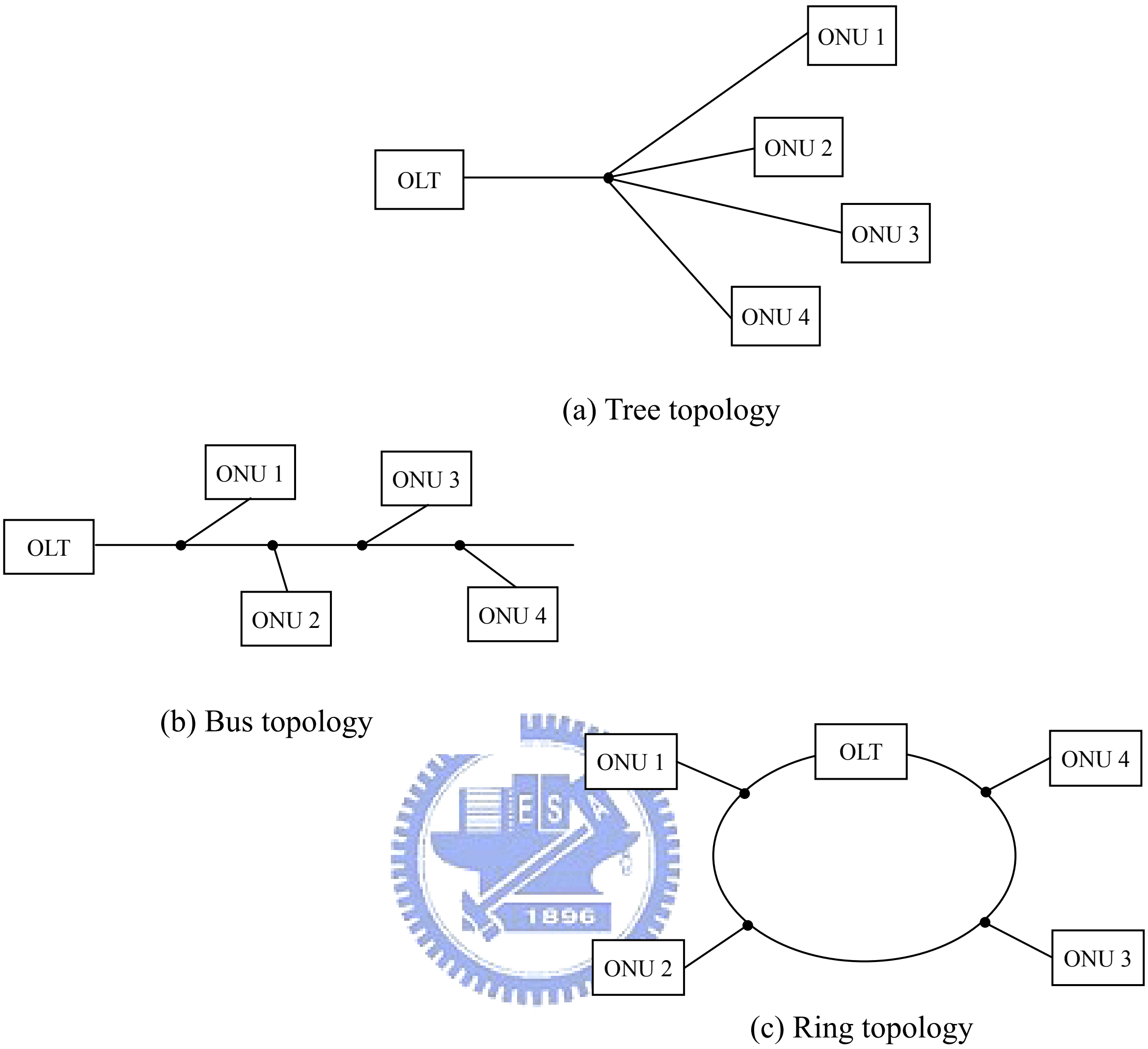


Figure 1.2: EPON topologies

Recently, PON technology has been discussed enthusiastically. Except the EPON technology, GPON (Gigabit-Capable PON) seems to be the most attractive one among xPON technologies. Table 1.2 is the analysis of technologies about APON (ATM PON), BPON (Broadband PON), GPON, and EPON. APON is a technology based on ATM structure and developed by FSAN[3] (Full Service Access Network, an interest group for the world's leading telecommunications services providers, independent test labs, and equipment suppliers to work towards a common goal of truly broadband fiber access networks.) in 1990s’. BPON, the extension of APON, is also developed in FSAN by 1998. Owing to the complicated ATM structure, APON and BPON are not deployed extensively. As for the GPON, it is also developed by FSAN and standardized in 2004. GPON can support not only the Ethernet service but also the ATM and TDM service, and therefore, GPON gets more and more interest lately.

Characteristic	APON / BPON	GPON	EPON
standard	ITU-T G.983	ITU-T G.984	IEEE 802.3ah
Downstream line rates (Mbps)	155 or 622	1244 or 2488	1250
Upstream line rates (Mbps)	155 or 622	155, 622, 1244, or 2488	1250
Payload encapsulation	ATM	ATM or GFP	Ethernet Framing
Addressing capability (minimum)	32	64	16
Addressing capability (maximum)	64	128	256
Logical reach (km)	20	20	10 or 20

Table 1.2: Analysis of technology in xPON

For its capability of supporting Ethernet service, ATM service, and TDM service, GPON is thought to be the powerful competitor of EPON. But according the analysis of experts in the market trends, EPON is still believed the best technology, why? We summarize the advantages of using EPON in subscriber access network as follows.

(i) EPON minimizes the fiber deployment in access network (both local central office and local loop).

(ii) EPON allows for long reach between central office and customers (over 20 km).

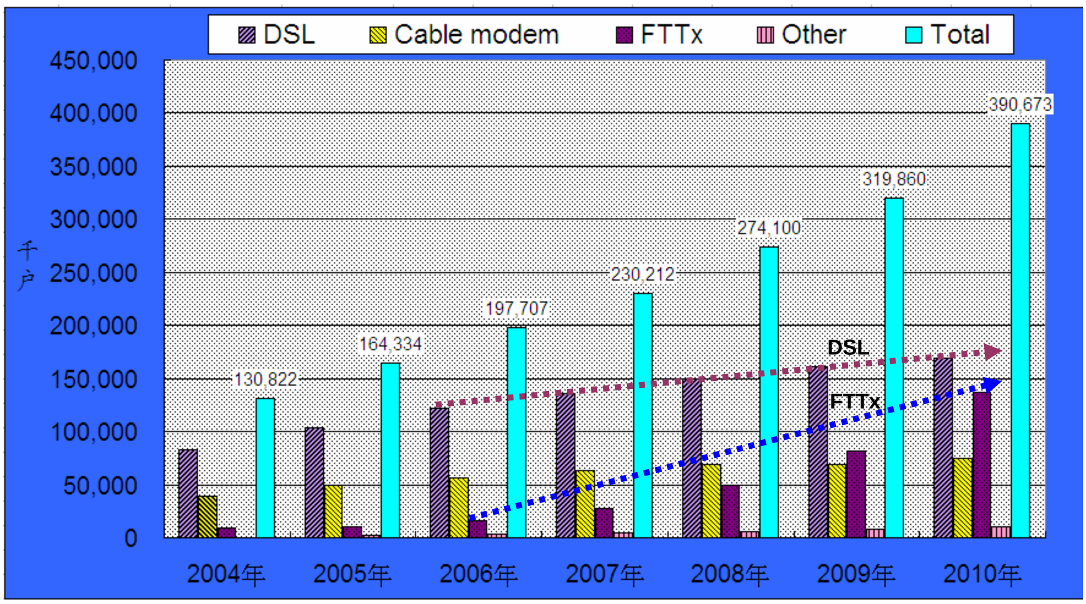
(iii) EPON provides higher bandwidth (gigabit solutions).

(iv) Because there are only passive elements in EPON, EPON eliminates the necessity to maintain the active curbside units and provide power to them.

(v) EPON allows upgrades to higher bit rates or additional wavelengths owing to optically transparent end to end.

(vi) The cost of deployment in EPON is lower than that in GPON due to the widely use of Ethernet.

Figure 1.3 shows the statistics and forecast of worldwide broadband customers from 2004 to 2010. According to the statistics of ITRI IEK, there are about 0.2 billion broadband customers in 2006, most of them are DSL users. But we can see that the increase of DSL users will slow down after 2008, and the customers of FTTx will increase rapidly. ITRI IEK estimates that there are about 0.4 billion broadband customers in 2010, and the customers of FTTx are about 0.13 billion, that is, the development of FTTx is prosperous in the future.

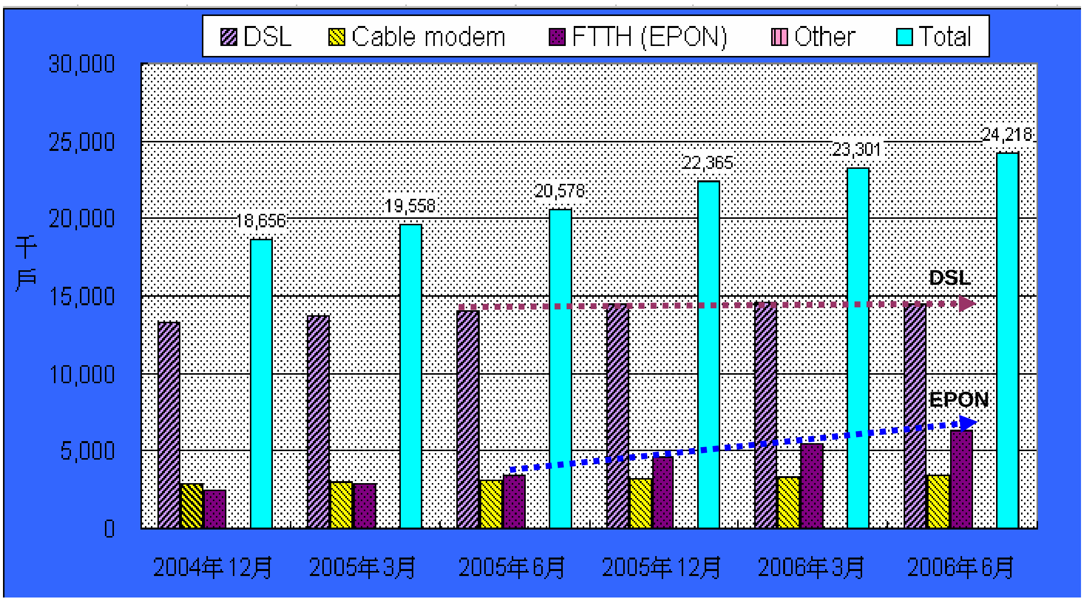


Source: Industrial Technology Research Institute IEK(工研院產業經濟與趨勢研究中心) (2007/05)

Figure 1.3: Statistics and forecast of worldwide broadband customers

EPON plays an important role in FTTx. Let's focus on the development of EPON in Japan, the customers of EPON increases rapidly than the customers of DSL since 2006, as depicted in Figure 1.4. We can also found that the customers of DSL have been decreasing from Q1 to Q2, 2006 (about 27 thousand customers' attenuation). According the explanation of the authorities in Japan, because the deployment of fiber is widely throughout the megalopolis in

Japan, and the price of EPON and DSL are almost the same, therefore, the development of EPON is faster than that of DSL in Japan.



Source: 日本總務省(MIC) (2006/09)

Figure 1.4: Statistics of broadband customers in Japan

In EPON, the upstream transmission (packets transmissions from ONU to OLT) uses a GATE-REPORT mechanism, and the research of the dynamic bandwidth allocation (DBA) is so attractive that our thesis will focus on the discussion of the DBA mechanism for EPON.

For the remainder of the thesis, chapter 2 describes some related works about scheduling algorithms. Chapter 3 proposes a PRNN-based predictive Q-DBA method. Chapter 4 discusses the result of simulation and evaluation of performance. Finally, conclusion remarks are presented in Chapter 5.

Chapter 2

Related Works

An EPON cannot be considered as a shared medium or a point to point network. In the downstream direction, Ethernet frames transmitted by the OLT pass through a 1:N passive splitter to reach each ONU. The packets broadcasted by the OLT will be filtered by their destination ONUs based on the medium access control (MAC) addresses. In the upstream direction, due to the directional property of a passive combiner, packets from any ONU can only reach the OLT, not other ONUs. Hence, ONUs have to share the trunk fiber channel in the upstream, and medium access control is required to deploy ONUs' transmission to prevent data from being corrupted owing to multiple ONU transmitting at the same time.

Owing to the directional property of optical splitter, ONUs cannot easily detect a collision at the OLT. So, the conventional carrier sense multiple access/ collision detection (CSMA/CD) is difficult to implement in EPON. Furthermore, the bandwidth distribution to each ONU cannot be controlled, guaranteed and make it very hard for any form of quality of service (QoS) to be supported. Considering the above factors, a time division multiple access (TDMA) scheme on a common wavelength for scheduling is more attractive for upstream traffic. In this Chapter, we present some related works about scheduling algorithms, including IPACT, QLP, LQF, GLQF, PLQF, and Q-DBA method.

2.1 Interleaved Polling with Adaptive Cycle Time (IPACT)

As proposed in [4], interleaved polling with adaptive cycle time (IPACT) is a well-designed scheme for dynamic bandwidth allocation in EPON because it minimizes unused bandwidth by using polling messages which are interleaved on downstream Ethernet traffic. In IPACT, the OLT distributes variable size time windows whose size depends on the amount of frames buffered at the respective ONUs, as reported by the respective ONUs, using control messages. Each ONU executes the same procedure driven by a grant message received from the OLT. It is an OLT-based centralized dynamic medium access arbitration scheme; thus, it is easy to adaptively change the scheduling at run-time based on the actual network condition. Also, it considers the large propagation delay of an EPON that could span as far as 20km. IPACT is an attractive proposition because, firstly, no synchronization is needed among ONUs, and, secondly, polling message is interleaved with frame transmission so that the overall overhead arisen from the propagation delay is reduced and the efficiency is higher. Unfortunately, IPACT does not explicitly consider the fact that different ONUs have different requirements due to the differences in subscribers service level agreements (SLAs). In addition, IPACT's dynamic bandwidth allocation scheme also introduces burstiness in the transmission of traffic from each ONU to the OLT.

The simulations in [4] assume that all ONUs have the same maximum transmission window W_{max}. How the OLT determine the granted window size if the requested window size $W^{[i]} < W_{max}$? A few approaches have been introduced, including, (i) Fixed service: It ignores the requested window size and always grants the maximum window. (ii)Limited service: It grants the requested windows size but no more than W_{max}. (iii)Constant credit scheme: A constant credit is added to the requested window size to compensate the data arriving between the time an ONU sent the Request and received the Grant message. (iv)Linear credit scheme: A similar method as the constant credit scheme, the credit size is proportional to the requested window. (v)Elastic service: It has an attempt to the fixed maximum window limit, and the maximum cycle time is the only limiting factor. Among these approaches described above, it is concluded that the limited service is the best service discipline.

2.2 QLP Scheduling Algorithm

In [5], queue length proportional (QLP) scheduling algorithm concentrates on how the bandwidth can be efficiently allocated according to the real traffic load. The bandwidth allocation is implemented by considering both the bandwidth and the queue length in each buffer proportionally. The QLP is able to achieve a high throughput as well as to improve the utilization of buffer space by avoiding the possible cell loss caused by the Max-Min fairness algorithm under the heavy congestion situation. The QLP algorithm assigns an input port with a rate proportional to its buffer queue length. Let L_1, L_2, …, L_N be the buffer queue length for input n (n=1,2,…,N), $L_T = L_1 + L_2 + \ldots + L_N$ be the total queue length. Let R_1, R2,…, R_N be the assigned bandwidth by output m (m=1,2,…,N) for input 1,2,…,N respectively. $R_T = R_1 + R_2 + \ldots + R_N$ is the available bandwidth of output m for the best effort traffic. Applying the QLP algorithm, we got

$$R_1 = \frac{L_1}{L_1 + L_2 + \ldots + L_N} R_T = \frac{L_1}{L_T} R_T ,$$

$$\vdots$$

$$R_N = \frac{L_N}{L_1 + L_2 + \ldots + L_N} R_T = \frac{L_N}{L_T} R_T .$$

$$\frac{R_i}{L_i} = \frac{R_T}{L_T} = \frac{1}{\mu}, \quad i = 1, 2, \ldots N.$$

Because the bandwidth allocation of QLP is based on the traffic queue length and the available bandwidth, so the buffer space and bandwidth will not be wasted for possible mismatch between them. A high throughput and low cell loss ratio cab be achieved because that QLP considers the constraint of the buffer space inside a switch. QLP has another conspicuous feature that the heavy load traffic in an input port can logically share buffers of the other input ports although there are no physical congestion among them.

After the analysis in this paper, the QLP is considered to be able to maximize overall throughputs and handle buffer utilization comparing to those in conventional scheduling algorithms.

2.3 LQF Scheduling Algorithm

It has been known that the throughput of the N x N port input-queued switch with FIFO (First In, First Out) queues is limited to 58.6%, [14]. The throughput is limited since a cell can be hindered by another cell ahead of it which is destined for a different output. This is so called HOL (Head of Line) blocking. This result applies only to input-queued switches with FIFO queues. The HOL blocking can be eliminated by using a simple buffering strategy, that is, rather than maintain a single FIFO queue for all cells, each input maintains a separate queue for all cells. When the traffic is independent and uniform, a number of algorithms demonstrate that the throughput can reach 100%. But all of these algorithms perform less well when traffic is non-uniform.

Nick McKeown proposed longest quest first (LQF) scheduling algorithm in [6], LQF gives preferential service to the longest queue by using a maximum weight matching algorithm, where each integer-value weight is set to the corresponding queue length. It was proved in N. Mckeown's paper that, for independent and either uniform or non-uniform arrivals, using the LQF scheduling algorithm, the throughput could be increased to approximately 100% compared to the throughput of traditional FIFO scheduling algorithm.

In Nick McKeown's previous paper, it was found that LQF can achieve 100% throughput for independent uniform and non-uniform traffic by considering the occupancies of queues (queue length). As a matter of fact, the LQF is very difficult to put into practice in hardware at high speed. A new algorithm called Longest Port First (LPF) [15] is proposed to conquer the complexity problems of LQF. In LPF each weight is a function of queue lengths (the weights are not exactly equal to the queue lengths, but are similar.). LPF can be implemented in hardware at high speed. It is proved that LPF can also achieve 100% throughput for both uniform and non-uniform traffic.

2.4 GLQF Scheduling Algorithm

Generalized longest queue first (GLQF) scheduling algorithm was proposed in [7]. Firstly, the sources are classified into N classes so that sources in one class have the same cell loss probability requirement. Under this scheduling algorithm, each buffer is assigned a positive number as its weight according to the product of the corresponding weight of class and the queue length. The scheduler transmits the cell from the buffer that has the maximal weight. The advantage of this algorithm is that it can adapt to momentary overload rapidly comparing to the traditional LQF scheduling algorithm. This feature ensures that the overloaded buffer could receive almost the overall service and reduce the losses owing to the buffer overflows and cell delay variation.

A method of analysis of which approximates the queue length distribution was developed for this GLQF. This method works well from low to medium utilization. As for the high utilization, a heavy traffic limit theorem for the Generalized Longest Queue First scheduling algorithm was proven. Finally, in this thesis, the smaller variance in the behavior of the maximum queue length explains that GLQF scheduling algorithm provides smaller cell delay variation.

2.5 PLQF Scheduling Algorithm

In [8], an improved scheduling algorithm called prediction-based longest queue first (PLQF) was introduced. Due to the natural randomness of broadband traffic in ATM networks, instantaneous traffic bursts may take place and surpass the network bandwidth. And the PLQF is designed to achieve minimal cell loss rate (CLR) when the incoming traffic is bursty. After analyzing the conventional LQF scheduling algorithm, it is found that the LQF may not be applicable for high bursty traffic. That is, when the incoming traffic to the buffer is bursty, the conventional LQF scheduling algorithm can cause poor buffer utilization and CLR. The PLQF is designed to solve this problem. Under this scheduling algorithm, the scheduler considers not only the queue length, but also the incoming traffic. The scheduler can adjust the queue lengths in advance to provide to the incoming traffic.

The scheduler of PLOF calculates a weight for each user considering the incoming traffic. The scheduler always serves the user who has the maximum weight. Based on the information of user queue length and the incoming traffic, the resource is allocated to the user who has the highest probability to overflow in the near future.

The measure QoS parameter in this paper is CLR. For the PLQF algorithm, two versions, on-line PLQF (in which predictor is used) and off-line PLQF (in which traffic information is pre-known) are simulated. Through the theoretical analysis in this paper, it is proved that the PLQF scheduling algorithm could reduce the CLR by 10% to 60% than conventional LQF scheduling algorithms.

2.6 Q-DBA Method Scheduling Algorithm

In [9], an improved scheduling algorithm called QoS-promoted dynamic bandwidth allocation (Q-DBA) was introduced. Six kinds of information of queues included in the report message of ONU were sent to OLT. In addition to the traditional information of the occupancy of real-time voice packets, real-time video packets, and non-real-time data packets, the

Q-DBA scheduling mechanism also considers three new kinds of information of queues, the delay criterion of video packets, the dropping probability of video packets, and the waiting time of data packets. Upon receiving the report messages from each ONU, the OLT allocates the available bandwidth in a proportional method. The sequence of bandwidth allocation is as follows. The voice packets get the first priority, then the video packets which violate the delay criterion and the dropping probability, the data packets whose waiting time is larger than the waiting bound acquire the fourth priority, and finally, the normal video and data packets have the lowermost priority. According to the simulation result in this paper, the Q-DBA method improves the performance in EPON, including the average video packet dropping probability, the average data packet blocking probability, the average video delay, and the overall fairness index of packet.

Chapter 3

The PRNN-based Predictive Q-DBA Method for EPON

3.1 System Architecture

The proposed EPON architecture is shown in Figure 3.1. There are an OLT, M ONUs and a 1:M splitter between OLT and ONUs. The splitter is used to broadcast packets from the OLT to all the ONUs. The downstream transmission is the packets broadcasting from the OLT to the ONU and the upstream transmission is the packets transmitted from the ONU to the OLT following a GATE-REPORT mechanism [2]. There are OLT with line rate R_E (bps) between OLT and each ONU, and M ONUs with line rate R_U (bps) between ONU and its own end users. In ONU*i*, $1 \le i \le$ M, three classes of queues are provided to store real-time voice, real-time video, and non-real-time data packets coming from the users and are denoted by $Q_{0,i}$, $Q_{1,i}$, and $Q_{2,i}$, respectively. The queue size of $Q_{0,i}$, $Q_{1,i}$ and $Q_{2,i}$ are $|Q_{0,i}|$, $|Q_{1,i}|$ and $|Q_{2,i}|$, respectively. The packet controller of the ONU puts the incoming packets from users into the corresponding queue according to their types and will drop some type of packets if the queue is full. Base on the Q-DBA method [9], the packet controller will also drop certain type of packets if they violate their delay criteria. The queue manager takes over the management of the transmission between the OLT and the ONUs, that is, it transmits and receives packets between OLT and the ONUs. In addition, the queue manager also takes charge of generating the REPORT message.

To transmit data in upstream channel (from the ONU to the OLT) of an EPON, the polling scheme is considered the better way. That is, the ONU sends a request message (REPORT) to the OLT to get the required timeslots in the next transmission cycle. According to the requests of the ONUs, the OLT sends a grant message (GATE) to the ONU and allocates the granted timeslots in a fixed or dynamic length. The former algorithm is called GATE-REPORT mechanism [2]. The OLT sends a GATE signal to inform the ONU of when and how to transmit data to the OLT. Relatively, the ONU will send a REPORT signal tailed with the data to inform the OLT of the requested bandwidth in the next transmission cycle.

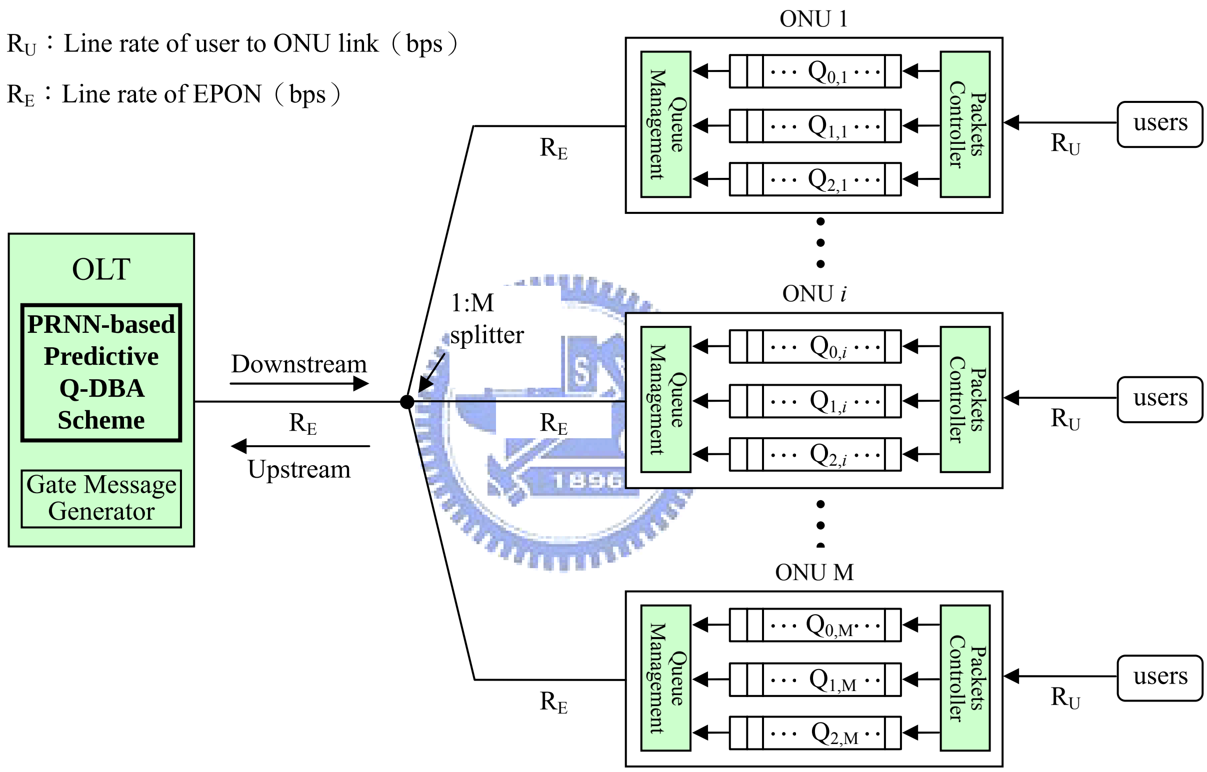


Figure 3.1: System architecture of EPON

Figure 3.2 is the PRNN-based predictive Q-DBA scheme, the PRNN predictor will be introduced in section 3.2, and the predictive Q-DBA will be presented in section 3.3.

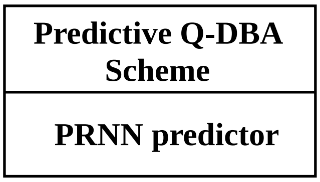


Figure 3.2: The PRNN-based predictive Q-DBA scheme

The format of GATE and REPORT messages are shown in Figure 3.3. Upon receiving the GATE message from the OLT, the ONU begins to transmit packets. In the meantime, the ONU will keep receiving packets from the user. After finishing the packets transmission, the ONU will generate a REPORT message to inform the OLT of the requested packets in the next cycle. If the packets in the queue are null, the ONU still sends REPORT message but the requested packets is zero.

GATE message	Octets
Destination address (DA)	6
Source address (SA)	6
Length/Type = $88-08_{16}$	2
Opcode = $00-02_{16}$	2
Timestamp	4
Number of grants/flags	1
[Grant #1 start time]	4
[Grant #1 length]	2
[Grant #2 start time]	4
[Grant #2 length]	2
[Grant #3 start time]	4
[Grant #3 length]	2
[Grant #4 start time]	4
[Grant #4 length]	2
Pad/Reserved	15/39
Frame check sequence (FCS)	4

REPORT message	Octets
Destination address (DA)	6
Source address (SA)	6
Length/Type = $88-08_{16}$	2
Opcode = $00-03_{16}$	2
Timestamp	4
Number of queue sets	1
[Report bitmap]	1
[Queue #1 report]	2
[Queue #2 report]	2
[Queue #3 report]	2
[Queue #4 report]	2
[Queue #5 report]	2
[Queue #6 report]	2
[Queue #7 report]	2
[Queue #8 report]	2
Pad /Reserved	0-39
Frame check sequence (FCS)	4

[ ]: optional

Figure 3.3: Format of GATE and REPORT messages

3.2 The PRNN-based Prediction

First, we define notations in the following:

cycle n (of ONUi) : the time difference between the OLT first beginning to receive the nth REPORT message from ONUi and the next beginning to receive the (n+1)st REPORT message from the same ONUi;

$L_{0,i}(\mathrm{n})$: the reported occupancy of queue $Q_{0,i}$ in bytes at cycle n;

$L_{1,i}(\mathrm{n})$: the reported occupancy of queue $Q_{1,i}$ in bytes at cycle n;

$L_{2,i}(\mathrm{n})$: the reported occupancy of queue $Q_{2,i}$ in bytes at cycle n;

$L_{dp,i}(\mathrm{n})$: the total amount of video packets (in bytes at cycle n) which will be dropped at the end of the next cycle if they are not transmitted at the next cycle because their delay time will violate the delay constraint;

T_d^* : the delay constraint of video packets in seconds;

$L_{d,i}(\mathrm{n})$: the total amount of video packets (in bytes at cycle n) which should be transmitted at the next timeslot in order to sustain the requirement of packet dropping probability;

P_d^* : the dropping probability bound of video packet;

$L_{w,i}(\mathrm{n})$: the total amount of data packets (in bytes at cycle n) whose waiting time is larger than a waiting bound;

T_w^* : the waiting bound of data packets in seconds;

$G_{m,i}(\mathrm{n})$: the granted bandwidth in bytes for $Q_{m,i}$ at cycle n;

$L_{m,i}(\mathrm{n})$: the reported occupancy of queues $Q_{m,i}$ at cycle n;

$\tilde{E}_{m,i}(\mathrm{n})$: the amount of estimated new arrival packets for $Q_{m,i}$ in bytes during cycle n;

$P_{m,i}(\mathrm{n})$: the predicted occupancy of $Q_{m,i}$ in bytes during cycle n;

$T_i(\mathrm{n})$: the cycle time of EPON in seconds at cycle n for ONUi;

$\tilde{\lambda}_{m,i}(\mathrm{n})$: the estimated packet arrival rate in bytes/sec. during cycle n;

$A_{m,i}$(n) : the amount of actual new arrival packets at $Q_{m,i}$ in bytes during cycle n;

$\lambda_{m,i}$(n) : the actual packet arrival rate in bytes/sec. during cycle n.

The proposed PRNN-based predictive QoS-promoted dynamic bandwidth allocation (PRNN-based predictive Q-DBA method) assumes that ONU*i*, $1 \leq i \leq$ M, sends report message including six sorts of information of queues, $L_{0,i}$(n), $L_{1,i}$(n), $L_{2,i}$(n), $L_{dp,i}$(n), $L_{d,i}$(n), and $L_{w,i}$(n). The $L_{0,i}$(n), $L_{1,i}$(n), and $L_{2,i}$(n) denote the reported occupancy of queues $Q_{0,i}$, $Q_{1,i}$, and $Q_{2,i}$ in bytes at cycle n, respectively. The $L_{dp,i}$(n) denotes the total amount of video packets (in bytes at cycle n) which will be dropped at the end of the next cycle if they are not transmitted at the next cycle because their delay time will violate the delay constraint of video packet, T_d^*. The $L_{d,i}$(n) denotes the total amount of video packets (in bytes at cycle n) which should be transmitted at the next timeslot in order to sustain the requirement of packet dropping probability of video packet, P_d^*. The $L_{w,i}$(n) denotes the total amount of data packets (in bytes at cycle n) whose waiting time is larger than a waiting bound, T_w^*.

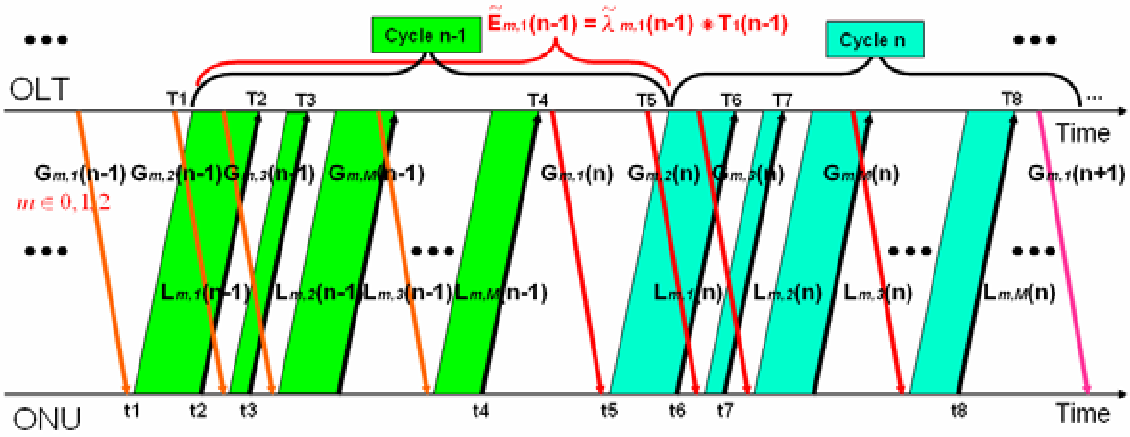


Figure 3.4: Upstream timeslot assignments

Figure 3.4 shows the upstream timeslot assignment between the OLT and the ONU. "A cycle n of ONU*i*" is defined as the time difference between the OLT first beginning to receive the nth REPORT message from ONU*i* and the next beginning to receive the (n+1)st REPORT message from the same ONU*i* (As depicted in Figure 3.4, T5-T1 is the cycle n-1 for ONU1).

When the ONU receives the GATE message from the OLT, it is ready for ONU to transmit packets at its timeslot. For example, as shown in Figure 3.4, upon receiving $G_{m,1}$(n-1) , $m \in 0$, 1, 2, the granted bandwidth in bytes for $Q_{m,1}$ with service type m at cycle n-1, ONU1 starts to transmit packets at t1 (assigned by OLT), the ONU1 will end its transmission at t2 and give a report message to OLT with the reported queue occupancy. In the meantime between T1 and T5, the ONU1 will keep receiving packets from the user.

Assume that the OLT is at the present cycle n-1(start at T1, as depicted in Figure 3.4). In order to make a better utilization of bandwidth, we need to consider the new arrival packets at cycle n-1 for ONUi. That is, we make a prediction of the amount of ONUi's new arrival packets at cycle n-1.

Denote $\tilde{E}_{m,i}$(n-1) to be the amount of estimated new arrival packets of $Q_{m,i}$ in bytes at cycle n-1 and $P_{m,i}$(n) to be the predicted occupancy of $Q_{m,i}$ in bytes at cycle n. $P_{m,i}$(n) can be obtained by

$$P_{m,i}(\text{n}) = L_{m,i}(\text{n-1}) + \tilde{E}_{m,i}(\text{n-1})\,. \tag{1}$$

Denote T_i(n-1) to be the cycle time of EPON in seconds at cycle n-1 for ONUi and $\tilde{\lambda}_{m,i}$(n-1) to be the estimated packet arrival rate in bytes/sec. during cycle n-1. The $\tilde{E}_{m,i}$(n-1) can be calculated as follows

$$\tilde{E}_{m,i}(\text{n-1}) = \tilde{\lambda}_{m,i}(\text{n-1}) * T_i(\text{n-1})\,. \tag{2}$$

Through the analysis of the relationship between the reported queue occupancy and the granted bandwidth, we could find the way to get the amount of actual new arrival packets at $Q_{m,i}$. For example, let $A_{m,i}$(n-2) to be the amount of actual new arrival packets at $Q_{m,i}$ in bytes at cycle n-2. At cycle n-1, the OLT can obtain the amount of actual new arrival packets $A_{m,i}$(n-2) according to the reported queue occupancy of $Q_{m,i}$, $L_{m,i}$(n-1), $L_{m,i}$(n-2) and the granted bandwidth $G_{m,i}$(n-1). If the reported queue occupancy $L_{m,i}$(n-1) is greater than zero, then the amount of actual new arrival packets $A_{m,i}$(n-2) can be calculated from

$G_{m,i}$(n-1), $L_{m,i}$(n-1), and $L_{m,i}$(n-2). If the reported queue occupancy equals to zero, assume that the actual new arrival packets arrive in uniform distribution, then the $A_{m,i}$(n-2) can be obtained as follows

$$A_{m,i}(\text{n-2}) = \begin{cases} G_{m,i}(\text{n-1}) + L_{m,i}(\text{n-1}) - L_{m,i}(\text{n-2}), & \text{if } L_{m,i}(\text{n-1}) > 0, \\ \\ \dfrac{G_{m,i}(\text{n-1}) - L_{m,i}(\text{n-2})}{2}, & \text{if } L_{m,i}(\text{n-1}) = 0. \end{cases} \tag{3}$$

Denote $\lambda_{m,i}$(n-2) to be the actual packet arrival rate at cycle n-2, it can be calculated as follows

$$\lambda_{m,i}(\text{n-2}) = \frac{A_{m,i}(\text{n-2})}{T_i(\text{n-2})}. \tag{4}$$

As mentioned earlier, in order to make a better utilization of bandwidth, we need to consider the mount of new arrival packets to the ONU between each ONU's report time. In this thesis, we estimate the mount of new arrival packets using the PRNN predictor. As depicted in Figure 3.6, let $\lambda_{m,i}$(n-2) to be the input of PRNN predictor, then the estimated packet arrival rate $\tilde{\lambda}_{m,i}$(n-1) can be obtained from the output of PRNN predictor. For example, as depicted in Figure 3.4, the OLT can make a prediction of the new packet arrival rate $\tilde{\lambda}_{m,1}$(n-1) at T2, then $\tilde{\lambda}_{m,2}$(n-1) at T3, and so forth. After the OLT has collected all the predicted data for ONU*i* at cycle n-1 (at T4), the proposed DBA mechanism allocates the available bandwidth to each ONU. The details of proposed DBA mechanism will be discussed in next section and the description of PRNN predictor will be introduced in the following.

Least mean square (LMS) and recursive least squares (RLS) are two linear adaptive structures for the prediction of signals. LMS is firstly introduced by Widrow and Hoff in 1959, the strength of the LMS resides on its simplicity and easy processing. Afterward there are some improved algorithms for LMS, such as normalized least mean square (NLMS), fast least mean square (FLMS), discrete cosine transform - least mean square (DCT - LMS) [10]. RLS is a representative of the other prediction algorithms, it is based on the method of the least squares on the theory of Kalman filters. The main difference between LMS and RLS is the inherent statistical conception. In RLS algorithm, we work with time-based averages calculated from different samples of the same random process, but on the contrary, in LMS algorithm, averaging involves values acquire from specific time but from different realization of one random process. Because of the calculation of time average requires more instructions, so the computational complexity of RLS-based is one order higher than that of LMS-based methods. But from the point of view of the performance of convergence rate, the RLS-based algorithms are several times speedy than LMS-based algorithms.

In 1995, Haykin and Li [11] presented a nonlinear predictor based on the pipelined recurrent neural network (PRNN). The PRNN is believed to outperform the LMS and RLS in the prediction of nonlinear and non-stationary signals. The PRNN is composed of a number of small recurrent neutral networks (RNN's) but keep its relatively low computational complexity. A real-time recurrent learning algorithm (RTRL) [12] is used by Haykin and Li for the training of the PRNN and it helps to predict the nonlinear and non-stationary signals more precisely and accurately.

The PRNN predictor is a modular neural network, and consists of a certain number r of RNN's as its module, each module is composed of N neurons. The PRNN predictor has good nonlinear prediction capability and fast convergent time. A fully connected RNN structure, which has N neutrons and p + q + N input nodes, is shown in Fig.3.5. The predicted data $\tilde{\lambda}_{m,i}$(n+1) is expressed as follows (For simplicity, the lower subscript is omitted.)

$$\tilde{\lambda}(\text{n+1}) = H(\tilde{\lambda}(\text{n}),\ldots,\tilde{\lambda}(\text{n-p+1});1;\tilde{\lambda}(\text{n}),\ldots,\tilde{\lambda}(\text{n-q+1});y_2(\text{n-1}),\ldots,y_N(\text{n-1})), \tag{5}$$

where $H(\cdot)$ is an unknown nonlinear function. The j th neutron first calculate a weighted sum, denoted by $v_j(n)$, and is given by equation (6)

$$v_j(n) = \sum_{i=1}^{p+q+N} w_{ij}(n)u_i(n), \tag{6}$$

where w_{ij} denotes the weight of the connection from the ith input node to the jth neuron, $u_i(n)$ is the ith input node. The output $y_j(n)$ is the transformation of the $v_j(n)$ by a sigmoidal activation function, which is given by

$$y_j(n) = \varphi\left(v_j\left(n\right)\right) = \frac{1}{1+exp(-v_j(n))}. \tag{7}$$

We can get the weight w_{ij} by using RTRL algorithm [13]

$$w(n+1) = w(n) - \eta\frac{\partial C(n)}{\partial w}, \tag{8}$$

where η is a fixed learning rate parameter, $C(n)$ is the cost function defined as

$$C(n) = \sum_{i=1}^{q} \alpha_n^{\,i-1} e^2(n-i+1), \tag{9}$$

where α_n is an exponential forgetting factor that lies in the range of $0 \le \alpha_n \le 1$, and e is the prediction error.

As shown in Fig.3.6, the PRNN is a pipelined structure of the NARMA-based RNN predictor. All the modules of the PRNN are designed to have exactly the same synaptic weight matrix W. The predicted data $\tilde{\lambda}_{m,i}$ (n-1) can be obtained from the first neutron of the first module.

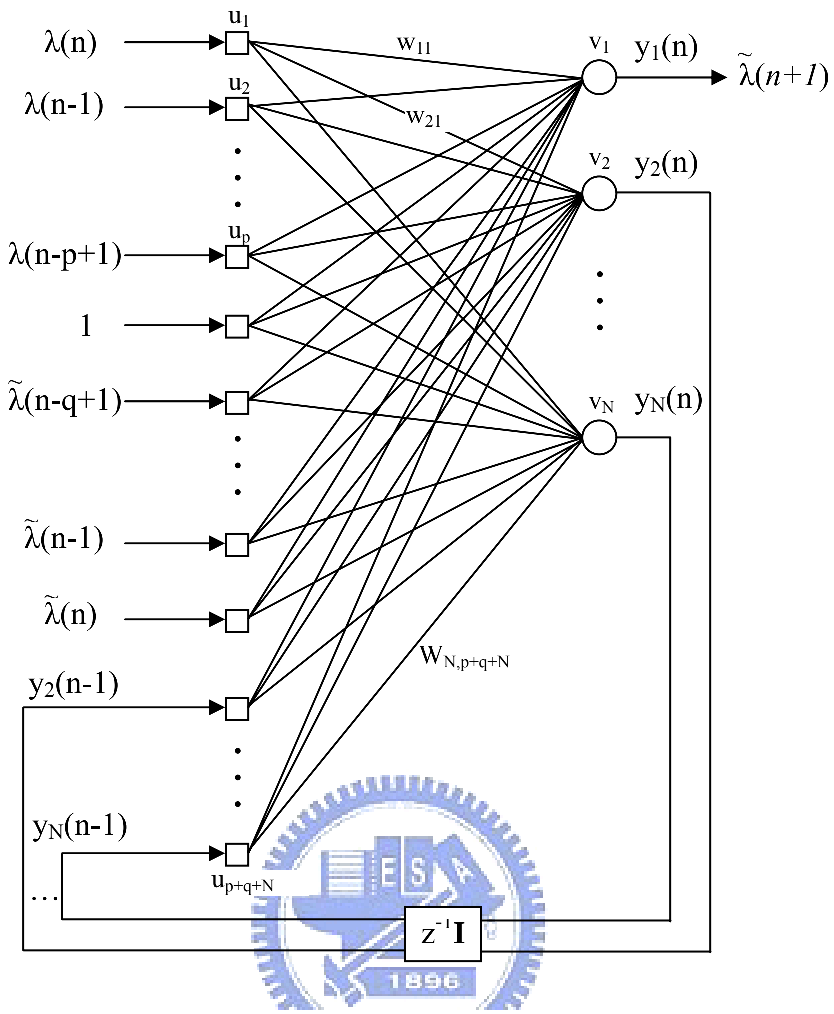


Figure 3.5: Fully Connected Structure of the RNN

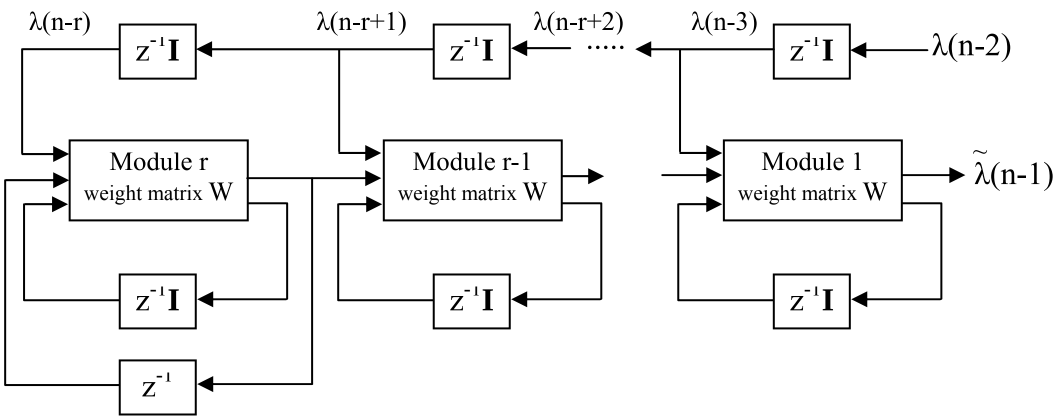


Figure 3.6: Pipelined Recurrent Neural Network (PRNN)

3.3 The Predictive Q-DBA Method

The OLT determines the bandwidth allocation. Upon receiving all report messages from each ONU, that is, $L_{0,i}(\text{n})$, $L_{1,i}(\text{n})$, $L_{2,i}(\text{n})$, $L_{dp,i}(\text{n})$, $L_{d,i}(\text{n})$, and $L_{w,i}(\text{n})$. For simplicity, the label of cycle n for each ONU will be omitted in the following discussion. According to the PRNN-based prediction of the new arrival packets of queues $Q_{0,i}$, $Q_{1,i}$, and $Q_{2,i}$ proposed earlier, the report messages can be adjusted to $P_{0,i}$, $P_{1,i}$, $P_{2,i}$, $P_{dp,i}$, $P_{d,i}$, and $P_{w,i}$, respectively. Denote B to be the total bandwidth of each ONU. The allocated bandwidths are sent to each ONU by the GATE message. The GATE message tells ONU*i* the start time of the packet transmission and the amount of bandwidth in bytes ONU*i* can transmit. As soon as ONUs receive their own GATE message, they are ready to transmit the packets.

In order to guarantee the QoS requirements, we classify the packets which come from users into six priorities. Due to its delay sensitivity, the predictive voice data $P_{0,i}$ get the highest priority. The predictive video packets $P_{dp,i}$ and $P_{d,i}$ which will be dropped if they are not transmitted at the next cycle have the second and third priorities. Then we consider the starvation of the predictive data packets, $P_{w,i}$, whose waiting time outrun the waiting bound, and they have the fourth priority in the bandwidth allocation. After that, we deal with the unallocated predictive video and data packets, $P_{1,i}$ and $P_{2,i}$, they have the fifth and sixth priorities. Finally, if there is residual bandwidth after the previous bandwidth allocation, the predictive video and data packets share the leftover bandwidth proportionally until the bandwidth is finished up.

The bandwidth allocation of proposed PRNN-based predictive Q-DBA method is described in detail as follows.

[*Step 1*]: Bandwidth allocation to voice

Denote $G'_{0,i}$ as the allocated bandwidth in $Q_{0,i}$. Based on the predicted queue occupancy of $Q_{0,i}$, $P_{0,i}$, and the total bandwidth B, $G'_{0,i}$ is given by

$$G'_{0,i} = \begin{cases} P_{0,i}, & \text{if } \sum_{i=1}^{M} P_{0,i} \le B, \\ B \times \dfrac{P_{0,i}}{\sum_{i=1}^{M} P_{0,i}}, & \text{elsewhere.} \end{cases} \tag{10}$$

[*Step 2*]: Bandwidth allocation to video packets with the second and the third priorities

Denote $G'_{1,i}$ as the allocated bandwidth in $Q_{1,i}$. Based on the predicted data, $P_{dp,i}$, $P_{d,i}$ and the total bandwidth B, $G'_{1,i}$ is given by

$$G'_{1,i} = \begin{cases} P_{dp,i}, & \text{if } B - \sum_{i=1}^{M} G'_{0,i} \ge \sum_{i=1}^{M} P_{dp,i}, \\ P_{d,i} + (B - \sum_{i=1}^{M} [G'_{0,i} + P_{d,i}])^{+} \times \dfrac{P_{dp,i} - P_{d,i}}{\sum_{i=1}^{M} (P_{dp,i} - P_{d,i})}, & \text{if } \sum_{i=1}^{M} P_{d,i} < B - \sum_{i=1}^{M} G'_{0,i} < \sum_{i=1}^{M} P_{dp,i}, \\ (B - \sum_{i=1}^{M} G'_{0,i})^{+} \times \dfrac{P_{d,i}}{\sum_{i=1}^{M} P_{d,i}}, & \text{if } B - \sum_{i=1}^{M} G'_{0,i} \le \sum_{i=1}^{M} P_{d,i}. \end{cases} \tag{11}$$

[*Step 3*]: Bandwidth allocation to data packets with the fourth priority

Denote $G'_{2,i}$ as the allocated bandwidth in $Q_{2,i}$. Based on the predicted data, $P_{w,i}$ and the total bandwidth B, $G'_{2,i}$ is given by

$$G'_{2,i} = \begin{cases} P_{w,i}, & \text{if } B - \sum_{i=1}^{M} [G'_{0,i} + G'_{1,i}] > \sum_{i=1}^{M} P_{w,i}, \\ (B - \sum_{i=1}^{M} [G'_{0,i} + G'_{1,i}])^{+} \times \dfrac{P_{w,i}}{\sum_{i=1}^{M} P_{w,i}}, & \text{elsewhere.} \end{cases} \tag{12}$$

[*Step 4*]: Bandwidth allocation to video packets with the fifth priority

Denote $G_{1,i}^{''}$ as the allocated bandwidth in $Q_{1,i}$. Based on the predicted data, $P_{1,i} - P_{dp,i}$ and the residual bandwidth , $G_{1,i}^{''}$ is given by

$$G_{1,i}^{''} = \begin{cases} P_{1,i} - G_{1,i}^{'}, \quad \text{if } B - \sum_{i=1}^{M}[G_{0,i}^{'} + G_{1,i}^{'} + G_{2,i}^{'}] > \sum_{i=1}^{M}(P_{1,i} - G_{1,i}^{'}), \\ (B - \sum_{i=1}^{M}[G_{0,i}^{'} + G_{1,i}^{'} + G_{2,i}^{'}])^{+} \times \dfrac{P_{1,i} - G_{1,i}^{'}}{\sum_{i=1}^{M}(P_{1,i} - G_{1,i}^{'})}, \quad \text{elsewhere.} \end{cases} \tag{13}$$

[*Step 5*]: Bandwidth allocation to data packets with the sixth priority

Denote $G_{2,i}^{''}$ as the allocated bandwidth in $Q_{2,i}$. Based on the predicted data, $P_{2,i} - P_{w,i}$ and the residual bandwidth , $G_{2,i}^{''}$ is given by

$$G_{2,i}^{''} = \begin{cases} P_{2,i} - G_{2,i}^{'}, \quad \text{if } B - \sum_{i=1}^{M}[G_{0,i}^{'} + G_{1,i}^{'} + G_{2,i}^{'} + G_{1,i}^{''}] > \sum_{i=1}^{M}(P_{2,i} - G_{2,i}^{'}), \\ (B - \sum_{i=1}^{M}[G_{0,i}^{'} + G_{1,i}^{'} + G_{2,i}^{'} + G_{1,i}^{''}])^{+} \times \dfrac{P_{2,i} - G_{2,i}^{'}}{\sum_{i=1}^{M}(P_{2,i} - G_{2,i}^{'})}, \quad \text{elsewhere.} \end{cases} \tag{14}$$

[*Step 6*]: Residual bandwidth allocation

Denote $G_{0,i}^{''}$ and $G_{1,i}^{'''}$ as the allocated bandwidth in $Q_{0,i}$ and $Q_{1,i}$, respectively. Based on the predicted data, $P_{0,i}$, $P_{1,i}$ and the residual bandwidth , $G_{0,i}^{''}$ and $G_{1,i}^{'''}$ is given by

$$\begin{cases} G_{0,i}^{''} = (B - \sum_{i=1}^{M}[G_{0,i}^{'} + G_{1,i}^{'} + G_{1,i}^{''} + G_{2,i}^{'} + G_{2,i}^{''}]^{+} > \dfrac{P_{0,i}}{\sum_{i=1}^{M}(P_{0,i} + P_{1,i})}, \\ G_{1,i}^{'''} = (B - \sum_{i=1}^{M}[G_{0,i}^{'} + G_{1,i}^{'} + G_{1,i}^{''} + G_{2,i}^{'} + G_{2,i}^{''}]^{+} > \dfrac{P_{1,i}}{\sum_{i=1}^{M}(P_{0,i} + P_{1,i})}. \end{cases} \tag{15}$$

[*Step 7*]: Gate message generation

Denote $G_{0,i}$, $G_{1,i}$, and $G_{2,i}$ to be the final granted bandw

idth in the GATE message to $Q_{0,i}$, $Q_{1,i}$,and $Q_{2,i}$ respectively, so they can be given by

$$\begin{cases} G_{0,i} = G'_{0,i} + G''_{0,i}. \\ G_{1,i} = G'_{1,i} + G''_{1,i} + G'''_{1,i}. \\ G_{2,i} = G'_{2,i} + G''_{2,i}. \end{cases} \tag{16}$$

$G_{1,i}$, $G_{1,i}$, and $G_{2,i}$ are included in the GATE message, and the OLT sends the GATE message to $Q_{0,i}$, $Q_{1,i}$,and $Q_{2,i}$, respectively. Each follows the information in the GATE message to transmit its own packets and sends the new REPORT message to OLT at the end of timeslot.

Chapter 4

Simulation Results and Discussions

4.1 Simulation Environment

An event-driven packet-based simulation is elaborated to the performance of the proposed PRNN-based predictive Q-DBA, the Q-DBA [9], and the DBAM [16]. The simulation environment is described as follows.

(i) A PON architecture of 32 ONUs connected in a tree topology (as depicted in Figure 3.1).

(ii) The distance between the OLT and the ONU is 20km.

(iii) The distance between the ONU and the splitter (1:32) is 500m.

(iv) The line rate between the OLT and the ONU is 1 Gbps.

(v) The line rate between the ONU and the end user is 100 Mbps.

(vi) For simulation simplification, we set the cycle time in a fixed value of 0.72 ms.

(vii) Each ONU has three priority queues, the buffer space of each ONU is set to 1 Mbytes.

(viii) The guard time between each transmission window is set to 1 μs.

We consider three kinds of packets, real-time voice packets, real-time video packets, and non-real-time data packets. The real-time voice packets possess the highest priority, then the real-time video packets, and the non-real-time data packets are the lowest. The ON-OFF voice traffic stream is generated by a two level MMDP (Markov Modulated Deterministic Process). To emulate T1 connection, there are 24 channels in a T1 link. The ONU aggregates the traffic from every channels. The average durations of talk spurs and silence periods are assumed to be exponentially distributed with $1/\alpha$=1 sec. and $1/\beta$=1.35 sec. The generation rate of a voice packet is constant bit rate (CBR) and the packet size is fixed to 70 bytes.

The video and data packets are modeled by ON/OFF Parato-distributed model in order to generate self-similar traffic. Most network traffic (i.e., http, ftp, VOD (Video on Demand), MOD (Multi-media on Demand), etc.) can be characterized by self-similarity and long-range independence (LRD) according to lots of studies [17]-[20]. The ON/OFF Parato-distributed model is to generate the highly bursty video and data packets. The packet size of each packet arrival is in uniform distribution and ranges form 64 to 1518 bytes.

The choice of voice delay criteria is according to the recommendation of ITU-T: 1.5ms one way propagation delay in access network. Since six kinds of priority packets are concerned, we have to decide the choice of video delay constraint T_d^*, the video dropping probability bound P_d^*, and the waiting bound of data packets T_w^*. These three parameters: T_d^*, P_d^*, T_w^* are defined 10ms, 1%, and 500ms respectively. For the performance parameters, at first we consider the average packet delay of voice, video and data packets individually. After that we show the dropping probability of video packets, the blocking probability and the starvation ratio of data packets. At last we discuss the overall system utilization.

4.2 Simulation Results

In this section, we show the performance of the PRNN-based predictive Q-DBA method. All the simulation results of the proposed PRNN-based predictive Q-DBA Method are comparing to that in DBAM [16] and Q-DBA method [9]. The traffic arrival rates are set as follows:

Voice service: 4.48Mbps x 32 (iid)

Video service: 0.55Mbps x 32 (iid) ~ 15.75Mbps x 32 (iid)

Data service: 0.28Mbps x 32 (iid) ~ 7.27Mbps x 32 (iid)

For DBAM [16], the ONUs send report message according to their queues' occupancies and the waiting times between last and present timeslots. A maximum window of total bandwidth requirement is pre-assigned according to service level agreement (SLA) in OLT. When the total bandwidth requirement of the ONU*i* is less than the pre-assigned maximum window, the allocated total bandwidth equals its requirement. When the total bandwidth requirement of the ONU*i* is larger than the pre-assigned maximum window, the allocated total bandwidth equals the pre-assigned maximum windows. Besides, the rule of allocating bandwidth to voice and video packets is similar as that in allocating the total bandwidth for ONU*i*. At last, the residual bandwidth is allocated to data packets.

For Q-DBA, The ONU*i* send six kinds of information of queues to the OLT. Upon receiving the report messages from each ONU, the OLT allocates the available bandwidth according to the priorities of report information in a proportional method. The steps in bandwidth allocation are set as follows:

(i) Bandwidth allocation to voice packets.

(ii) Bandwidth allocation to video packets with the 2nd and 3rd priority.

(iii) Bandwidth allocation to data packets with the 4th priority.

(iv) Bandwidth allocation to video packets with the 5th priority.

(v) Bandwidth allocation to data packets with the 6th priority.

(vi) Residual bandwidth allocation.

Since the voice dropping probability is zero whether in predictive Q-DBA, Q-DBA or DBAM, we omit to show the simulation result, this result is due to the QoS requirement for voice packets.

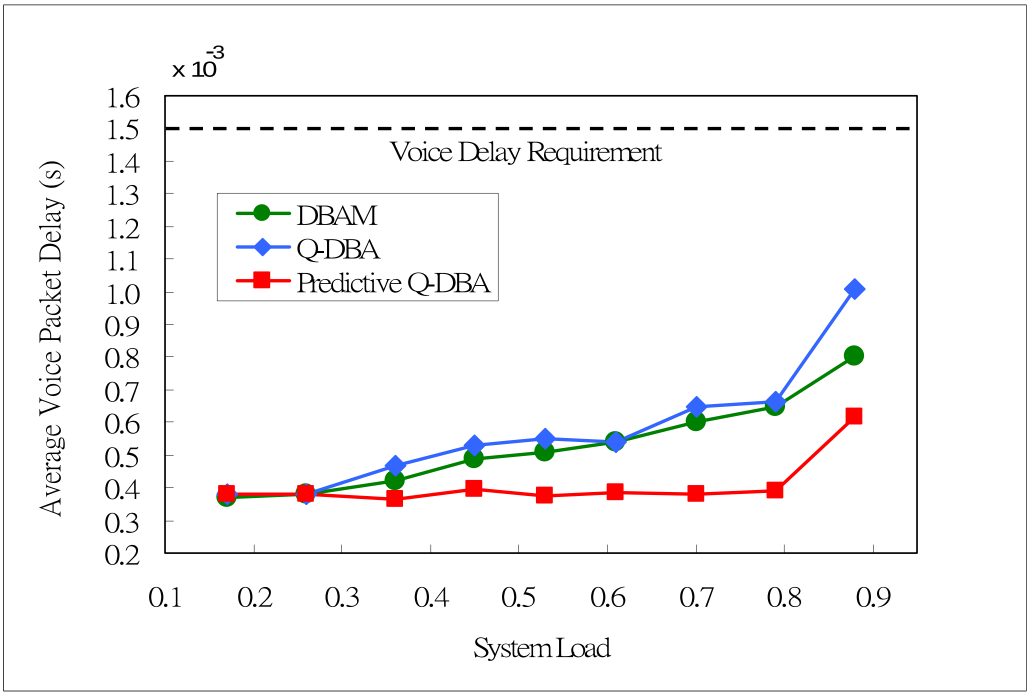


Figure 4.1: Average voice delay time versus the system load in EPON

Figure 4.1 shows the average voice delay time versus the system load in EPON. It can be found that all the voice delays of the three schemes are within the requirement. However, the voice delay in DBAM and Q-DBA increases with the increasing of the system load, but in predictive Q-DBA, the delay time is almost the same when the system load is below 0.8. It is because the predictive Q-DBA makes a prediction of the new arrival voice packets which arrive between two consecutive reporting times for each ONU, thus the allocated bandwidth could meet the actual traffic condition of each ONU. When the system load is larger than 0.8, it can be seen that the delay time increases apparently because of the greatly increasing of the burst packets. It also can be found that the delay time in predictive Q-DBA has improved by about 26% and 21% on the average over Q-DBA and DBAM, respectively.

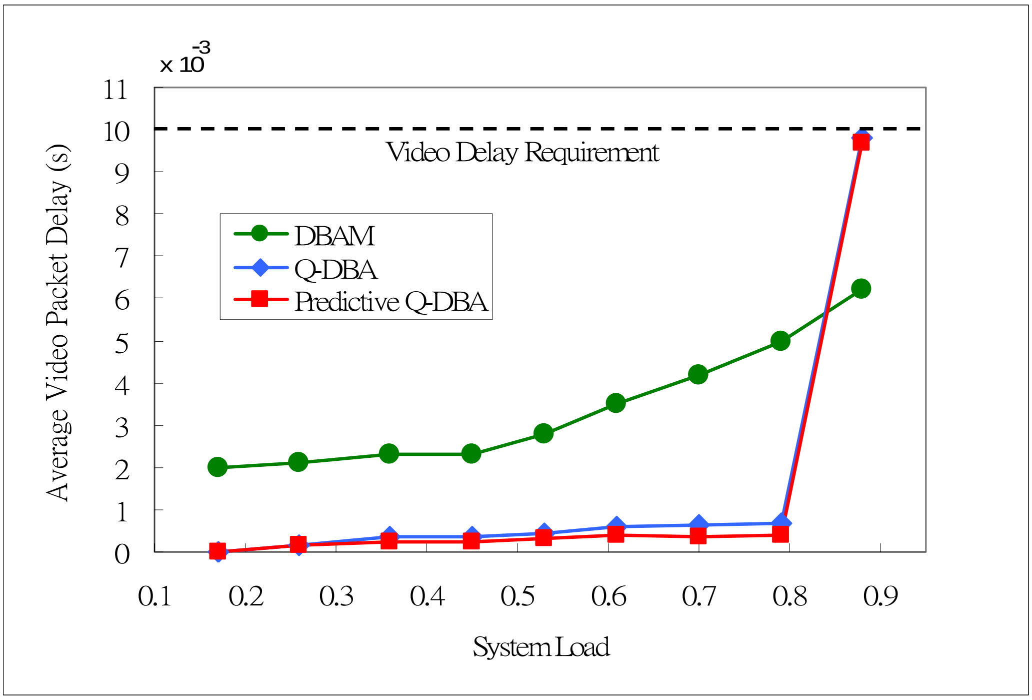


Figure 4.2: Average video delay time versus the system load in EPON

Figure 4.2 shows the average video delay time versus the system load in EPON. It can be found that, when the system load is below 0.8, the average video delay in Q-DBA and predictive Q-DBA is far from the video delay requirement because the system has enough bandwidth to allocate, so the ONUs could get more bandwidth to transmit the video packets which are not reported. But in DBAM, the average video delay increases almost smoothly with the increasing of system load because of the maximum window in DBAM, and the prediction in ONUs. Since the PRNN-based predictor is known to be good at the prediction of burst traffic, the predictive Q-DBA improves the delay time by about 29% over Q-DBA (90% better than DBAM) when the system load is below 0.8. When the system load is larger than 0.8, the packets are dropped because the capacity of fiber link is limited, so the average video delay is close to the video delay requirement. At last, it can be easily found that the video delay requirement is still satisfied and is good when the system load is below 0.8.

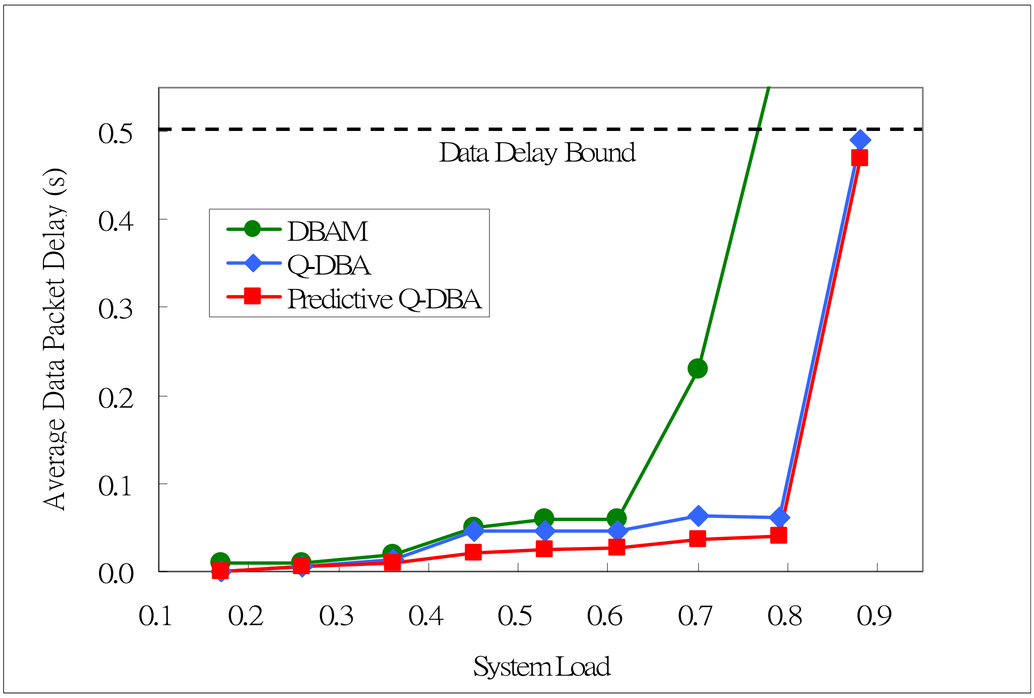


Figure 4.3: Average data delay time versus the system load in EPON

Figure 4.3 illustrates the average data delay time versus the system load in EPON. In DBAM, it can be seen that the average data delay increases with the increasing of system load, because the maximum window does not meet the real requirement of data packets, and the burst arrival cannot be served instantly. When the system load is below 0.8, the average data delay in Q-DBA and predictive Q-DBA increases almost smoothly with the increasing of system load. This is the same phenomenon as in video delay. Since both the Q-DBA and predictive Q-DBA have the prediction of the new arrival data packets, the delay time is small when the system load is below 0.8. Furthermore, due to the Q-DBA and predictive Q-DBA considers the condition of waiting bound, the data packets in Q-DBA and predictive Q-DBA does not violate the delay bound as early as that in DBAM does. For the better prediction performance of the PRNN-based predictor, the predictive Q-DBA improves the delay time about 34% than Q-DBA and 43% than DBAM. When the system load is larger than 0.8, the delay time increases apparently because of the greatly increasing of the burst packets. The reason is similar to the average video delay. As same as video delay, it also can be found that the data delay requirement is still satisfied and is good when the system load is below 0.8.

Figure 4.4 illustrates the average video dropping probability versus the system load in EPON. It can be found that the average video dropping probability of Q-DBA and predictive Q-DBA equals to zero when the system load is below 0.8. It is because in both Q-DBA and predictive Q-DBA, the priority of video packets with the problem of delay requirement will be raised. The video packets with higher priority will be served prior to the video packets with original priority. When the system load is larger than 0.8, the average video dropping probability exceeds the video dropping probability requirement because the capacity of fiber link is limited. It can also be found that the dropping probability in DBAM cannot be guaranteed. It is because the maximum window cannot totally support the burst arrival, therefore the video dropping probability is violated.

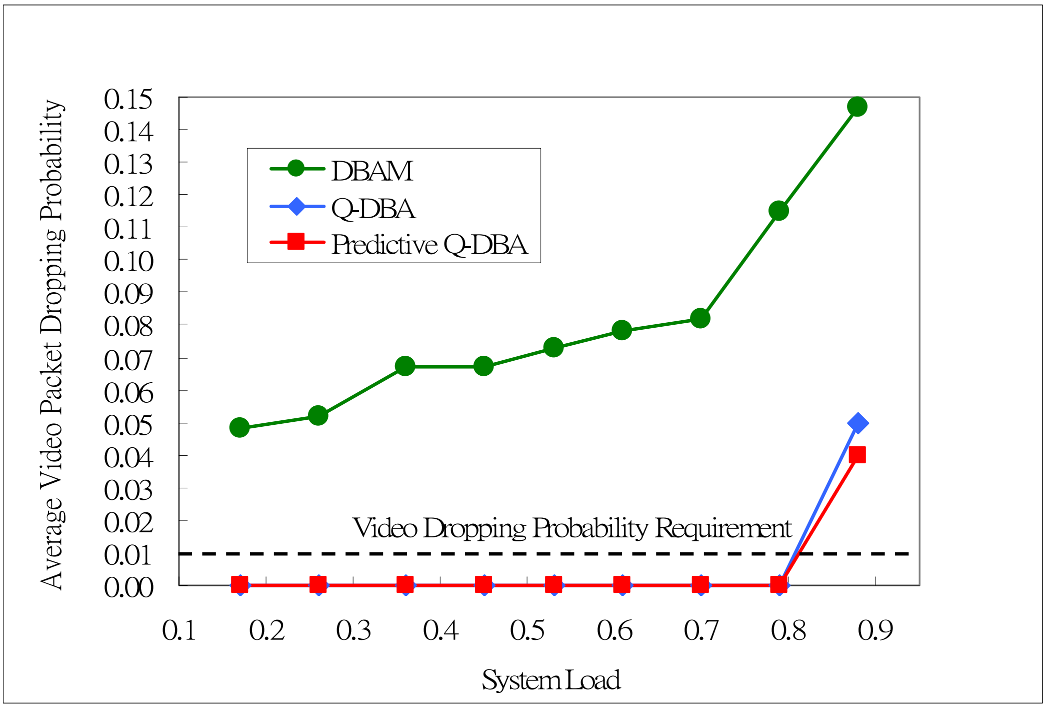


Figure 4.4: Average video dropping probability versus the system load in EPON

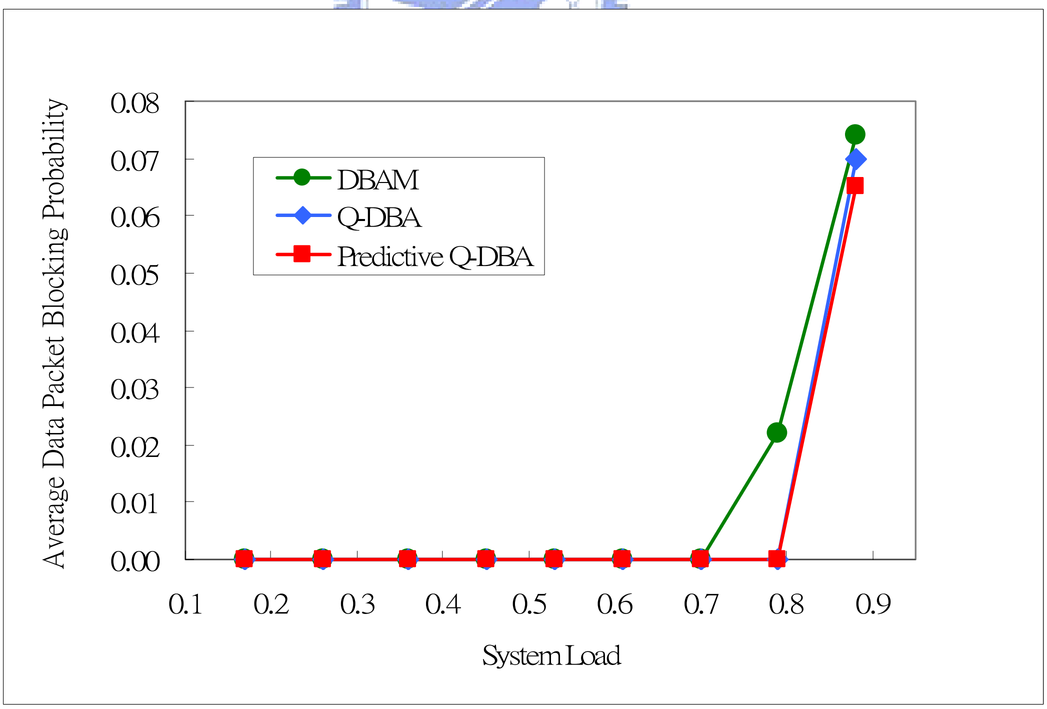


Figure 4.5: Average data packet blocking probability versus the system load in EPON

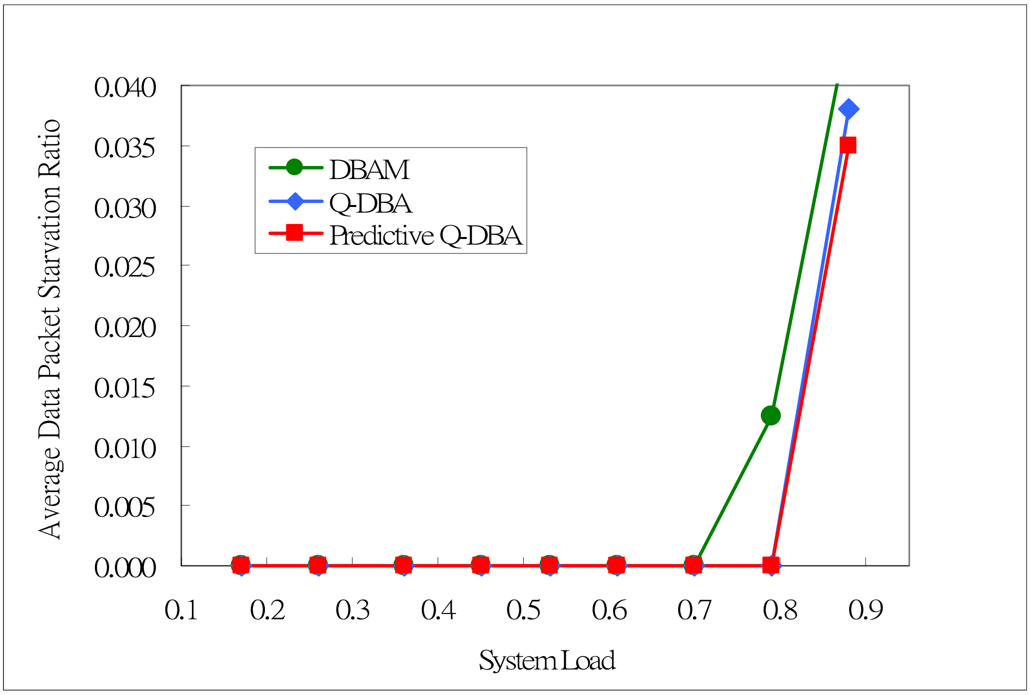


Figure 4.6: Average data starvation ratio versus the system load in EPON

Figure 4.5 illustrates the average blocking probability of data packets versus the system load in EPON. It can be found that the average blocking probability of data packets in DBAM, Q-DBA and predictive Q-DBA equals to zero when the system load is below 0.7. It is because the system has enough capacity to support the system load. When the system load is larger than 0.7, the blocking probability increases greatly. It is because the system cannot afford the burst arrival data packets. Due to no limitation of allocated bandwidth in Q-DBA, predictive Q-DBA and the consideration of waiting bound for data packets, the blocking probability in Q-DBA and predictive Q-DBA does not increase as greatly as that in DBAM.

Figure 4.6 illustrates the average starvation ratio of data packets versus the system load in EPON. The starvation ratio of data packets is defined to express the percentage of data packets whose delay time exceed the delay bound among the total transmitted data packets. It can be seen that the starvation ratio of data packets in DBAM, Q-DBA and predictive Q-DBA is zero when the system load is less than 0.7. That is, the starvation does not occur in all of them. It is because the system has enough bandwidth to support the system. When the system load is larger than 0.8, the starvation ratio goes high. It is because the data packets have lower priority than video packets. When the arrival video and data packets increase rapidly simultaneously, the video packets will be served earlier than data packets. Thus, the starvation ratio of data packets will have a conspicuous increase. However, due to the priority of data packets will be raised when considering the waiting bound in Q-DBA and predictive Q-DBA, and the data packets with a higher priority are more easily transmitted than before. Therefore, the starvation in Q-DBA and predictive Q-DBA is less than that in DBAM.

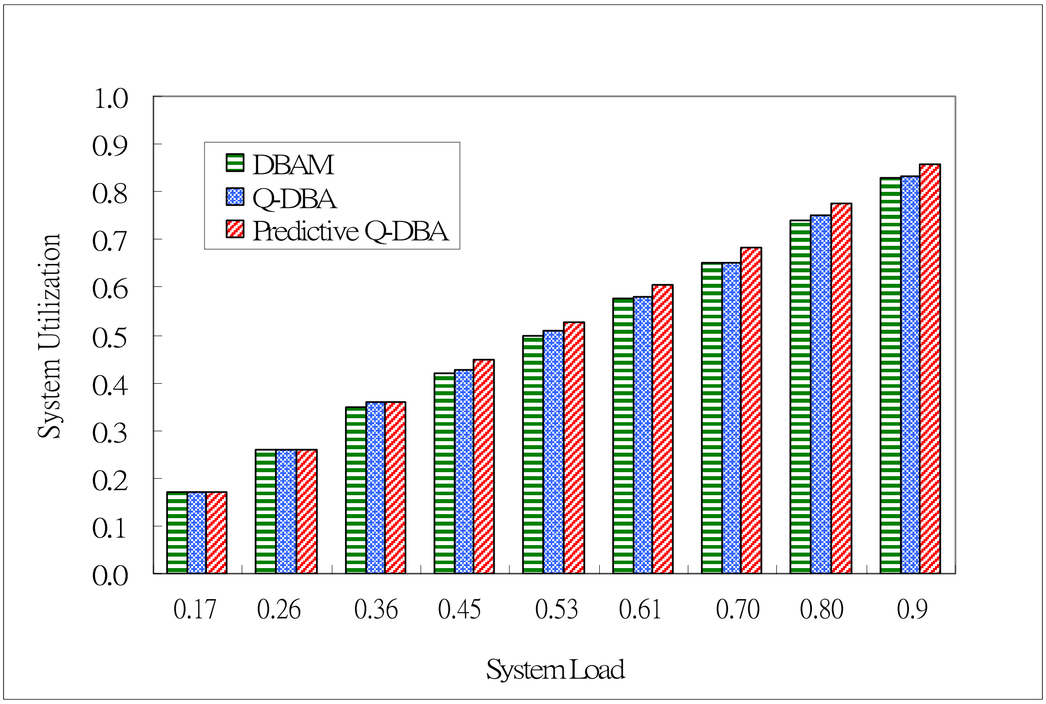


Figure 4.7: System utilization versus the system load in EPON

Figure 4.7 shows the system utilization versus the system load in EPON. It can be found that the system utilization in predictive Q-DBA is better than that in DBAM, Q-DBA by an amount of 4%, 2% on the average. Firstly, because the bandwidth in Q-DBA and predictive Q-DBA is allocated step by step to different classes rather than set a maximum window to each class in advance, so the system utilization in Q-DBA and predictive Q-DBA is better than that in DBAM. Secondly, with the increasing of system load, the available residual bandwidth decreases, rather than allocates the residual bandwidth to all classes of packets proportionally in Q-DBA, the predictive Q-DBA allocates more residual bandwidth to the high priority traffic than that in Q-DBA. That is, the packets with higher priority in predictive Q-DBA will get more bandwidth to transmit. Therefore, the sytem utilization in predictive Q-DBA is superior to that in Q-DBA method.

Chapter 5

Conclusion

In this thesis, we propose a predictive Q-DBA method which introduced a PRNN-based predictor. Owing to the characteristics of accurate and fast convergence, the PRNN-based predictor is known to be good for the prediction of the burst traffic and the ability to make the decision fleetly. Three classes of packets, real-time voice, real-time video, and non-real-time data are taking into consideration. Voice packets are strictly delay sensitive, and video packets are delay sensitive. The dropping probability of voice and video packets are also concerned. The predictive Q-DBA method sets six different priorities in order to meet the QoS requirement. They are voice packets, video packets with delay and dropping problem, data packets with starvation problem, video and data packets. The OLT combines all these six reported information and the prediction of the unreported packets which arrive between two consecutive report times, therefore the OLT could allocate more bandwidth to the high priority traffic. Thus, the total performance will improve in a certain degree, especially when the system load is high.

The performance of the proposed predictive Q-DBA is compared to DBAM [16] and Q-DBA [9]. The simulation results show that the performance of predictive Q-DBA is better than DBAM and Q-DBA in many respects. The predictive Q-DBA improves the average voice delay time by an amount of 26% and 21% over Q-DBA and DBAM, respectively. It also improves the average video delay time by an amount of 29% and 90% over Q-DBA and DBAM, respectively. As for the data service, the predictive Q-DBA improves the average data delay time about 34% than Q-DBA and 43% than DBAM. The system utilization in predictive Q-DBA is better than that in DBAM, Q-DBA by an amount of 4% and 2% on the average. As for the video dropping probability, data blocking probability, and data starvation ratio, the predictive Q-DBA is better than DBAM and almost the same with Q-DBA.

In this thesis, the predictive Q-DBA can support most of the system load without violating the QoS requirements. If the proposed predictive Q-DBA method could be adopted to manage the resource in EPON, the customers will get more benefits.

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Vita

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