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Dynamic bandwidth management with service differentiation over ethernet passive optical networks

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ABSTRACT

DYNAMIC BANDWIDTH MANAGEMENT WITH SERVICE DIFFERENTIATION OVER ETHERNET PASSIVE OPTICAL NETWORKS

by
Yuanqiu Luo

Ethernet passive optical networks (EPONs) address the first mile of the communication infrastructure between the service provider central offices and the customer sites. As a low-cost, high speed technology, EPONs are deemed as the solution to the bottleneck problem of the broadband access network.

A major feature of EPONs is the utility of a shared upstream channel among the end users. Only a single optical network unit (ONU) may transmit during a timeslot to avoid data collisions. In order to provide diverse quality of service (QoS), the bandwidth management of the upstream channel is essential for the successful implementation of EPONs, and thus, an efficient medium access control is required to facilitate statistical multiplexing among local traffics.

This dissertation addresses the upstream bandwidth allocation over EPONs. An efficient mechanism, *i.e.*, *limited sharing with traffic prediction* (LSTP), has been proposed to arbitrate the upstream bandwidth among ONUs. The MultiPoint Control Protocol (MPCP) messages, which are stipulated by the IEEE 802.3ah Ethernet in the First Mile (EFM) Task Force, are adopted by LSTP to facilitate the dynamic bandwidth negotiation between an ONU and the OLT. The bandwidth requirement of an ONU includes the already enqueued frames and the predicted incoming frames during the waiting time. The OLT arbitrates the bandwidth assignment based on the queue status report from an ONU, the traffic prediction, and the agreed service contract.

With respect to the performance evaluation, theoretical analysis on the frame loss, the frame delay, and the queue length has been conducted. The quantitative

results demonstrate that 1) the innovative LSTP mechanism dynamically allocates the upstream bandwidth among multiple ONUs; 2) the traffic predictor at the OLT delivers satisfactory prediction for the bursty self-similar traffic, and thereby, contributing to the reduction of frame loss, frame delay, and queue length; and 3) the bandwidth arbitration at the OLT effectively restricts the aggressive bandwidth competition among ONUs by adopting the service level agreement (SLA) parameter as the upper bound. Aside from analysis, the LSTP mechanism has been substantiated by experimental simulations.

In order to differentiate the service provisioning among diverse users, LSTP is further enhanced with the support of dynamic bandwidth negotiation based on multiple queues. The incoming traffics are first classified into three classes, and then enqueued into the corresponding queues. A traffic predictor is dedicated to one class of traffic from an ONU. Service differentiation among classes are provided by the combination of queuing and scheduling at the ONU side. At the OLT side, the bandwidth allocation for each class of traffic is based on the reported queue status and the traffic prediction, and is upper-bounded by the SLA parameter. Experimental simulations have justified the feasibility of providing service differentiation over the broadband EPONs.

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by
Yuanqiu Luo

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APPROVAL PAGE

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CHAPTER 1

INTRODUCTION

The access network is part of a carrier network that connects subscribers to the service provider central office over the public ground [1]. With the expansion of services offered in the Internet, a dramatic increase of bandwidth has been facilitated in the backbone network through the use of wavelength division multiplexing (WDM), providing tens of Gbps per wavelength [2]. At the same time, the local area networks (LANs) have been scaled up from 10 Mbps to 100 Mbps and are being upgraded to the gigabit speed [3]. The access network in between, however, only runs at submegabit or even kilobit of bandwidth. The tremendous growth of Internet traffic has accentuated the growing gap between the capacities of the backbone and local networks, and the serious bottleneck of the much lower capacities of the access networks in between. Such a mismatch is the so-called "last mile" problem from the service provider's point of view, or the "first mile" problem from the end users' perspective, and it calls for the upgrading of the current access network with a low-cost and high-speed solution to provide broadband access services.

Ethernet passive optical networks (EPONs) [4] address the first mile of the communication infrastructure between the service provider central office and the customer sites. As an inexpensive, simple, and scalable technology, and with the capability of delivering integrated services, EPONs were deliberated in the standardization process of the IEEE 802.3ah Ethernet in the First Mile (EFM) Task Force [5], which aims to significantly increase the broadband service performance while minimizing equipment, operation, and maintenance costs [6]. In June 2004, the EPON technology was ratified as the IEEE 802.3ah standard [7].
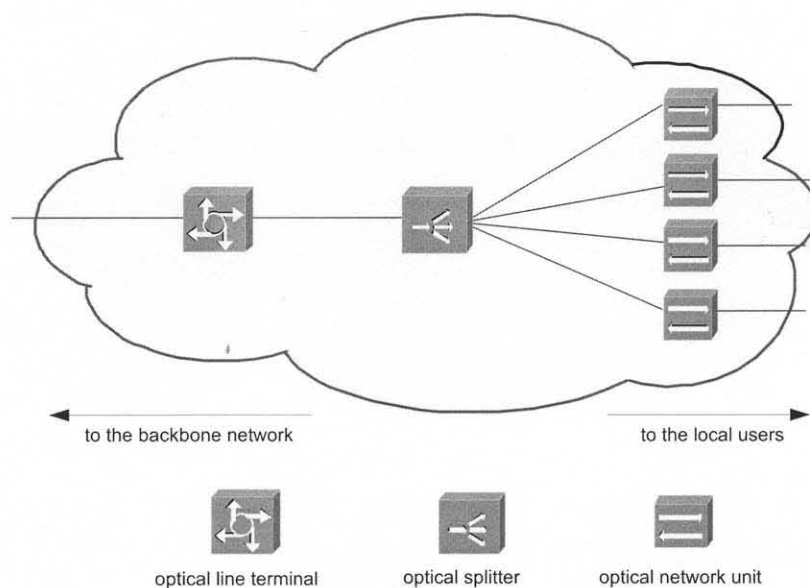
Figure 1.1 An Ethernet passive optical network.

Typically, an EPON consists of an optical line terminal (OLT) located at the provider central office and a set of associated optical network units (ONUs) that deliver broadband voice, data, and video services to the end users. The optical distribution network (ODN) comprises of fibers with a passive optical splitter [8] lying between each OLT and its associated ONUs. As shown in Fig. 1.1, a single fiber extends from an OLT to a 1:n passive optical splitter. The splitter fans out to multiple single fiber drops, which are connected to different ONUs [9]. EPONs eliminate the active electronic components such as regenerators and amplifiers in ODN, and replace them with the less expensive passive optical splitters, which are simpler and easier to maintain. With data encapsulated in IEEE 802.3 Ethernet frames, EPONs rely on the ubiquitous Ethernet technology, which is inexpensive and interoperable with legacy equipments.

As compared to the current access network technologies, such as digital subscriber line (DSL) and cable modem, EPONs have the following advantages:

- EPONs eliminate the necessity of installing multiplexers and demultiplexers in the OLT and ONUs.

- EPONs allow for longer distance between the service provider central office and the customer sites.

- EPONs minimize fiber deployment and provide higher bandwidth.

The utility of a shared upstream channel among the end users is a major feature of EPONs. In the downstream transmission, data are broadcasted from the OLT to each ONU using the entire bandwidth of the downstream channel, and all of the downstream data are carried in one wavelength. ONUs selectively receive frames destined to themselves by matching the addresses in the Ethernet frames. The broadcasting nature of Ethernet perfectly matches the EPON downstream transmission, and the "broadcast and select" architecture allows downstream multimedia services like video broadcasting.

The process of transporting data downstream to the customer sites over EPONs is different from that of transporting data upstream to the OLT. In the upstream direction, multiple ONUs share the common upstream channel, and another wavelength is employed for the upstream traffic. Only a single ONU may transmit during a timeslot in order to avoid data collisions. Because of the directional nature of the passive optical splitter, each ONU transmits directly to the OLT, but not to other ONUs. An ONU buffers the frames from its end users until its timeslot arrives. The buffered frames would be "bursted" out to the OLT in the exclusively assigned timeslot at the full channel speed.

In order to provide diverse quality of service (QoS), the bandwidth management of the upstream channel is a critical issue for the successful implementation of EPONs. The intriguing questions are: "How to manage the shared upstream channel in order to improve the bandwidth efficiency?", and "How to differentiate the services over

EPONs through medium access control?". These questions are the starting points of this dissertation.

1.1 Motivation

Access networks are cost sensitive, and thereby, over-provisioning like the backbone networks is not allowed. Towards the end of service provisioning, efficient bandwidth management of the upstream channel is desperately essential for the successful implementation of EPONs. Furthermore, from the perspective of service providers, there is quite a need for a mechanism that enables service differentiation according to different requirements. To meet these requirements, the objectives of this dissertation are:

- Creating a mechanism that enables the dynamic bandwidth sharing among multiple ONUs.

- Providing service differentiation for diverse traffics.

- Improving the overall QoS of the EPON-based access network.

1.2 Studying Scope

The efficient upstream bandwidth allocation and the service differentiation are the focus of this dissertation. From this point, a mechanism will be proposed to allocate the upstream bandwidth based on the traffic dynamics. Further, the proposed mechanism will be theoretically analyzed, and extended by considering the following desirable properties:

- Bandwidth efficiency

- Data loss control

- Delay reduction

- Queue management

- Service differentiation

1.3   Contributions

The main contributions of this dissertation are summarized as follows:

- With respect to the upstream bandwidth allocation over the EPON-based access network, the *limited sharing with traffic prediction* (LSTP) mechanism is proposed. Based on the network traffic dynamics, LSTP has the following properties: 1) compatible with the IEEE 802.3ah standard by adopting the MultiPoint Control Protocol (MPCP) messages; 2) negotiate the bandwidth requirements on-line through the REPORT/GATE process; and 3) improve the upstream channel efficiency by employing traffic prediction and dynamic bandwidth allocation.

- With respect to the LSTP performance, theoretical investigation on the data loss, data delay, and queue length is conducted. The analysis reveals the major determining factors, such as traffic prediction and traffic load. LSTP achieves performance enhancement over other existing EPON bandwidth allocation proposals, including less data loss, improved latency reduction, and more controllable queue size.

- With respect to the service differentiation of broadband access, LSTP is enhanced to accommodate diverse traffics. The properties include: 1) the incoming traffics at an ONU are classified into three classes with different QoS requirements; 2) the buffer at an ONU is managed according to the priority queuing policy, and different traffics have different precedences to access the shared upstream channel; and 3) the bandwidth is arbitrated by the OLT, and

the allocation decision is based on the bandwidth requirement and the service level agreement (SLA).

1.4 Organization

The rest of this dissertation is organized as follows:

An overview of the EPON features and related research issues are presented in Chapter 2. The first part reviews the access technologies, particularly, the EPON technology and its major features. The issue of upstream bandwidth allocation is discussed in the second part. The typical proposals are introduced as the starting points of this dissertation. Finally, the issue of service differentiation over EPONs is reviewed in the third part.

In Chapter 3, the LSTP mechanism is proposed as an efficient solution to the upstream bandwidth allocation. The MPCP messages are first introduced with the focus on the utility of the REPORT and GATE message for bandwidth negotiation. In the second part, the bandwidth requirement accuracy is improved by the implementation of traffic predictors. The bandwidth requirement for the next timeslot and bandwidth arbitration are presented in the third and fourth part, respectively. The overall operation of LSTP is presented in the last part of Chapter 3.

The performance of the LSTP mechanism is investigated in Chapter 4. The accuracy of the traffic prediction is discussed in the first part for data loss evaluation. The frame delay metric is analyzed in the second part with the focus on the LSTP delay reduction over other proposals. The queue length control issue is theoretically analyzed to verify the feasibility of LSTP. Performance comparison with existing proposals is conducted through experimental simulations in the last part.

Chapter 5 enhances the LSTP mechanism with service differentiation. In order to provide service differentiation, the local traffics at an ONU are classified according

to the DiffServ [10] framework, and the REPORT/GATE negotiation process is enhanced with the support of multiple queues. The class-based queue management and queue status report at the ONU side are investigated in the second part. The third part presents the class-based traffic prediction and bandwidth arbitration at the OLT side. The data loss is analyzed in the fourth part to verify the service differentiation model. Simulation results are demonstrated in the last part.

Finally, contributions and future work are presented in Chapter 6.

CHAPTER 2

BACKGROUND

This chapter provides an overall picture of the EPON technology along with its features. It introduces the issues of upstream bandwidth allocation and service differentiation. The pros and cons of various proposals are reviewed as a background of this dissertation.

2.1 Access Technologies

2.1.1 Digital Subscriber Line (DSL)

Telecommunications typically use twisted-pair copper wire to provide voice services to their customers. The demand for more bandwidth has resulted in the deployment of DSL equipment to provide simultaneous voice and higher speed data services. DSL modems contain an internal signal splitter that carries voice signals on the usual low frequencies (from 0 up to 4kHz) and data signals on the unused high frequencies [11], allowing simultaneous access to the wire by the telephone and the computer.

As shown in Fig. 2.1, a DSL network provides point-to-point (P2P) dedicated public network access between a service provider central office and the customer site. The term xDSL covers a number of similar yet competing forms of DSL, including
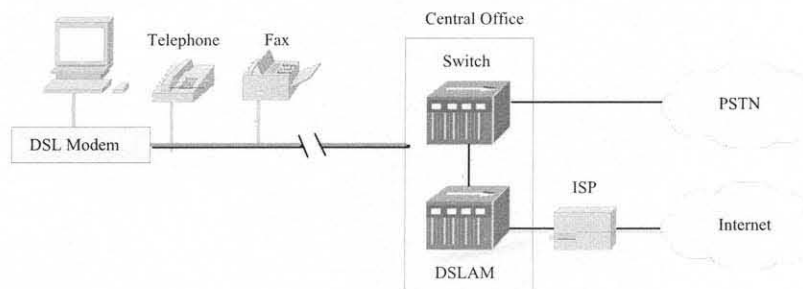
Figure 2.1 DSL network.

asymmetric DSL (ADSL), symmetric DSL (SDSL), high-bit-rate DSL (HDSL), rate-adaptive DSL (RADSL), and very-high-data-rate DSL (VDSL) [12]. DSL is distance sensitive, and the supported data rate varies depending on the transmission distance. Essentially, the longer one's telephone line runs from their house to the central office, the less performance they can achieve with DSL as compared to neighbors who might live closer to the central office. A typical ADSL system provides 8 Mbps downstream bandwidth in the wire length of 9,000 feet, or 1.5 Mbps downstream bandwidth in 18,000 feet [13]. The DSL technology draws significant attention from service providers because it delivers data services to dispersed locations with relatively small changes to the existing telecommunications infrastructure.

The major problems with sending a high frequency signal, such as DSL, over an unshielded pair of copper wires include signal fading and crosstalk. As the length of wires increases, the signal at the customer side may become too weak to be correctly detected. Yet, simply increasing the power of the signals at the central office tends to transfer the signals to the wires in the same bundle. This transferring of signals is called crosstalk, and as a result, performance will be severely impaired.

2.1.2 Cable Modem

Cable companies offer Internet access through the traditional cable TV (CATV) network. The Internet access requires two types of equipment: a cable modem on the customer end and a cable modem termination system (CMTS) at the cable provider's end [14]. The CMTS located at the cable operator's network hub is a data switching system specifically designed to route data from many cable modem users over a multiplexed network interface. It controls access to cable modems on the network. Traffic is routed from the CMTS to the backbone of a cable Internet service provider (ISP), which, in turn, connects to the Internet.
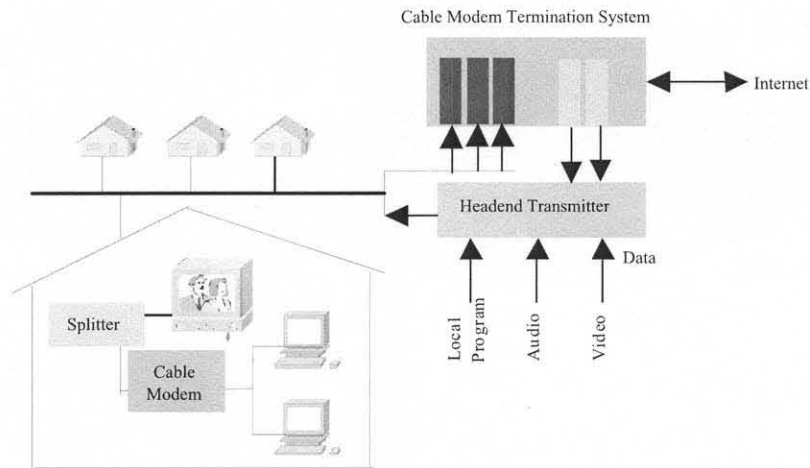
Figure 2.2 Cable modem network.

Fig. 2.2 illustrates the elements and services in the cable modem network. In the upstream, data from individual users are filtered by upstream demodulators for further processing by the CMTS. In the downstream, a cable headend combines the data channels with the video, audio, and local programs, and transmits them throughout the cable distribution network. At the user location, a one-to-two splitter separates the coaxial cable line serving the cable modem from the line that serves the TV sets.

Different from DSL, cable modem service uses a shared cable line to provide service to an entire neighborhood. Essentially, all cable customers in the region belong to the same LAN. Cable modem speeds vary widely. While cable modem technology can theoretically support up to about 30 Mbps, most providers offer service with between 1 Mbps and 6 Mbps bandwidth for downstream (from CMTS to cable modem), and bandwidth between 128 Kbps and 768 Kbps [15] for upstream (from cable modem to CMTS).

Besides the signal fading and crosstalk problems, cable modem has its own technical difficulties. The major one is that the original CATV infrastructure is designed to broadcast TV signals in just one direction - from the CATV provider

to the home end users. The Internet, however, is a two-way system where data also need to flow from the subscriber to the service provider. In order to enable the two-way transmission, the upstream channel capacity has to be significantly increased by encroaching the service provider's content bandwidth [16].

2.1.3 Passive Optical Networks (PONs)

As bandwidth demands grow beyond their unsupported level, neither DSL nor cable modem can remain successful. Besides, DSL has severe problems with respect to distance and noise limitations, and cable modem is not optimized to carry data traffic for its capacity asymmetry [17]. The recent development of optical fiber technologies, especially the maturity of integration and new packaging technologies, has enabled passive optical networks (PONs) as a promising solution for the provisioning of high bit rate at a reasonable cost.
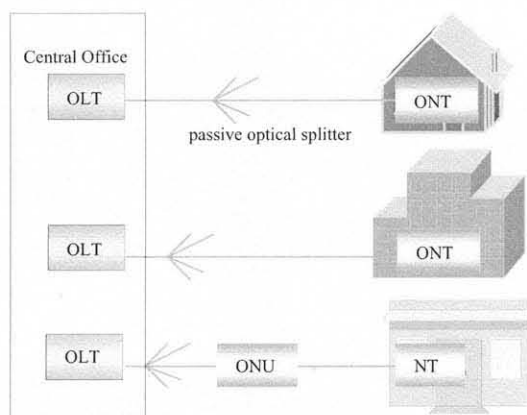
Figure 2.3 Passive optical network.

Aiming to break the bottleneck of broadband access, PONs are drawing much attention from both research communities and service providers. A typical PON, as illustrated in Fig. 2.3, consists of one OLT, which is located at the service provider central office, and n associated ONUs or optical network terminals (ONTs), which deliver data to the end users. A single fiber extends from the OLT to a 1:n

passive optical splitter, fanning out n single fiber drops to connect to the associated ONUs/ONTs. PONs are point-to-multipoint (P2MP) optical networks with no active electronic components in the signal path from the source to the destination [18].

The active components require powering, and are generally comprised of processors or memory chips to process information in the signal path. In PONs, the active components in DSL or cable modem, such as regenerators, repeaters, and amplifiers, are eliminated and replaced with the less expensive and longer lived passive optical splitters [8]. The splitter is a very simple device with no electronics, allowing the downstream traffic from the OLT and the upstream traffic to the OLT to be split from and combined onto the shared portion of the fiber. It is merely the device passing or restricting light, and has no power or processing requirements [19]. Therefore, the splitter has virtually unlimited mean time between failures (MTBF), thus lowering the overall maintenance costs for the service provider. The employment of passive optical splitters reduces the feeder fiber counts in PONs. Another advantage is that PONs can be easily upgraded by only changing electronics at both extremes of the network, while the passive network infrastructure remains the same [20, 21]. As compared to other options, PONs offer more available bandwidth to provide a broad range of services, and more reliability due to the use of optical fiber. In this way, PONs are deemed as the technology to provide cost-efficient and highly flexible access networks in the sense that a broad range of future services may be easily provisioned [22].

Data-Link Technologies in PONs The data-link technology is a critical challenge when designing a PON-based access network. Two major data-link technologies are Asynchronous Transfer Mode (ATM) and Ethernet. Accordingly, two options of PON-based access networks are ATM PONs (APONs) and Ethernet PONs (EPONs). The APON technology was introduced in the 1990s by the Full Service Access Network (FSAN) group. The International Telecommunications Union (ITU)

ratified ITU-T G.983x recommendations [23] and the name "APON" was soon broadened to "Broadband PON" (B-PON). The Ethernet PON (EPON) technology was standardized by IEEE as the IEEE 802.3ah standard [7]. EPONs adopt Ethernet frames to encapsulate the carried data, and are compatible to the ubiquitous IEEE 802.3 Ethernet standards. Gigabit-capable PONs (G-PONs) are the continuation and evolution of B-PONs, ratified by the ITU-T G.984x recommendations [24]. G-PONs supports Ethernet, ATM, and TDM traffic over the P2MP PON network topology, and describe higher line bit rates. The comparison of different "flavor" of PONs is summarized in Table 2.1.

As compare to B-PONs, EPONs are tailormade to carry the unprecedentedly growing IP traffic in today's network [25]. First, transporting IP traffic over B-PONs is quite inefficient. With the data carried in fixed 53-byte ATM cells, B-PONs have to segment the variable-length IP packets into many fixed-length and much shorter ATM cells. This excessive segmentation causes a considerable delay in the communication process. Furthermore, the so-called "ATM cell tax", *i.e.*, the 5-byte cell header, causes an onerous overhead of the transmission of IP packets over B-PONs. By contrast, EPONs encapsulate the IP packets in the Ethernet frames, with the length ranging from 64 bytes to 1518 bytes, thus reducing the time consuming segmentation process relative to B-PONs. In addition, the available upstream bandwidth in B-PONs (a maximum of 622 Mbps) is smaller than that in EPONs (1.25 Gbps). Finally, Ethernet is a widely used LAN protocol all over the network world. If Ethernet were used in the access network, it would be unnecessary to convert between protocols as required in B-PONs. Although G-PONs support up to 2.5 Gbps bandwidth, the major concern is the supporting equipment cost. The requirement to support 2.5 Gbps in the upstream direction soars up the price of both the transmitter in ONUs/ONTs and the receiver in the OLT [26].

Owing to the expansion of Internet services and the ubiquitous deployment of uncontested Ethernet standard, IP Ethernet architecture is poised to become the dominant means of delivering broadband voice, data, and video services over a single platform [27], and therefore, the research scope of this dissertation focuses on EPONs. As illustrated in Table 2.1, the critical issue of EPONs is to facilitate QoS provisioning. The following chapters study this issue from the service providers' point of view.

Table 2.1 Comparison of PONs

	B-PONs	EPONs	G-PONs
Driven by	ITU-T/FSAN	IEEE EFM	ITU-T/FSAN
Standards	G.983x	802.3ah	G.984x
Max bandwidth	622 Mbps	1.25 Gbps	2.5 Gbps
Payload	ATM	Ethernet	ATM,TDM,Ethernet
Data unit size	fixed,53 bytes	variable,64~1518 bytes	fixed or variable
Split ratio	1:32	1:16	1:128
QoS support	yes	no	yes
Hardware price	high	low	very high

Medium Access Technologies in PONs   Another design challenge in PONs is the separation of upstream channels belonging to different ONUs. Without such separation, two or more ONUs may start transmission such that their data, when reaching the optical splitter in the upstream channel, may overlap, or partially overlap each other, and data collision is inevitable. In order to mitigate this problem, a suitable multiplexing technology must be chosen to tackle the medium access among multiple ONUs.

The available medium access technologies are wavelength division multiple access (WDMA), code division multiple access (CDMA), and time division multiple access (TDMA) [28]. In WDMA, each ONU transmits its data to the OLT using a specific wavelength laser. The OLT has to have a transmitter array to support multiple ONUs, thus increasing the cost of the access network. Moreover, it is difficult to add a new ONU unless the transmitters were overprovisioned in advance. In CDMA, the inter-channel interference (ICI) increases as the number of ONUs increases. Reducing such interference calls for very complicated signal processing chips, and would add more cost to maintain a CDMA-based access network. Furthermore, in order to provide CDMA among the data from different ONUs, the network components must be able to handle signal rate much higher than the data rate, thereby soaring the access network price tremendously. By dividing the upstream channel into timeslots, TDMA allows ONUs to transmit their data in different exclusive timeslots, and achieves the granularities finer than one wavelength. Only one transmitter is needed at the OLT to multiplex the upstream data no matter how many ONUs are connected. A new ONU can be easily added by employing the PON control protocol. To support TDMA, synchronization between the OLT and the ONUs is necessary.

Without the benefits of large-scale cost sharing, access networks must strive to minimize cost. Service providers desire to fulfill the medium access with the cost as low as possible, while achieving the granularity as fine as possible and building up access networks as scalable as possible. Toward this end, TDMA and TDM were chosen by the PON standard bodies for the upstream and downstream data transmission, respectively.
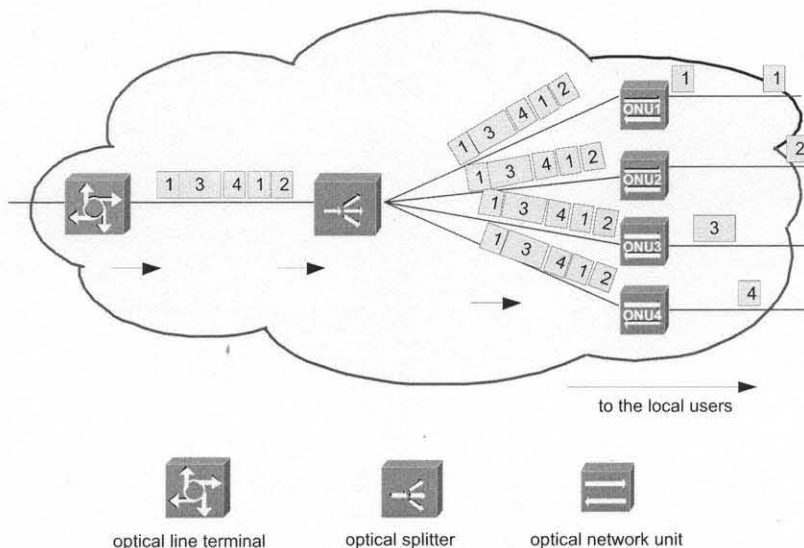
Figure 2.4 EPON downstream transmission.

2.2 Upstream Bandwidth Allocation over EPONs

The downstream data transmission to multiple ONUs is different from transporting data upstream from multiple ONUs to the associated OLT. As illustrated in Fig. 2.4, the downstream frames are broadcasted by the OLT to ONU1~ONU4. The traffic is divided into four separate signals at the splitter, carrying all of the frames. When the frames reach an ONU, the ones intended for it are extracted and delivered to the local users. For example, ONU1 receives all of the five downstream frames from the OLT while only forwards the two frames destined to it.

In the upstream direction, EPONs are a multipoint-to-point network. Multiple ONUs share a common upstream channel, and at most one ONU may transmit frames to the OLT at a particular timeslot. The utility of the shared upstream channel calls for an efficient medium access control (MAC) protocol to fairly allocating the upstream bandwidth among multiple ONUs, providing non-overlapping timeslots and preventing the collisions among Ethernet frames from different ONUs. As exemplified in Fig. 2.5, the incoming two frames from end users are first buffered at ONU4. When the assigned exclusive timeslot of ONU4 arrives, these two frames are "bursted" out

to the backbone network

optical line terminal   optical splitter   optical network unit

Figure 2.5 EPON upstream transmission.

to the OLT at the full upstream channel speed without the collisions between the frames from ONU1~ONU3. There have been numerous proposals in the literature to tackle the upstream bandwidth allocation over EPONs. The typical ones are reviewed in the rest of this section.

2.2.1 Fixed Bandwidth Allocation (FBA)

Fixed bandwidth allocation (FBA) grants each ONU a fixed timeslot length in every service cycle [28]. A service cycle is defined as the time that each ONU transmits its data once to the OLT. In FBA, the timeslot of each ONU is pre-decided and fixed no matter how fast or how slow the traffic arrives, *i.e.*,

$$b_i^g = B_i, \tag{2.1}$$

where b_i^g is the allocated bandwidth by the OLT to ONUi, and B_i is a constant.

Without the overhead of the queue status report and the grant transmission, FBA is simple to be implemented. On the other hand, without considering the on-line traffic dynamics, the major disadvantages of FBA include low bandwidth utilization,

long data delay, and heavy data loss. An ONU will occupy the upstream channel for its assigned timeslot even if there is no frame to transmit, thus resulting in the increased delay for all the Ethernet frames buffered at other ONUs. Many frames could be backlogged at the buffers while the upstream channel is lightly loaded or even idle, hence leading to underutilization of the upstream channel.

2.2.2 Limited Bandwidth Allocation (LBA)

In limited bandwidth allocation (LBA) [29,30], an ONU negotiates with the OLT on the timeslot length as

$$
b_i^g = \begin{cases} b_i^r \ , & when \ b_i^r \leq B_i^{max} \\ B_i^{max} \ , & otherwise \end{cases} , \tag{2.2}
$$

where b_i^g is the allocated bandwidth for ONUi, b_i^r is the bandwidth requested by ONUi, and B_i^{max} is the maximum timeslot length of ONUi, a parameter specified in the service level agreement (SLA). When the bandwidth requirement from an ONU is less than or equal to the limit, the OLT grants the bandwidth requirement; otherwise, B_i^{max} is granted.

LBA tracks the traffic load by means of the bandwidth requirement b_i^r, which is reported by an ONU according to the queue length. The granted timeslot length varies according to the dynamic traffic, and the timeslot length is upper-bounded by the SLA parameter. The service cycle varies because ONUs may be assigned with different timeslot length in different service cycles. The conservative feature of LBA confines each ONU by its own limit, thus restricting the aggressive competition for the upstream bandwidth. Its major disadvantages include the deferred service for the Ethernet frames arrived during the waiting time (as will be investigated in Chapter 3).

2.2.3 Excessive Bandwidth Reallocation (EBR)

In LBA, there might be some lightly loaded ONUs with the bandwidth requirements less than the SLA limits. The sum of the under-exploited bandwidth of the lightly loaded ONUs is called the excessive bandwidth b_{exc}, where

$$b_{exc} = \sum_i (B_i^{max} - b_i^r) \ , \quad i \in I, b_i^r < B_i^{max}. \tag{2.3}$$

As an extension of LBA, excessive bandwidth reallocation (EBR) [31] exploits b_{exc} by redistributing it among the heavily loaded ONUs. The heavily loaded ONUi obtains an additional bandwidth b_i^a, where

$$b_i^a = \frac{b_{exc} \times b_i^r}{\sum_i b_i^r} \ , \quad i \in \bar{I}, b_i^r > B_i^{max}, \tag{2.4}$$

and the assigned bandwidth to the heavily loaded ONUi is

$$b_i^g = B_i^{max} + b_i^a \ , \quad i \in \bar{I}, b_i^r > B_i^{max}. \tag{2.5}$$

In order to redistribute the excessive bandwidth, the OLT grants the lightly loaded ONUs instantaneously while the grants for the heavily loaded ONUs being deferred until all bandwidth reports have been collected. The drawback is that the service order of ONUs changes in every service cycle, with the heavily loaded ONUs always being served after the lightly loaded ones, and therefore, the frames at the heavily loaded ONUs suffer longer delay and heavier loss.

2.3 Service Differentiation over EPONs

Other than the upstream channel bandwidth allocation among different ONUs, a major challenge of EPONs is the provisioning of diverse QoS to support the flourishing of new applications [32]. The concept of QoS did not exist at the beginning of the Internet. According to the "first come first serve" policy, the Internet only provided the best effort (BE) service. With the expansion of the Internet, more and

more new applications are carried over the Internet, such as voice over IP (VoIP), video conferencing, and video on demand (VoD), and QoS provisioning has become a necessity.

From the customer site point of view, QoS is the service quality they experience. For different customers and different applications, QoS means different things. The main metrics to measure QoS quantitatively are delay, bandwidth, and data loss rate [33].

From the service providers' perspective, QoS-based value-added services are desperately needed to increase their revenue. To the service provider, QoS refers to the ability to provide different treatments to different traffics of different customers. The primary goal is to increase the overall utility of the network by granting priority to the higher-value or more performance-sensitive traffics. "Priority" means either lower drop probability or preferential queuing under the condition of network congestion. It should be noted that QoS does not prevent congestion or generate more bandwidth; it only adds "intelligence" that allows the network to make intelligent decisions on how to allocate the network resources.

Since Ethernet does not support QoS directly while the access network is required to accommodate various kinds of traffics, service differentiation is a distinguished feature that EPONs are expected to provide [34]. Categorizing the traffics of an ONU into different classes is a practical approach for service differentiation [35]. In the DiffServ framework [10], the high priority class is the expedited forwarding (EF), which is delay sensitive and requires bandwidth guarantees. The medium priority class is the assured forwarding (AF), which is not delay sensitive, but requires bandwidth guarantees. The low priority class is the best effort (BE), which is neither delay sensitive nor bandwidth guaranteed. The service differentiation over EPONs can be approached by means of a combination of queuing, scheduling, and class-based bandwidth allocation.

Many recent studies focus on the service differentiation over EPONs. Ma, *et al* [36] proposed the bandwidth guaranteed polling (BGP) scheme to arbitrate the upstream bandwidth. BGP divides the ONUs into two groups, with the ONUs in the first group receiving bandwidth guaranteed services, and the ONUs in the second group receiving the BE services. The major drawback is that BGP does not consider the service differentiation among the local traffics at a single ONU. In the real world, it is very difficult to decide to which group an ONU belongs, since one ONU may carry the traffics from a small business or an office building, including different applications from different users. Therefore, the service differentiation should be provided among diverse traffics instead of the ONUs.

Choi and Huh [37] investigated the provisioning of multimedia services over EPONs. The class-based bandwidth allocation is handled by collecting the REPORT messages from all ONUs before making decisions. The OLT assigns a fixed bandwidth to the EF traffics of all ONUs regardless of their dynamics. The AF requirements are granted as follows: if the sum of the AF requirements of all ONUs is less than or equal to the leftover bandwidth after having served the EF services, all AF requirements are granted; otherwise, the leftover bandwidth is equally distributed among all AF requirements. The leftover bandwidth after having served the EF and the AF traffics is distributed among all BE requirements. The major drawbacks include the fixed bandwidth allocation for the EF traffics, which penalizes the AF and the BE traffics by increasing their frame delay; and the long report collection time, which does not end until having received reports from all ONUs.

During the time of bandwidth negotiation, each ONU experiences a waiting time, which ranges from sending the queue status report to sending the buffered frames. When reporting the queue status, an ONU usually informs the number of already buffered frames to the OLT, and therefore, frames arrived during the waiting time have to be deferred even if the upstream channel is lightly loaded. This is unfair

for the frames arrived during the waiting time, since they are deferred not because of the lack of available bandwidth but the unfair bandwidth allocation mechanism. The deferred frames increase the queue size at the ONU, and will eventually result in data loss when the buffer overflows. Assi, *et al* [31] proposed a dynamic bandwidth allocation scheme by adding a credit x in the bandwidth requirement of the EF traffics. This credit is the amount of arrived frames during the waiting time of the previous service cycle. The reported EF traffic bandwidth is the sum of the EF queue length plus the credit, while the reported AF and BE traffics are the actually buffered amount. The drawback is that the service order of ONUs changes in every service cycle, with the heavily loaded ONUs always being served after the lightly loaded ones, and therefore, the advantages of the EF traffic credit are severely impaired because the waiting time of each ONU may change drastically. On the other hand, the AF and the BE traffics are the majority over EPONs. Therefore, the unfairness of bandwidth allocation is not alleviated if the arrived AF and BE frames during the waiting time are not transmitted (or partially transmitted) within the next timeslot.

Other proposals, such as the deterministic effective bandwidth (DEB) approach [38], and the decentralized architecture [39], are either incompatible with the IEEE 802.3ah standard or impractical due to high complexity and significant overhead. Most importantly, however, QoS metrics, such as data loss, delay, and queue length, have only been addressed in the above studies from the experimental aspect, and no theoretical analysis has been conducted to justify the achieved experimental performance. In the rest of this dissertation, a bandwidth management mechanism is proposed, followed by a set of theoretical analysis to testify its performance improvement. The proposed mechanism provides the following characteristics: First, it enables dynamic bandwidth negotiation by employing the control messages in MPCP, implying that it is seamlessly compatible with the IEEE 802.3ah standard. Second, on-line traffic prediction is facilitated based on network traffic self-similarity,

and data delay is thus reduced by allocating flexible timeslots dynamically. Third, the aggressive bandwidth competition among multiple ONUs is restricted by upper-bounding the allocated bandwidth to each ONU. Fourth, improved QoS provisioning is achieved by facilitating service differentiation at both of the OLT and ONU side.

2.4 Chapter Summary

This chapter illustrates the implementation of the EPON technology, highlighting its compatibility with the IEEE 802.3 Ethernet standards and its capability to support the IP traffic. Two major issues, *i.e.*, the upstream bandwidth allocation and the service differentiation, have been investigated based on the review of the studies in these fields. The discussion on their features furnishes the background of the following chapters.

CHAPTER 3

THE PROPOSED LIMITED SHARING WITH TRAFFIC
PREDICTION (LSTP) MECHANISM

In this chapter, a mechanism for the upstream bandwidth allocation over EPONs, *i.e.*, *limited sharing with traffic prediction* (LSTP) is proposed. The bandwidth negotiation in LSTP is facilitated by using the REPORT and GATE message in MPCP. The ONU reports the local queue status to the OLT. The OLT centralizes the upstream bandwidth arbitration by adopting traffic prediction and the upper-bound specified in the SLA, limiting the aggressive bandwidth competition among ONUs.

3.1 Bandwidth Negotiation

MPCP is developed by the IEEE 802.3ah Task Force to specify the mechanism between an OLT and the associated ONUs to facilitate efficient upstream data transmission. MPCP is a frame-based protocol, which introduces the following five new 64-byte MAC control messages to provide the real-time control and manipulation of data transmission.

- REGISTER_REQUEST

- REGISTER

- REGISTER_ACK

- REPORT

- GATE

Each MAC control message has a 48-bit destination address, a 48-bit source address, a 16-bit type code, a unique opcode, and a 32-bit frame check sequence.
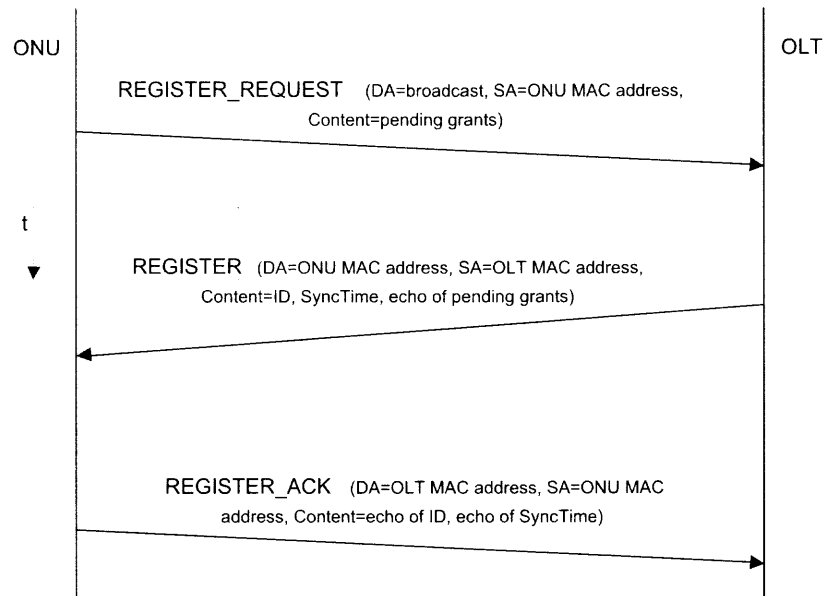
Figure 3.1 Auto-discovery of a new ONU.

REGISTER_REQUEST, REGISTER, and REGISTER_ACK messages are utilized in the auto-discovery process to harmonize a new ONU, register the ONU, assign a unique ID to the new ONU, and negotiate parameters with the new ONU. As shown in Fig. 3.1, when joining an EPON, the new ONU sends a REGISTER_REQUEST message, including its 48-bit MAC address. The OLT replies the ONU with a REGISTER message, including the MAC address of the OLT and the synchronized time. After the ONU replies the OLT with a REGISTER_ACK message, which contains the echo of the synchronized time and the MAC addresses, the registration of a new ONU is completed. The control messages for bandwidth management are REPORT and GATE.

3.1.1 REPORT Message

As illustrated in Fig. 3.2, a REPORT message contains a timestamp for synchronization and ranging between the OLT and the sending ONU. The report content

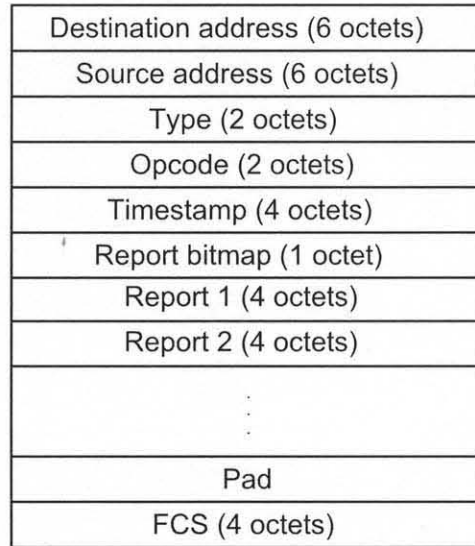| |
|---|
| Destination address (6 octets) |
| Source address (6 octets) |
| Type (2 octets) |
| Opcode (2 octets) |
| Timestamp (4 octets) |
| Report bitmap (1 octet) |
| Report 1 (4 octets) |
| Report 2 (4 octets) |
| ⋮ |
| Pad |
| FCS (4 octets) |

Figure 3.2 REPORT message format.
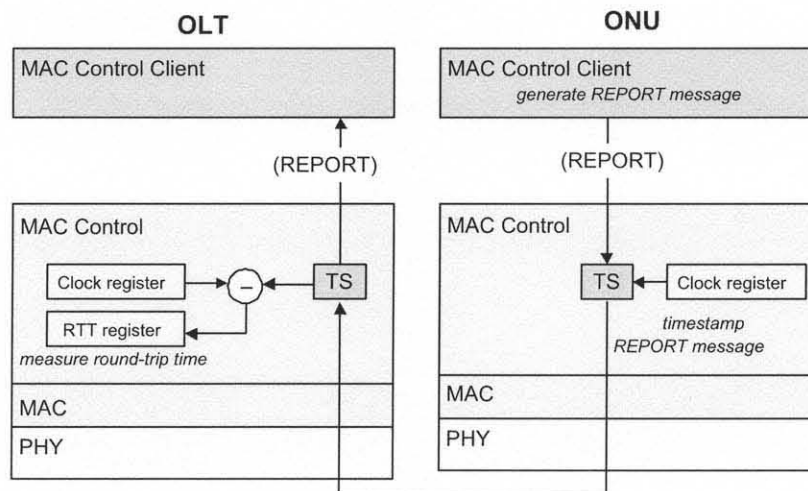
Figure 3.3 REPORT message operation.

is the local information, which is responsible for reporting the queue length at the ONU.

The operation of the REPORT message is shown in Fig. 3.3. A REPORT message is generated at the MAC control client layer of an ONU, containing the queue status at the ONU. The MAC control layer then timestamps the REPORT message according to the local clock register, and forwards the message to the associated OLT. Upon receiving the REPORT message, the OLT calculates the round-trip time (RTT) based on the reported timestamp. RTT is employed to perform ranging and synchronization.

Ranging is to support the upstream collision avoidance by finding a specific delay for every ONU. Different physical distances from the OLT to ONUs are adjusted by the ranging process, and thereby, all ONUs are placed at the same virtual distance from the OLT, and their upstream transmissions arrive at the OLT without collision. Since TDMA is adopted for the upstream channel sharing, the key issue is the establishment of exclusive timeslots for different ONUs, and thus, synchronization is critical. Good synchronization is required to support collision avoidance and ensure low data delay. The timestamp in the REPORT message is incorporated by the OLT for synchronization between the ONUs and the OLT.

The queue status content in the REPORT message is used by the OLT to make the allocation decision. As shown in Fig. 3.2, one REPORT message may contain multiple queue status. This implies that one ONU could manage several queues, which share a common physical buffer at the ONU. Most bandwidth allocation algorithms, such as LBA [29] and EBA [31], only inform the OLT about the actual queue length at the time of sending the REPORT message, ignoring the possibility that the frames arrived during the waiting time could prolong the queue size. The LSTP mechanism, which is to be proposed in the following, takes consideration of this phenomenon.

The OLT predicts the prolonged length of the queue size, and adds this prediction into the queue status report as the bandwidth requirement for the next transmission.

3.1.2 GATE Message

Destination address (6 octets)
Source address (6 octets)
Type (2 octets)
Opcode (2 octets)
Timestamp (4 octets)
Number of grants (1 octet)
Grant level (1 octet)
Grant start time (4 octets)
Grant length (2 octets)
Grant start time (4 octets)
Grant length (2 octets)
⋮
Pad
FCS (4 octets)

Figure 3.4 GATE message format.

The OLT sends a GATE message downstream to a particular ONU, containing the information of the timestamp, grant start time, and grant length. Fig. 3.4 illustrates the GATE message format. The destined ONU updates its local clock register, slot start register, and slot stop register by the received timestamp, grant start time, and grant length in the GATE message, respectively.

Fig. 3.5 shows the GATE message operation. The GATE message is generated at the OLT MAC control client layer. The bandwidth assignment decision is included in the message, with the "grant start time" indicating when to begin the data transmission, and the "grant length" indicating how long the transmission is. The GATE message is then forwarded to the MAC control layer, which timestamps the
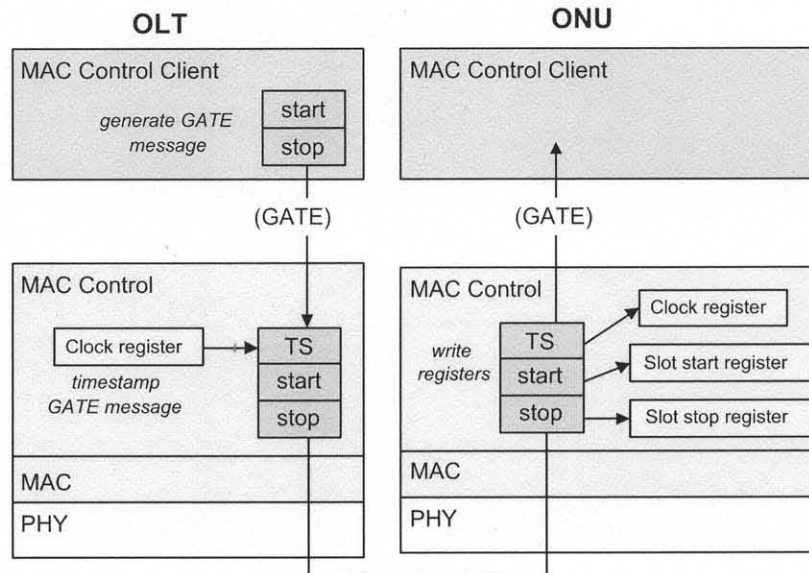
Figure 3.5 GATE message operation.

message with the content of the OLT local clock register. After receiving the GATE message, the destined ONU updates its local clock by the OLT timestamp, thus avoiding any potential clock drift. The data transmission starts once the start timer in the slot start register expires. Multiple Ethernet frames may be transmitted from the ONU to the OLT in one timeslot. The buffered frames are transmitted from the grant start time in the grant length without any contention from other ONUs. No packet fragmentation is allowed within a timeslot, and the "unfit" Ethernet frame will be deferred to the next timeslot.

3.1.3 REPORT/GATE in LSTP

Since the downstream channel is broadcasting in nature, no bandwidth negotiation is required. Each ONU filters the received Ethernet frames, and selectively forwards the ones destined to it. The OLT allocates the upstream bandwidth by deciding the start time and the length of the timeslots for all the associated ONUs. LSTP adopts the REPORT and GATE message for the upstream bandwidth negotiation. After receiving a GATE message, an ONU updates its local registers, and transmits
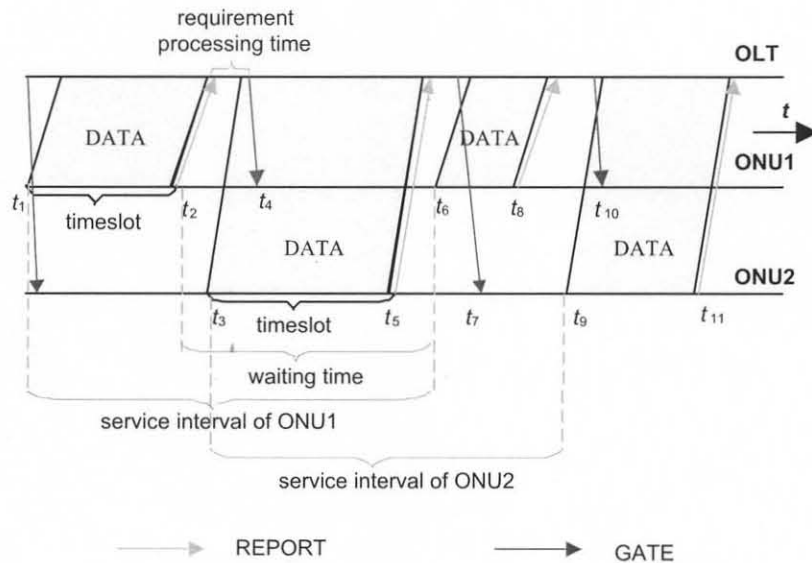
Figure 3.6 Bandwidth negotiation in LSTP.

the buffered frames in its exclusively assigned timeslot. The following are the terms adopted by LSTP.

- Service cycle: a service cycle is defined as the time duration, in which the OLT serves all of its associated ONUs once.

- Service interval: with respect to an ONU, the service interval is the time between its data transmission, *i.e.*, the time ranges from the start point of the current timeslot to the start point of the next timeslot.

- Service order: the sequence that the OLT serves all of its ONUs in a service cycle.

- Waiting time: with respect to an ONU, in a service interval, the waiting time ranges from sending its queue status to sending its buffered frames, *i.e.*, from the end point of one timeslot to the start point of the next timeslot.

Fig. 3.6 illustrates the bandwidth negotiation process in LSTP. The example EPON is composed of one OLT and two ONUs. Each ONU transmits the buffered

frames to the OLT in its exclusively assigned timeslot. An ONU piggybacks its queue status information by utilizing the REPORT message. The OLT grants the requirement by sending back a GATE message. ONU1 transmits its data to the OLT in the timeslot from t_1 to t_2, and piggybacks a REPORT message at the timeslot end. Time t_2 to time t_4 is the RTT between ONU1 and the OLT plus the bandwidth arbitration time.

The service interval of an ONU is the time between its data transmission. For example, as shown in Fig. 3.6, a service interval, say n, with respect to ONU1, ranges from time t_1 to time t_6. Service interval $(n+1)$ of ONU1 begins at time t_6, and the granted timeslot from time t_6 to time t_8 is decided on the REPORT message sent at time t_2. With respect to ONU2, service interval n begins at time t_3 and ends at time t_9. Time t_3 to time t_5 is the exclusive timeslot assigned for ONU2, and a report of its queue status is sent at time t_5. Time t_5 to time t_9 is the waiting time of ONU2 in service interval n, during which more Ethernet fames arrive at ONU2. The two consecutive service cycles are from time t_1 to time t_5 and from time t_6 to time t_{11}, respectively.

The major features of the upstream bandwidth negotiation in LSTP are the following:

1. Flexible service cycle length — The OLT serves each ONU once and only once in a service cycle. The length of the service cycle is not fixed, and it may change from time to time according to the traffic load of an EPON.

2. Fixed service order — The service order among all of the associated ONUs is fixed. If an ONU has no data to transmit, the allocated timeslot length for it is zero.

3. Piggybacked report — Each ONU piggybacks a REPORT message at the end of its data transmission in the current timeslot, indicating its local queue status.

4. Instantaneous response — After receiving a REPORT message, the OLT processes it immediately. A GATE message containing the bandwidth assignment decision is sent back to the ONU. The OLT makes the bandwidth allocation decision without collecting the queue status information from other associated ONUs.

Different from EBA [31], in LSTP, the OLT serves the ONUs in a fixed order, *e.g.*, the OLT serves the two ONUs alternately as in the example of Fig. 3.6. Such a fixed order service facilitates the upstream bandwidth efficiency, and the reason will be further verified in the following sections.

3.2 Traffic Prediction

3.2.1 Deferred Frames

During the time of bandwidth negotiation, each ONU experiences a waiting time, which ranges from sending the queue status report to sending the buffered frames. More frames will be enqueued at the buffer during the waiting time. As exemplified in Fig. 3.6, time t_2 to time t_6 is the waiting time of ONU1 in service interval n, and time t_5 to time t_9 is the waiting time of ONU2 in service interval n. Without consideration of these incoming frames when making the bandwidth arbitration decision, they cannot be transmitted in the next timeslot even if the upstream channel is lightly loaded, and have to be deferred one more service interval to be reported. The deferred frames increase the queue size at the ONU, and will eventually result in data loss when the buffer overflows.

Simulations are conducted to investigate the deferred frames arrived during the waiting time. Denote b^w as the traffic in bytes arrived during the waiting time, and b^q as the enqueued traffic in bytes when sending a REPORT message. Two ONUs and one OLT as in Fig. 3.6 are contained in the simulation EPON, and the input trace is self-similar with the Hurst parameter $H = 0.8$. Fig. 3.7 shows the ratio of b^w
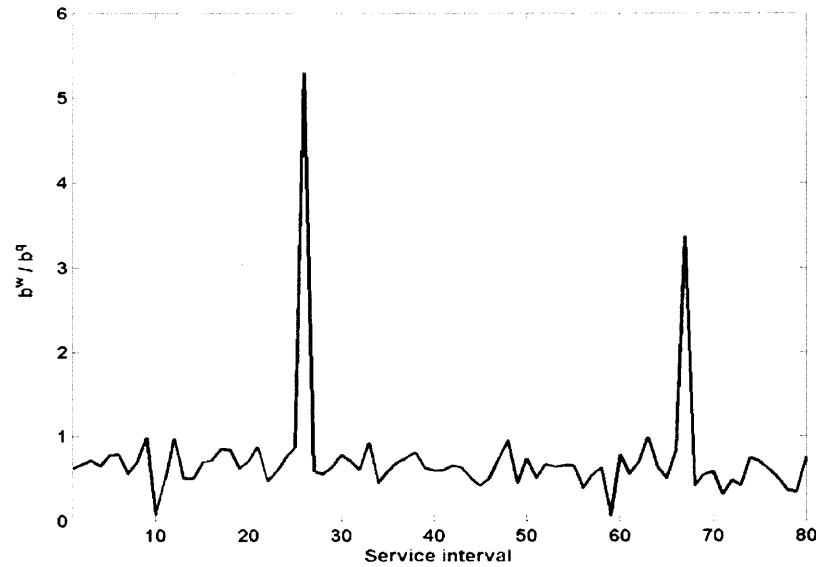
Figure 3.7 Deferred frames arrived during the waiting time.

vs. b^q, *i.e.*, the *deferral index*, at ONU1 in different service intervals. It is observed that in each service interval, data do arrive during the waiting time, and the ratio of b^w/b^q mostly falls in the range of 0.4~0.8. Without reporting these frames, around 29%~44% traffic in bytes arrived during a service interval have to be deferred for one more service interval to be reported, thus suffering extra delay. With the constraint of limited buffer size, frames have to be dropped once the accumulatively buffered frames overflow the buffer limit. For example, two bursts arrive at ONU1 during the waiting time of service interval 26 and 67, resulting in hiking the b^w/b^q ratio. It is almost impossible to hold the huge bytes of the bursts at a limited buffer for two service intervals while more frames keep arriving, and thereby, data loss due to frame dropping is inevitable.

3.2.2 Predict the Deferred Frames

In order to alleviate the extra delay experienced by the deferred data, the intuitive idea based on the aforementioned observation is that, rather than delivering the reported data, the bandwidth arbitration at the OLT should consider the incoming

data arrived during the waiting time, and thereby, minimize their impact on the data delay and loss. Towards this end, LSTP embeds traffic predictors at the OLT to take into consideration of the incoming frames arrived during the waiting time. At the end of an upstream transmission timeslot, a REPORT message is piggybacked, indicating the already enqueued data in the transmitting ONU's physical buffer. After receiving the REPORT message, the embedded traffic predictor in the OLT conducts the proper prediction procedure, adding a prediction in the ONU bandwidth requirement. The notations adopted by LSTP are listed in Table 3.1.

Table 3.1 Notations in LSTP

$b_i^q(n)$	The enqueued traffic in bytes at ONUi in service interval n.
$b_i^w(n)$	The arrived traffic in bytes at ONUi during the waiting time of service interval n.
$\widehat{b_i^w}(n)$	The predicted traffic in bytes at ONUi during the waiting time of service interval n.
$b_i^r(n)$	The reported bandwidth requirement at ONUi for service interval n.
$b_i^g(n)$	The granted bandwidth to ONUi for service interval n.
L	The order of the predictor.
$\alpha_{i,k}(n)$	The kth weight factor of the traffic predictor at ONUi in service interval n, $k \in \{0, 1, ..., L-1\}$.
$\mu_i(n)$	The step size of the predictor at ONUi in service interval n.
$e_i(n)$	The prediction error in bytes of service interval n at ONUi.
B_i	The maximum timeslot length in bytes of ONUi.

Once a REPORT message from ONUi is received, the OLT predicts the incoming traffic in bytes arrived during the waiting time at ONUi based on

information obtained in several previous service intervals. The intuition behind this prediction is the network traffic self-similarity [40], which indicates that the actual network traffic exhibits long-rang dependence (LRD). The traffic self-similarity implies that the burstiness of the traffic does not decrease with the time scale from which the traffic is observed or with the amount of multiplexing that occurs at a node [41]. Owing to the self-similarity, the correlation in network traffic does not decay rapidly, and traffic is correlated between timeslots. The theoretical consequences of self-similar traffic include larger queue size, greater data loss, and longer data delay in the network [42].

An immediate consequence of the study on network traffic self-similarity is the demonstration of the limitations of the conventional resource allocation methods [43, 44, 45, 46]. Hence, optimal allocation of network resources in order to smooth the bursty traffic is a major subject in the field of networking, and incorporating the characteristics of self-similarity into resource management is necessary to improve the overall network performance. An efficient way is to predict the incoming traffic and pre-reserve the network resource. With the advantages of low computational complexity, fast convergence, and no prior knowledge of the traffic statistical charac-teristics, linear predictor (LP) is deemed as a practical tool to conduct the on-line traffic prediction [47]. The following LP is adopted by LSTP to predict the traffic in bytes arrived during the waiting time.

$$\widehat{b_i^w}(n+1) = \sum_{k=0}^{L-1} \alpha_{i,k}(n) b_i^w(n-k). \qquad (3.1)$$

In Eq. (3.1), the output predicted quantity $\widehat{b_i^w}$ is a linear function of the observations b_i^w in previous service intervals. The weight factor of the predictor, i.e., $\alpha_{i,k}(n)$, indicates the effect of b_i^w on the output predicted result. It is determined by the actual traffic pattern, and adjustablely updated by the least mean squares

(LMS) algorithm as [48]

$$\alpha_{i,k}(n+1) = \alpha_{i,k}(n) + \mu_i(n)\frac{e_i(n)}{b_i^w(n)}, \tag{3.2}$$

where $e_i(n)$ is the prediction error of service interval n, and is defined as

$$e_i(n) = b_i^w(n) - \widehat{b_i^w}(n). \tag{3.3}$$

The update of the weight factor in LSTP is an adaptive process, which makes the bandwidth prediction in Eq. (3.1) performs satisfactorily in the environment where the complete knowledge of the incoming traffic statistics is not available [49]. The predicted incoming traffic in bytes, $i.e.$, $\widehat{b_i^w}(n)$, if optimal, should be equal to the actually arrived traffic in bytes during the waiting time, $i.e.$, $b_i^w(n)$. Owing to the imperfection of the predictor, the predicted results may turn out to be smaller or larger than the actual ones. The prediction error in Eq. (3.3) is thus employed to adaptively adjust the weight factor $\alpha_{i,k}(n)$, with the purpose to improve the prediction accuracy.

The computational complexity for the bandwidth prediction is $O(L)$. One would assume that a larger L produces better prediction because of the slow decay of network traffic correlation. The interesting thing is that only a short history of past data is enough to predict the traffic in bytes arrived during the waiting time [50, 51]. Simulations are conducted to compare the performance among different order LPs in the LSTP mechanism. The inverse signal to noise ratio listed in Eq. (3.4) is the assessment criterion for comparison.

$$SNR_i^{-1} = \sum_n e_i^2(n) / \sum_n b_i^{w2}(n). \tag{3.4}$$

In Table 3.2, H is the Hurst parameter of the self-similar trace, and LP with order 4 generates the very similar SNR^{-1} as that of the higher order LPs. With respect to the same traffic trace, increasing the LP order yields no SNR^{-1} improvement, implying

that the prediction performance in LSTP is not sensitive to the predictor order. Such an insensitivity could be attributed to the fact that, as formulated in Eq. (3.2), LMS emphasizes the most recent data. The prediction error and the actually arrived traffic of the latest service interval play the major role when adaptively adjusting the weight factor. In the rest of this dissertation, without specification, LP with order 4 is employed.

Table 3.2 SNR^{-1} of Different Linear Predictors

Linear predictor order	H=0.7	H=0.8	H=0.9
4	0.0875	0.2164	0.3304
6	0.0881	0.2185	0.3312
8	0.0884	0.2193	0.3324
10	0.0891	0.2198	0.3335
12	0.0893	0.2207	0.3339

In LSTP, the OLT works as the central controller, and the traffic prediction is done at the OLT side. This is because that, in an EPON system, the cost of ONUs constitutes roughly 80% of the deployment cost regardless of the scenario [52]. Therefore, shifting the prediction function to the OLT side facilitates centralized network management and EPON cost reduction.

3.3 Operation of LSTP

3.3.1 System Model of an ONU

A physical buffer residing at an ONU is assumed to support multiple customers. As shown in Fig. 3.8, the incoming local traffics are enqueued in the shared buffer. The traditional first-in-first-out (FIFO) queuing with drop-tail is employed. When the buffer overflows, the incoming frames are dropped by the dropping module.
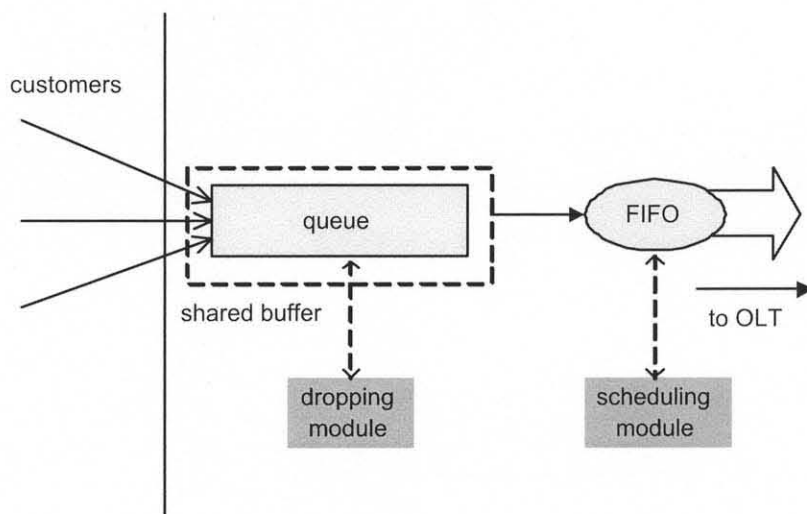
Figure 3.8 System model of an ONU.

The data transmission flow at an ONU is listed in Fig. 3.9. Data transmission at an ONU starts when the slot start timer times out, and it ends when the slot stop timer expires. A REPORT message is piggybacked at the end of each transmission timeslot, indicating the already enqueued data. The two timers are set by the decision information contained in the received GATE message.

3.3.2 Bandwidth Requirement and Bandwidth Arbitration at the OLT

When a REPORT message from ONUi is received, the OLT calculates the bandwidth requirement of ONUi for the next transmission as the sum of the reported queue length and the prediction, *i.e.*,

$$b_i^r(n+1) = b_i^q(n) + \widehat{b_i^w}(n). \tag{3.5}$$

The OLT instantaneously makes the bandwidth allocation decision after having calculated the bandwidth requirement. The granted bandwidth to ONUi for service interval $n+1$ is

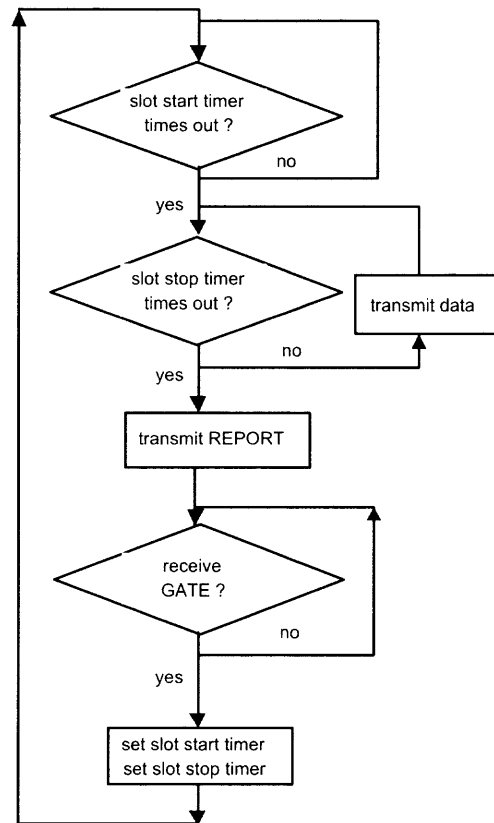$$b_i^g(n+1) = min\left\{b_i^r(n+1), B_i\right\}, \tag{3.6}$$

Figure 3.9 Data transmission flow at an ONU.

where B_i is the maximum timeslot length in bytes of ONUi, a parameter specified in the SLA.

The bandwidth allocated to ONUi is upper-bounded by the smaller value of the bandwidth requirement $b_i^r(n+1)$, which is the sum of the reported queue length and the prediction, and the maximum timeslot length B_i, which is specified in the contract between the service provider and the customer.

When the bandwidth requirement is no more than the maximum timeslot length, an ONU is called "underloaded". The assigned bandwidth to an underloaded ONU dynamically changes upon the on-line traffic. A portion of the upstream bandwidth is pre-reserved to transmit the traffic arrived during the waiting time, thus dramatically alleviating the frame deferral phenomenon. When the bandwidth requirement is more than the maximum timeslot length, an ONU is called "overloaded". In this case, the ONU violates the agreed SLA. Therefore, B_i is employed as an upper-bound, limiting the aggressive competition for the upstream bandwidth and ensuring data transmission of the underloaded ONUs. Fig. 3.10 illustrates the transmission process at the OLT.

On receiving the GATE message from the OLT, the ONU updates its local clock, and programs the local registers with the "grant start time" and the "grant length" values. When its dedicated timeslot comes, the ONU bursts out its frames to the OLT without the contention from other ONUs.

In the optimal case, when the actually incoming traffic is equal to the predicted result, and the bandwidth requirement is granted, the traffic arrived during current service interval could be transmitted from the ONU to the OLT, and no frames would be deferred for one more service interval. When the actual traffic is less than the prediction, the assigned timeslot is long enough for the enqueued traffic, and the prediction is also deemed a success. If the actual traffic exceeds the prediction, the assigned timeslot can only transmit part of the enqueued frames, and the leftover
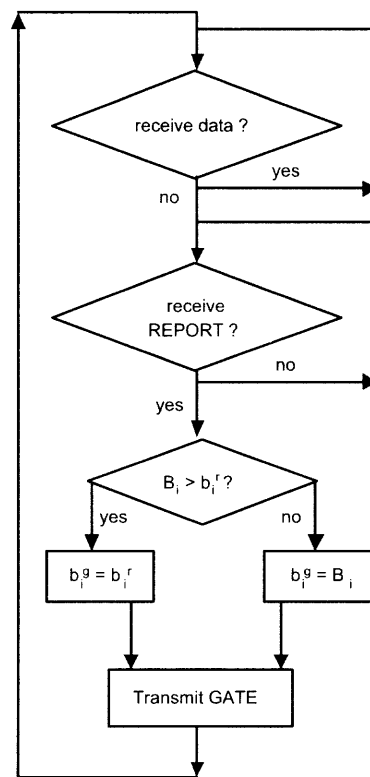
Figure 3.10 Data transmission flow at the OLT.

ones have to wait for one more service interval. The prediction fails in the last case. The prediction success probability and its impact on the network performance will be theoretically analyzed in the next chapter.

3.4   Chapter Summary

This chapter presents the proposed LSTP mechanism, which tackles the issue of upstream bandwidth allocation over EPONs. With respect to the bandwidth negotiation, LSTP adopts the REPORT and GATE message to facilitate the dynamic bandwidth negotiation between ONUs and the OLT. With respect to reducing delay of the deferred frames arrived during the waiting time of a service interval, predictors are employed at the OLT to forecast the traffic arrived during the waiting time, and a portion of the upstream channel bandwidth will be pre-reserved to facilitate data delivery. With respect to limiting the aggressive bandwidth competition among ONUs, the OLT assigns the upstream bandwidth among ONUs by employing the SLA parameter as the upper-bound. By adding one low-order predictor for each ONU, LSTP effectively curbs the unfair delay of the deferred frames, and improves the performance of EPONs.

CHAPTER 4

LSTP PERFORMANCE ANALYSIS

Continuing from the previous work, this chapter theoretically analyzes the performance of LSTP, including data loss, delay, and queue length. The key factors are the success probability of traffic prediction and the network traffic load. Simulations and discussions are then applied to LSTP and available proposals, examining the performance improvement under LSTP.

4.1 Frame Loss

In this section the performance of LSTP in terms of frame loss is analyzed. There is no frame loss due to data collisions once LSTP is employed, since all of the ONUs are allocated their exclusive timeslots. Frame loss occurs when the buffer of an ONU is full while more frames are arriving. Because of the limited physical buffer size, such incoming frames have to be dropped, resulting in frame loss. Table 4.1 lists the notations used to investigate the performance of LSTP. For notational simplicity, the referencing of the service interval is omitted in the following analysis.

4.1.1 Success Probability of Prediction

The prediction error plays a key factor on the network performance. In LSTP, the OLT conducts traffic prediction for all ONUs. The traffic prediction for an ONU fails if the frame length sum of the actually arrived traffic during the waiting time is larger than the predicted one, $i.e.$, $e_i = b_i^w - \widehat{b_i^w} > 0$. Otherwise, the traffic prediction for the ONU succeeds.

There are two subcases of the successful prediction: 1) the frame length sum of the actually arrived traffic is equal to the predicted one, and 2) the frame length sum

43

of the actually arrived traffic is less than the prediction. The incoming traffic during the waiting time could be delivered in the next timeslot in both of the subcases, given that the requirement is less than or equal to the SLA parameter. Traffic prediction is called successful if $e_i = b_i^w - \widehat{b_i^w} \leq 0$. Therefore, the success probability of traffic prediction is

$$P_i^s = P\left\{e_i \leq 0\right\}. \tag{4.1}$$

Table 4.1 Notations Used in LSTP Performance Analysis

P_i^s	Probability that traffic prediction succeeds for ONUi.
P_i^f	Probability that traffic prediction fails for ONUi.
e_i	Prediction error in bytes at ONUi with mean m_i and variance σ_i^2.
P_i^{loss}	Frame loss probability of ONUi.
C_i	Fixed buffer size in bytes of ONUi.
t_{int}	Average service interval length of ONUi.
D_i	Average delay of the deferred frames at ONUi.
β	Delay reduction index.
Z_i	Average queue length in bytes of ONUi.

Fig. 4.1 shows the simulation of the mean-squared prediction error. In an EPON with one OLT and 16 ONUs, LSTP is implemented. The step size is set as $\mu_i = \frac{L}{\sum_{k=0}^{L-1} \left[b_i^w(n-k)\right]^2}$ [48]. Each ONU has a finite buffer of 20 Mbytes, and both of the downstream and upstream channels support 1.25 Gbps transmission speed. The incoming traffic is self-similar with the Hurst parameter of 0.8. The normalized mean-squared prediction error at ONU1 is illustrated in Fig. 4.1, and other ONUs exhibit the similar result. The mean-squared error converges after the first several

service intervals, indicating that the LMS-based LP in LSTP performs well in tracking the self-similar traffic.
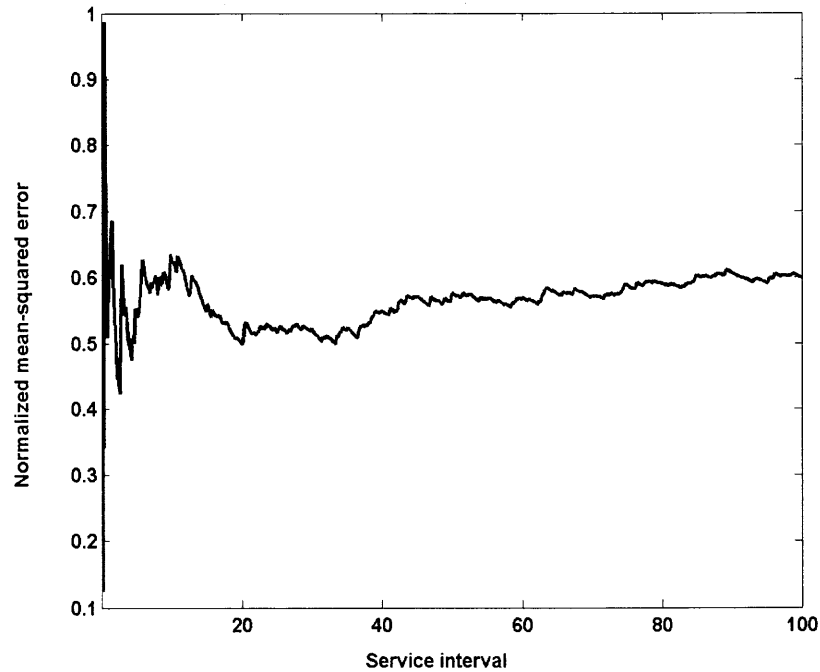
Figure 4.1 Normalized mean-squared prediction error.

The comparison between the success probability of prediction and the cumulative distribution function (CDF) of a Gaussian random variable is shown in Fig. 4.2. It is observed that the prediction error is approximately Gaussian distributed. The intuition behind the phenomenal is that the traffic predictor employs the traffic correlation information, and thus the prediction error is approximately uncorrelated. It was also found by numerous simulations that the autocorrelation of the prediction error in Eq. (3.3) for self-similar network traffic is close to that of the Gaussian, a rather uncorrelated process [53, 54]. Hence, the prediction error delivered by the underlying adaptive linear filter can be assumed to be Gaussian with mean m_i and

Gaussian distribution
Success probability of prediction

1.2

1

0.8

0.6

0.4

0.2

m_i-3σ_i m_i-2σ_i m_i-σ_i m_i m_i+σ_i m_i+2σ_i m_i+3σ_i
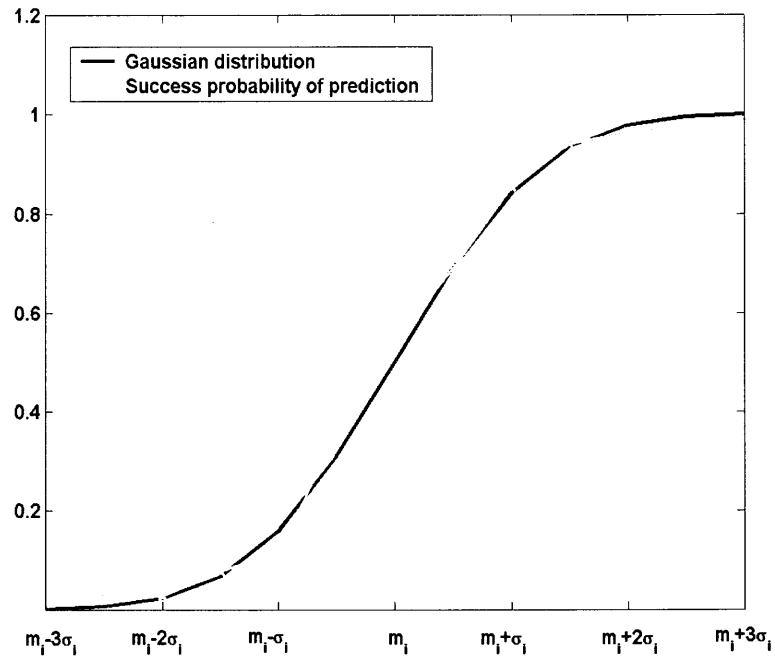
Figure 4.2 Comparison between P_i^s and Gaussian distribution.

variance σ_i^2, i.e., $e_i \sim N(m_i, \sigma_i^2)$, the success probability of traffic prediction is

$$
\begin{aligned}
P_i^s &= P\{e_i \leq 0\} \\
&= \frac{1}{\sqrt{2\pi}\sigma_i} \int_{-\infty}^{0} e^{-(x-m_i)^2/2\sigma_i^2} dx \\
&= 1 - Q(-\frac{m_i}{\sigma_i}) \\
&= Q(\frac{m_i}{\sigma_i})
\end{aligned}
\tag{4.2}
$$

$Q(.)$ is the Q-function [55], defined as $Q(a) = \int_a^{\infty} \frac{1}{\sqrt{2\pi}} e^{(-\frac{x^2}{2})} dx$. The probability that the prediction fails is $P_i^f = 1 - P_i^s$. The inherent property of the Q-function implies that the success probability of prediction relies on the prediction error. When m_i decreases, the bandwidth requirement of ONUi increases, the OLT tries to reserve more upstream bandwidth to transmit the frames arrived during the waiting time, and it is more likely that these frames can be delivered in the next timeslot.

4.1.2 Frame Loss Probability *vs.* Traffic Prediction

As mentioned before, an incoming frame experiences loss if the buffer is full. Assuming the fixed buffer size at ONUi is C_i, the frame loss probability at ONUi is

$$P_i^{loss} = P\left\{b_i^w + b_i^q > C_i\right\} . \tag{4.3}$$

Subtracting $\widehat{b_i^w}$ at both sides of the probability yields

$$
\begin{aligned}
P_i^{loss} &= P\left\{b_i^w - \widehat{b_i^w} + b_i^q > C_i - \widehat{b_i^w}\right\} \\
&= P\left\{b_i^w - \widehat{b_i^w} > C_i - b_i^q - \widehat{b_i^w}\right\} . \\
&= P\left\{e_i > C_i - b_i^q - \widehat{b_i^w}\right\}
\end{aligned}
\tag{4.4}
$$

Employing the assumption that e_i is Gaussian, the frame loss probability at ONUi is

$$P_i^{loss} = Q\left(\frac{C_i - b_i^q - \widehat{b_i^w} - m_i}{\sigma_i}\right) . \tag{4.5}$$

Lemma 4.1: Decreasing m_i and σ_i decreases the frame loss.

Proof: In Eq. (4.5), since Q-function is monotonically decreasing, decreasing m_i and σ_i decreases the Q-function value. Therefore, the frame loss is reduced. ∎

Lemma 4.1 shows the relationship between the frame loss and the traffic prediction. Decreasing m_i increases the bandwidth requirement of ONUi; the OLT will reserve more upstream bandwidth to transmit the frames arrived during the waiting time, and the frame loss could be reduced. When σ_i decreases, the prediction error varies in a smaller range, *i.e.*, the predictor at the OLT side is less likely to underestimate the frames arrived during the waiting time, thereby resulting in less frame loss. In the extreme case, no traffic prediction is employed, $\widehat{b_i^w} = 0$, and $e_i = b_i^w$, then m_i and σ_i^2 essentially are the mean and variance of b_i^w, which are much larger than the ones with traffic prediction. According to Lemma 4.1, an EPON without traffic prediction suffers much heavier data loss as compared to an EPON with the LSTP mechanism.

Lemma 4.2: Given the buffer size of an ONU and the predictor at the OLT, increasing the network traffic load results in heavier frame loss.

Proof: When the buffer size and predictor are given, C_i, m_i, and σ_i in Eq. (4.5) are known. Increasing the network load results in a larger value of b_i^q. Since Q-function is a monotonically decreasing function, a larger value of b_i^q in Eq. (4.5) increases the frame loss probability P_i^{loss}, and more frames will be dropped due to buffer overflow. ∎
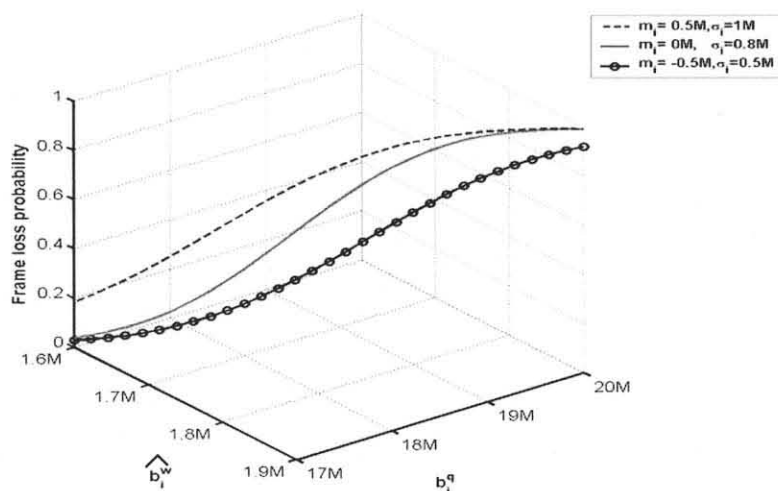
Figure 4.3 Frame loss probability *vs.* traffic load and traffic prediction, C_i=20M.

The numerical results are shown in Fig. 4.3. When the buffer size C_i is fixed, the frame loss probability is closely related to the incoming traffic load and the accuracy of the predictor. When the traffic load is heavy, more frames are enqueued when sending the REPROT message, *i.e.*, a larger b_i^q, and it is more likely that the buffer is fully occupied, and hence, the ONU experiences heavier frame loss. As the mean and variance of the prediction error are decreased, more frames could be accommodated by the fixed buffer, and the frame loss probability is thus decreased.

4.2   Frame Delay

In this section the performance of LSTP in terms of frame delay is theoretically analyzed. The frame delay is defined as the time from enqueuing a frame at an ONU buffer to sending out the last bit of the frame to the OLT. The focus of the following analysis is the delay of the incoming frames during the waiting time.

4.2.1   Frame Delay *vs.* Traffic Prediction

In LSTP, the frame delay of the frames arrived during the waiting time differs according to the prediction result and the OLT bandwidth arbitration. When the prediction succeeds, *i.e.*, $e_i = b_i^w - \widehat{b_i^w} \leq 0$, and the assigned bandwidth is the required value, *i.e.*, $b_i^g = b_i^r$, the allocated timeslot is enough to transfer the incoming frames during the waiting time, and thus, the frame delay is decided by the average service interval length. Assuming the average service interval length is t_{int}, the frame delay in the above case is γt_{int}, where $0 \leq \gamma \leq 1$.

On the other hand, when the traffic prediction fails, *i.e.*, $e_i = b_i^w - \widehat{b_i^w} > 0$, or when the assigned bandwidth is less than the requirement, *i.e.*, $b_i^g < b_i^r$, the frames arrived during the waiting time have to wait for one more service interval to be delivered. The corresponding delay is $(1 + \gamma)t_{int}$.

Combining both of the above cases, the average delay of the frames arrived during the waiting time is

$$D_i = P_i^s P \left\{ b_i^g = b_i^r \right\} \gamma t_{int} + (1 - P_i^s P \left\{ b_i^g = b_i^r \right\})(1 + \gamma)t_{int} \cdot \qquad (4.6)$$

when $B_i \geq b_i^r$, the maximum timeslot length is no less than the bandwidth requirement, and $b_i^g = b_i^r$ occurs. Therefore, $P\{b_i^g = b_i^r\}$ can be further deduced as

$$
\begin{aligned}
P\{b_i^g = b_i^r\} &= P\{B_i \geq b_i^r\} \\
&= P\left\{B_i \geq b_i^q + \widehat{b_i^w}\right\} \\
&= P\left\{B_i - b_i^w \geq b_i^q - b_i^w + \widehat{b_i^w}\right\} \cdot \\
&= P\{e_i \geq b_i^q + b_i^w - B_i\} \\
&= Q(\tfrac{b_i^q + b_i^w - B_i - m_i}{\sigma_i})
\end{aligned}
\tag{4.7}
$$

When the traffic load is heavy, $b_i^q + b_i^w$ increases, and it is more likely that the requested bandwidth is larger than B_i, thereby less possibility that $b_i^g = b_i^r$. When B_i increases, the value of Eq. (4.7) increases, implying that a larger value of the maximum timeslot length leads to higher possibility that $b_i^g = b_i^r$. Combining Eqs. (4.2), (4.6), and (4.7), the average delay of the frames arrived during the waiting time is

$$
D_i = Q(\tfrac{m_i}{\sigma_i})Q(\tfrac{b_i^q + b_i^w - B_i - m_i}{\sigma_i})\gamma t_{int} + \left[1 - Q(\tfrac{m_i}{\sigma_i})Q(\tfrac{b_i^q + b_i^w - B_i - m_i}{\sigma_i})\right](1+\gamma)t_{int} \cdot \tag{4.8}
$$

Let $A = Q(\tfrac{m_i}{\sigma_i})Q(\tfrac{b_i^q + b_i^w - B_i - m_i}{\sigma_i})$, then $0 \leq A \leq 1$, and the average delay is

$$
\begin{aligned}
D_i &= A\gamma t_{int} + (1-A)(1+\gamma)t_{int} \\
&= (1+\gamma)t_{int} - A t_{int}
\end{aligned}
\tag{4.9}
$$

Fig. 4.4 illustrates the numerical results of the average frame delay *vs.* A and the average service interval length t_{int}, when $\gamma = 0.5$. The average delay D_i is determined by the service interval length, the success probability of prediction, and the incoming traffic load. In the case of $A = 0$, which means either the prediction fails or the allocated bandwidth is not enough, the frames arrived during the waiting time are held at the buffer for one more service interval. Therefore, the average waiting time is $(1+\gamma)t_{int}$. In the optimum case, $A = 1$, all of such frames are transmitted in the next timeslot, and the delay equals γt_{int}.
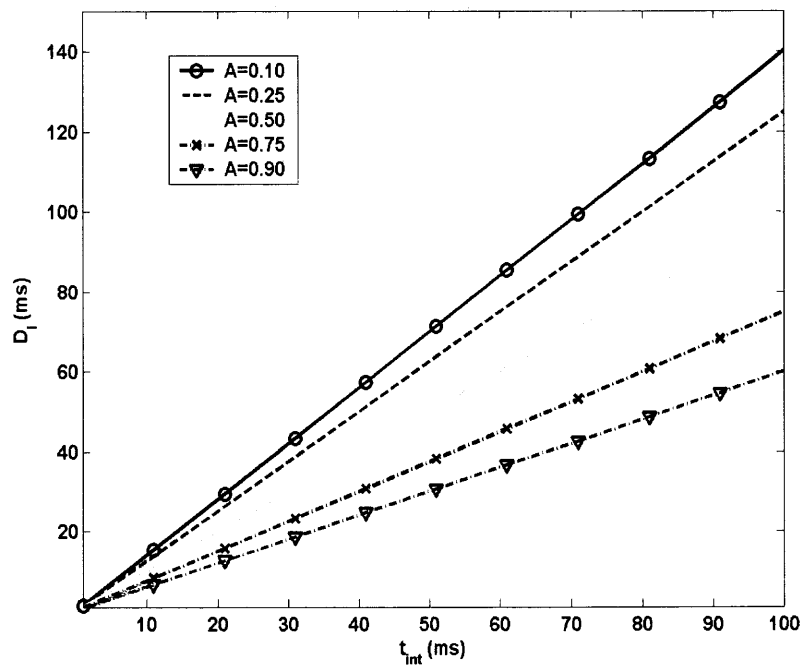
Figure 4.4 Average frame delay *vs.* service interval, $\gamma = 0.5$.

Lemma 4.3: Increasing the success probability of prediction decreases the average delay of the frames arrived during the waiting time.

Proof: Since $A = P_i^s P\{b_i^g = b_i^r\}$, increasing P_i^s increases A. According to Eq. (4.9), a larger A results in a shorter average delay of the frames arrived during the waiting time. Therefore, the performance of the traffic predictor plays a key factor on the average frame delay of the frames arrived during the waiting time. ∎

Lemma 4.4: Increasing the service interval results in longer average frame delay of the frames arrived during the waiting time.

Proof: With respect to the frames arrived during the waiting time, the average frame delay is formulated as Eq. (4.9). Increasing t_{int} increases the average frame delay D_i. ∎

Lemma 4.5: Increasing the traffic load increases the average frame delay of the frames arrived during the waiting time.

Proof: Increasing the traffic load results in a larger $b_i^q + b_i^w$. Since Q-function is monotonically decreasing, increasing $b_i^q + b_i^w$ decreases the value of $Q(\frac{b_i^q+b_i^w-B_i-m_i}{\sigma_i})$, thereby decreasing the value of A. According to Eq. (4.9), a smaller A results in a longer average delay of the frames arrived during the waiting time. ■

4.2.2 Frame Delay Reduction

As compared to a system without traffic prediction, LSTP improves the frame delay of the frames arrived in the waiting time by

$$\beta = \frac{D_{noprediction} - D_i}{D_{noprediction}} = \frac{(1+\gamma)t_{int} - [(1+\gamma)t_{int} - At_{int}]}{(1+\gamma)t_{int}} = \frac{A}{1+\gamma}. \tag{4.10}$$

The delay reduction depends on A. Since $0 \leq A \leq 1$, the maximum delay reduction is achieved when $A = 1$, and the according delay reduction gain is $\frac{1}{1+\gamma}$. Based on Lemma 4.3, increasing the success probability of prediction results in a larger value of A, and higher delay reduction is thus achieved. According to Lemma 4.5, a lighter traffic load leads to a larger value of A, and delay of the frames arrived during the waiting time could be further reduced.

4.3 Queue Length

In this section the performance of LSTP in terms of queue length is theoretically analyzed. The targeted queue length is the difference between the total enqueued frames and the granted bandwidth in bytes during a service interval.

4.3.1 Average Queue Length

The average queue length Z_i is defined as

$$Z_i = \sum_{z=0}^{\infty} zP\{Z_i = z\}. \tag{4.11}$$

Assuming the fixed buffer size of ONUi is C_i, the average queue length then is

$$Z_i = \sum_{z=0}^{C_i} z P\left\{Z_i = z\right\}.$$
(4.12)

4.3.2 Queue Length *vs.* Traffic Prediction

One way to analyze the queue length is from the viewpoint of traffic prediction and bandwidth arbitration. At the end of a service interval, the queue size is the difference between the total enqueued frames and the granted bandwidth in bytes. That is, $P\left\{Z_i = z\right\}$ could be formulated as

$$
\begin{aligned}
P\left\{Z_i = z\right\} &= P\left\{b_i^q + b_i^w - b_i^g = z\right\} \\
&= P\left\{b_i^q + b_i^w - \widehat{b_i^w} - b_i^g = z - \widehat{b_i^w}\right\}. \\
&= P\left\{e_i = z + b_i^g - b_i^q - \widehat{b_i^w}\right\}
\end{aligned}
$$
(4.13)

The difference between the bandwidth requirement and the response is shown as $(b_i^g - b_i^q - \widehat{b_i^w})$, *i.e.*, $(b_i^g - b_i^q - \widehat{b_i^w}) = b_i^g - b_i^r$. Consider the two subcases. When the requirement is less than or equal to B_i, the requirement is granted, and $b_i^g - b_i^r = 0$. When the requirement is more than B_i, B_i is granted, and $b_i^g - b_i^r = B_i - b_i^r < 0$. Including the two subcases into Eq. (4.13), $P\left\{Z_i = z\right\}$ becomes

$$
\begin{aligned}
P\left\{Z_i = z\right\} &= P\left\{e_i = z + b_i^g - b_i^q - \widehat{b_i^w}\right\} \\
&= P\left\{e_i = z + b_i^g - b_i^r\right\} \\
&= P\left\{B_i \geq b_i^r\right\} P\left\{e_i = z\right\} \\
&\quad + (1 - P\left\{B_i \geq b_i^r\right\}) P\left\{e_i = z + B_i - b_i^r\right\}
\end{aligned}
$$
(4.14)

Let W be a Gaussian random variable with the same mean and variance as e_i, then by the central limit theorem [55], $P\left\{e_i = z\right\}$ is approximately equal to the integral of the Gaussian probability density function (pdf) in an interval of unit length about z.

$$
\begin{aligned}
P\left\{e_i = z\right\} &\approx P\left\{z - \tfrac{1}{2} < W < z + \tfrac{1}{2}\right\} \\
&= \frac{1}{\sqrt{2\pi}\sigma_i} \int_{z-\frac{1}{2}}^{z+\frac{1}{2}} e^{-(x-m_i)^2/2\sigma_i^2} dx
\end{aligned}
$$
(4.15)

The above approximation can be simplified by approximating the integral utilizing the product of the integrand at the center of the interval of integration (that is, $x = z$) and the length of the interval of integration (that is, 1) as

$$P\{e_i = z\} \approx \tfrac{1}{\sqrt{2\pi}\sigma_i} e^{-(z-m_i)^2/2\sigma_i{}^2} \ . \tag{4.16}$$

Moreover, $P\{B_i \geq b_i^r\}$ is formulated in Eq. (4.7) as $P\{B_i \geq b_i^r\} = Q(\tfrac{b_i^q + b_i^w - B_i - m_i}{\sigma_i})$, and therefore, the average queue length is

$$\begin{aligned}
Z_i &= \textstyle\sum_{z=0}^{C_i} z P\{Z_i = z\} \\
&= \textstyle\sum_{z=0}^{C_i} z [Q(\tfrac{b_i^q + b_i^w - B_i - m_i}{\sigma_i}) \tfrac{1}{\sqrt{2\pi}\sigma_i} e^{-\frac{(z-m_i)^2}{2\sigma_i^2}} \\
&\quad + (1 - Q(\tfrac{b_i^q + b_i^w - B_i - m_i}{\sigma_i})) \tfrac{1}{\sqrt{2\pi}\sigma_i} e^{-\frac{(z+B_i-b_i^r-m_i)^2}{2\sigma_i^2}}]
\end{aligned} \tag{4.17}$$
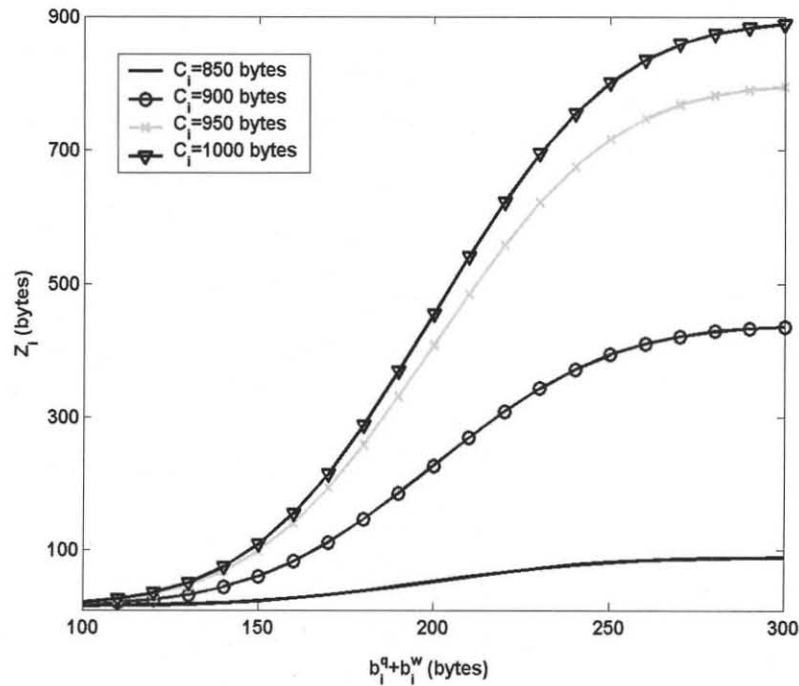

Figure 4.5 Average queue length *vs.* traffic load and buffer size.

The numerical results of the average queue length *vs.* the buffer size C_i, the mean of prediction error m_i, the variance of prediction error σ_i^2, and the maximum
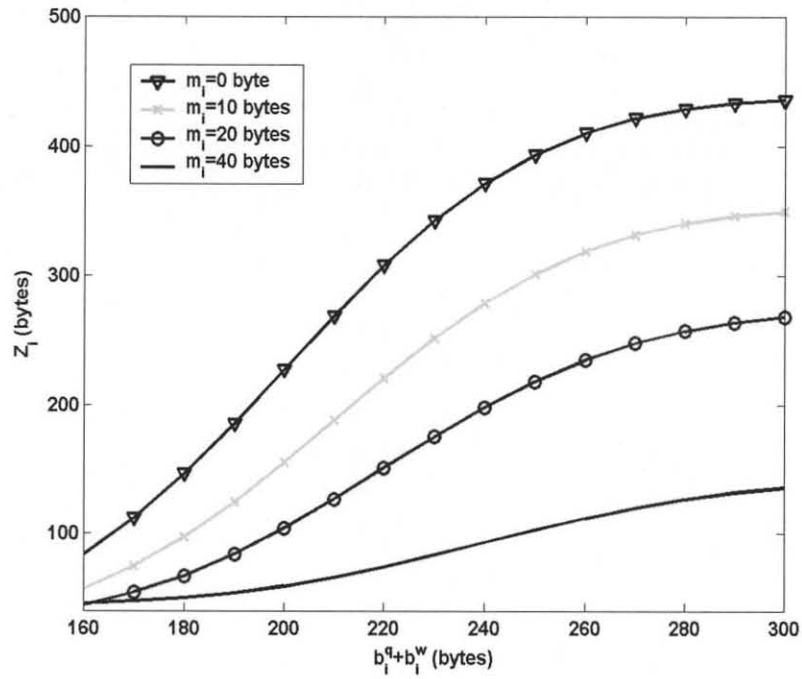
Figure 4.6 Average queue length *vs.* traffic load and m_i.
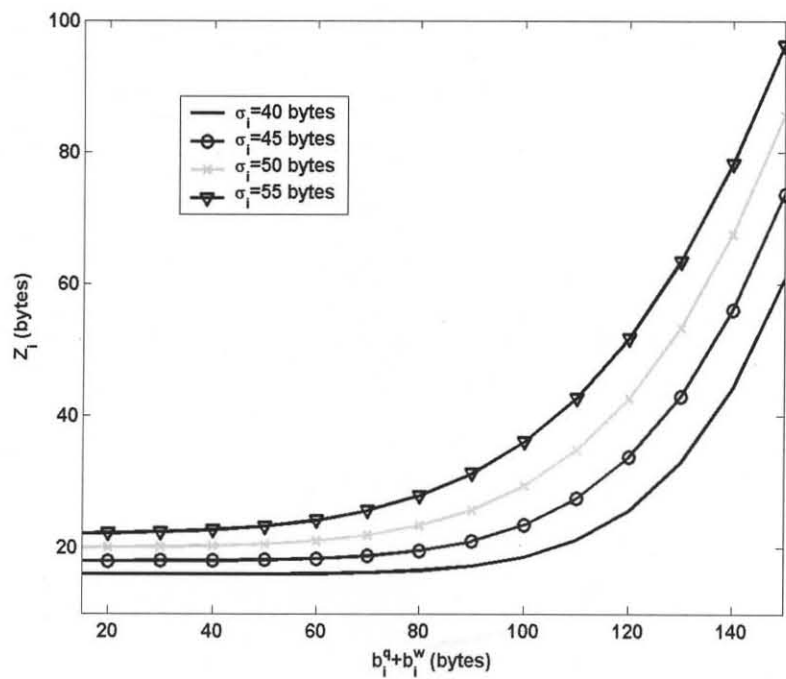
Figure 4.7 Average queue length *vs.* traffic load and σ_i.
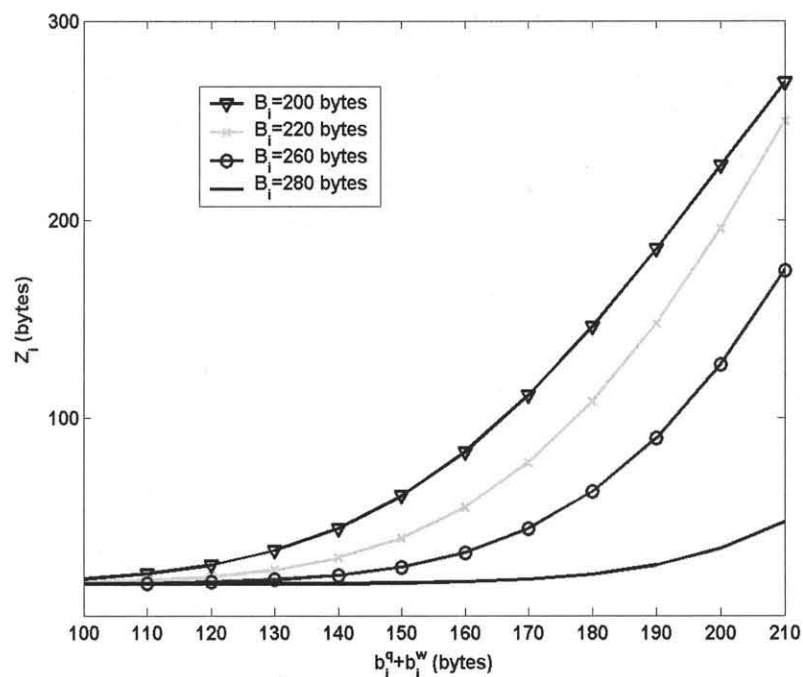
Figure 4.8 Average queue length *vs.* traffic load and B_i.

timeslot length B_i are shown in Figs. 4.5, 4.6, 4.7, and 4.8, respectively. The following properties are observed from the numerical analysis.

Property 4.1: Increasing the mean of prediction error decreases the average queue length of an ONU.

Property 4.2: Increasing the variance of prediction error increases the average queue length of an ONU.

Property 4.3: Increasing the maximum timeslot length decreases the average queue length of an ONU.

4.4 Simulation Results

The LSTP mechanism performance is further evaluated via simulation results. A system model shown in Fig. 2.5 is set up in the OPNET simulator with one OLT and 16 ONUs. The distance from an ONU to the OLT is assumed to vary from 10 km to 20 km. Each ONU has a finite buffer of 20 Mbytes, and the downstream

and upstream channels are both 1.25Gbps. The incoming traffic is self-similar with the Hurst parameter of 0.8. The length of Ethernet frames randomly varies from 64 bytes to 1518 bytes. The total traffic load of the network, defined as the average arrival to the service rate, is changing from 0.1 to 0.8. For comparison purposes, FBA, LBA [29], EBA [31], and LSTP are applied. In FBA, the upstream bandwidth is evenly distributed among the 16 ONUs. In LBA, B_i of 8 ONUs are set as the evenly distributed bandwidth as in FBA, and B_i of the other 8 ONUs are set 10% $\sim$ 50% larger. B_i in EBA and LSTP is set the same as that in LBA. The order of the predictor, $i.e.$, L, in LSTP is set to 4, and the step size $\mu_i(n)$ is set by [48]

$$\mu_i(n) = \frac{L}{\sum_{k=0}^{k=L-1} [b_i^w(n-k)]^2}.$$
(4.18)
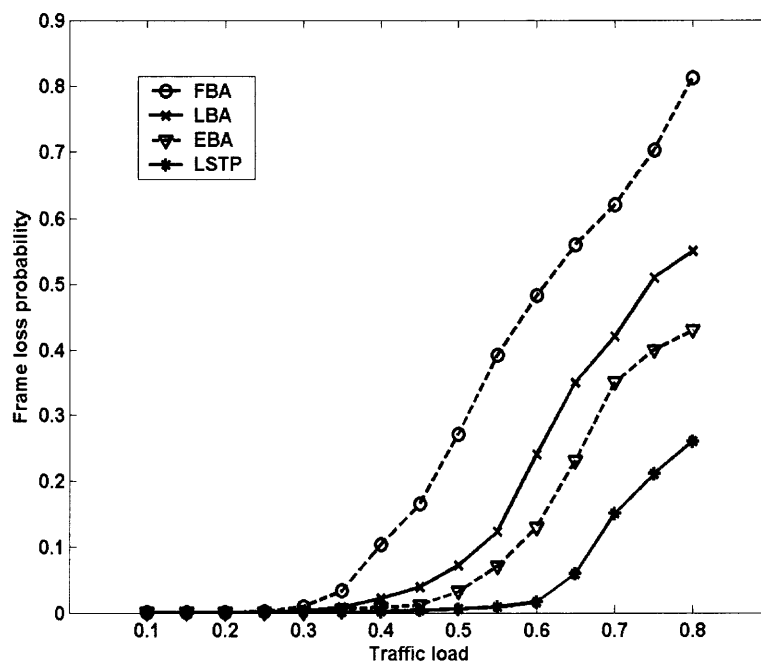
Figure 4.9 Simulations on frame loss probability $vs.$ traffic load.

The figures of merits are the frame loss, the frame delay, and the queue length. Fig. 4.9 illustrates the relationship between the frame loss probability and the network

traffic load. The frame loss probability is defined as the number of dropped frames *vs.* the total number of incoming frames. FBA experiences the heaviest frame loss, which is attributed to the fact that FBA disregards the dynamics of the incoming traffic, and thus, more frames are likely deferred to one more service interval. Even at the load of 0.4, the frame loss probability is very high (about 10%). This is attributed to the bursty nature of the traffic, and most frames arrive in bursts. The number of frames in bursts are so large that the local buffer at an ONU overflows, and about 10% of the frames are dropped. LBA alleviates this problem by accommodating the REPORT and the GATE messages to keep track of the incoming traffic. EBA redistributes the under-exploited bandwidth of the lightly loaded ONUs among the heavily loaded ones, thereby alleviating the frame loss of the heavily loaded ONUs. LSTP outperforms all of the above three mechanisms. Several points contribute to the lowest frame loss probability in LSTP. First, LSTP predicts the frames arrived during the waiting time, and pre-reserves bandwidth to transmit them in the next timeslot, thus dramatically reducing the possibility of buffer overflow. Second, LSTP implements the fixed ONU service order instead of the dynamic service order in EBA, and reduces the drastic change of the service interval length of an ONU in EBA, thus facilitating the traffic prediction. Third, the OLT responds to the ONU bandwidth requirement instantaneously in LSTP. In EBA, the heavily-loaded ONUs are always served after the lightly-loaded ones, and the deferred service for those heavily-loaded ONUs results in longer delay of the incoming frames.

Fig. 4.10 shows the average delay of all frames transmitted over the EPON. It exhibits the similar trend to that of the frame loss. Again, FBA has the longest average frame delay, and LSTP has the least, implying that the traffic prediction, the fixed service order, and the instantaneous bandwidth allocation provided by LSTP decrease the average frame delay, thus reducing the extra delay that a frame experiences when arriving during the waiting time. In fact, LSTP monitors the
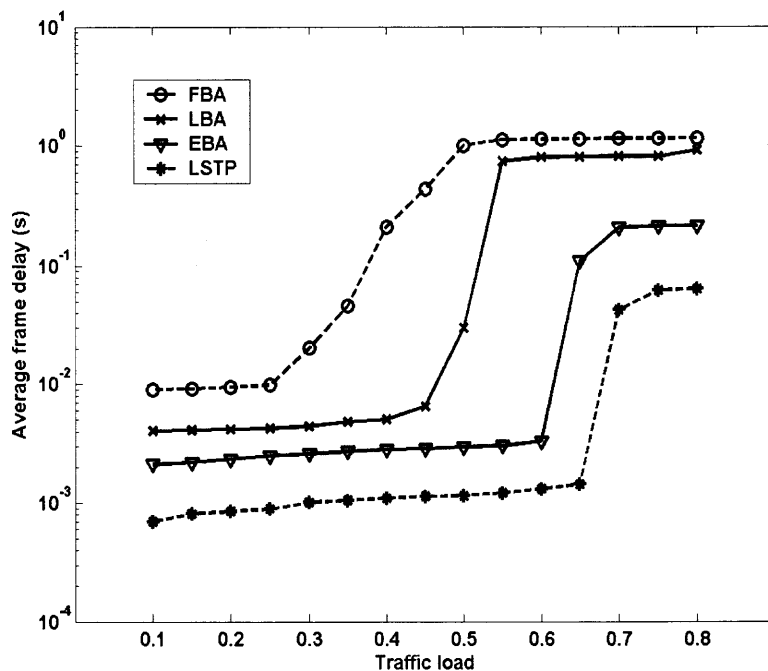
Figure 4.10 Simulations on average frame delay *vs.* traffic load.

network traffic load, and the shrunk or the extended service cycles adapt exactly to the amount of data offered to the network. The LMS-based predictor delivers satisfactory prediction to the self-similar traffic, and the length sum of the frames arrived during the waiting time can be forecasted very well.

The performance of FBA, LBA, EBA, and LSTP in terms of the average queue length exhibits the similar trend to that of the average frame delay. As shown in Fig. 4.11, FBA has the longest queue, while LSTP has the shortest queue. The queue size of FBA increases drastically even in light traffic load (less than 0.4). The major reason is the fixed service cycle, which does not take the on-line traffic pattern into consideration. Therefore, in the worst case, incoming frames of some ONUs have to be backlogged when the upstream channel is occupied by the other ONUs without data transmission. When the traffic load is light (less than 0.4), the queue size of LBA, EBA, and LSTP increases slowly; this shows that dynamic bandwidth negotiation using REPORT/GATE plays a major role in reducing the queue length.
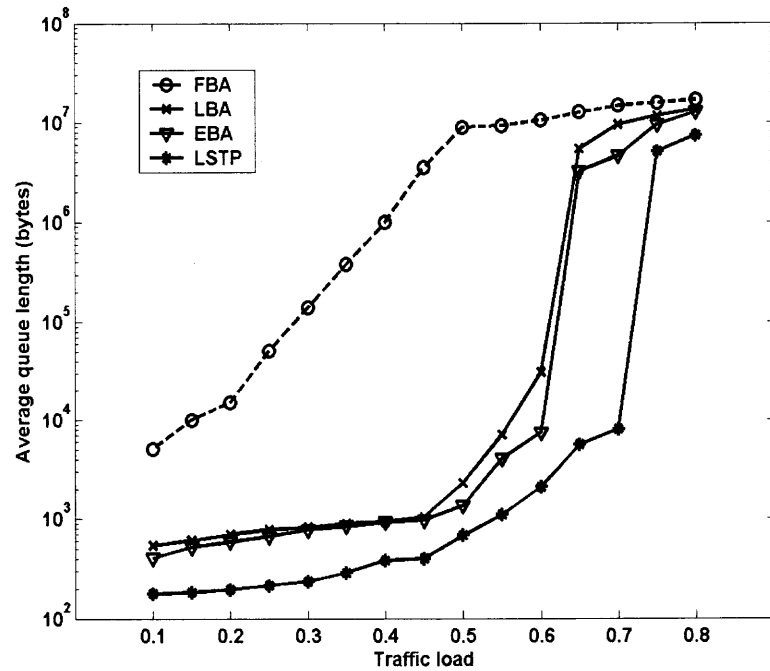
Figure 4.11 Simulations on average queue length *vs.* traffic load.

The shortest queue length in LSTP can be attributed to the fact that LSTP provides the least fame delay. A shorter average frame delay means that the ONUs transmit the frames faster, and therefore, less number of frames are held at the buffer.

4.5 Chapter Summary

This chapter presents the theoretical approaches to analyze the frame loss probability, frame delay, and queue length for the LSTP mechanism. Guidelines based on the success probability of prediction and the network traffic load are proposed and validated. The properties and numerical results justify the contribution of the LMS-based predictor to the performance improvement. As compared to FBA, LBA, and EBA, LSTP is able to curb the unfair delay of the frames arrived during the waiting time, and thus decreasing the frame loss and queue length over EPONs.

CHAPTER 5

LSTP ENHANCEMENT WITH SERVICE DIFFERENTIATION

As mentioned in Chapter 2, service differentiation is a distinguished feature that the broadband access network is expected to provide, since Ethernet does not support QoS directly while the access network is required to accommodate various kinds of traffics. In this chapter, LSTP is enhanced with the provisioning of service differentiation. Approached by means of a combination of traffic classification, priority queuing, scheduling, and class-based bandwidth allocation, the enhanced LSTP aims for supporting data, voice, and video services to the end users.

5.1 Class-Based Bandwidth Negotiation

Service differentiation is desired to accommodate heterogeneous applications and user expectations, permitting differentiated pricing of services. Categorizing the traffics of an ONU into different classes is a practical and scalable approach to provide service differentiation over broadband access networks. A "class" defines some significant characteristics of data transmission from the end users to the access network. These characteristics may be specified in quantitative or statistical terms, such as bandwidth, delay, jitter, loss, and throughput, or may otherwise be specified in terms of some relative priority of access to network resources.

In this dissertation, characteristics adopted to classify traffics are bandwidth and delay. The idea is borrowed from DiffServ [10], and traffics at an ONU are classified into three classes. The high priority class is the expedited forwarding (EF), which is delay sensitive and requires bandwidth guarantees. The medium priority class is the assured forwarding (AF), which is not delay sensitive, but requires bandwidth

guarantees. The low priority class is the best effort (BE), which is neither delay sensitive nor bandwidth guaranteed.

Different traffic classes have different priorities of access to network resources, and thereby, the upstream bandwidth negotiation is required to be done for each class separately. The class-based bandwidth negotiation requires that an ONU is capable of sending bandwidth requirement of each class, and the bandwidth arbitration is decided by the OLT at the traffic class level.

As shown in Fig. 3.2, one 64-byte REPORT message from an ONU reports up to 8 queues' status. When a REPORT message is received by the OLT, the latter should be able to classify the reported status to the particular queues, and hence, the 1-byte "report bitmap" field is included in the REPORT message to identify the order of the reported queues. For example, "11100000" indicates that three queues ($q1$, $q2$, and $q3$) have reported and their status follows in the same order in the remaining part of the REPORT message.

Similarly, as shown in Fig. 3.4, one 64-byte GATE message carries up to 6 grants to a particular ONU, and each grant contains a grant start time and a grant length. When a GATE message is received by an ONU, the latter should be able to classify grants to their particular queues. The 1-byte "number of grants" field specifies how many grants are in the GATE message, and the 1-byte "grant level" field indicates the order of the queues to which grants are generated. For example, "11100000" indicates that three queues ($q1$, $q2$, and $q3$) have been assigned grants and their grant content follows in the same order in the remaining part of the GATE message.

If one queue contains the information about a specific class of traffic, it is possible to report the status of multiple classes of traffics at an ONU. Specifically, if the incoming traffics at an ONU are classified into EF, AF, and BE classes, an ONU could manage these three classes of traffics by accommodating three queues.

Their bandwidth requirements are sent to the OLT in one REPORT message, and the OLT sends back one GATE message after processing the report, including at least one grant, depending on the bandwidth allocation algorithm. In the enhanced LSTP with service differentiation, a REPORT message is piggybacked at the end of each timeslot, including the bandwidth requirement of three classes of traffics. Once the OLT makes the bandwidth arbitration decision, a GATE message is sent back to a particular ONU, including the bandwidth assignment to the three classes of traffics.

5.2   Service Differentiation at an ONU

5.2.1   System Model

A physical buffer residing at an ONU is assumed to support classified traffics. Fig. 5.1 shows the functional blocks of an ONU. Three separate queues sharing a common physical buffer are maintained at an ONU. One classifier is used to "steer" customer traffics, matching the classification rule to frames for further processing. For example, the classifier checks the type-of-service (TOS) field of the IP packet encapsulated in an Ethernet frame, and classifies the frames into EF, AF, or BE traffics. EF traffic has the highest priority, AF traffic has the medium priority, while BE traffic has the lowest priority. The classified incoming frames are then placed into their appropriate queues.

Instead of the drop-tail policy described in Chapter 3, the dropping module controls the amount of enqueued frames of each class, and drops the unnecessary ones. Once the assigned transmission timeslot arrives, the scheduling module decides the transmission order of the enqueued frames. In order to alleviate the delay of the frames arrived during the waiting time, as in Chapter 3, a traffic predictor is employed to predict that portion of traffics in a service interval.
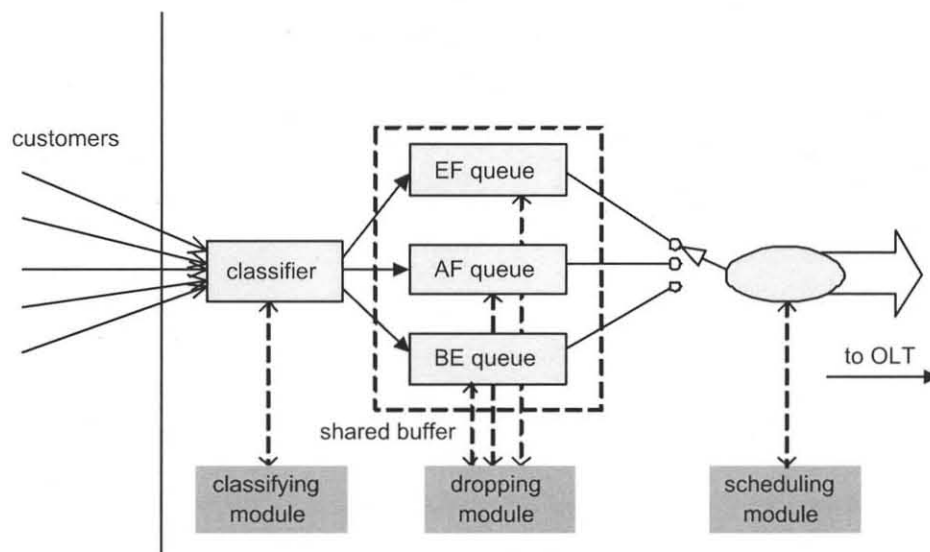
Figure 5.1 System model of an ONU with service differentiation.

5.2.2 Priority Queuing

The SLA parameters specifying the maximum length of enqueued EF, AF, and BE traffics at ONUi are $q_{EF,i}$, $q_{AF,i}$, and $q_{BE,i}$, respectively ($q_{EF,i} \leq C_i$, $q_{AF,i} \leq C_i$, and $q_{BE,i} \leq C_i$, where C_i is the buffer size of ONUi). The flow of traffic policing is illustrated in Fig. 5.2. The priority queuing discipline works as follows: 1) when the sum of the enqueued EF frame length surpasses $q_{EF,i}$, the incoming EF frames are dropped immediately; 2) when the sum of the enqueued AF frame length surpasses $q_{AF,i}$, the incoming AF frames are dropped immediately; 3) when the sum of the enqueued BE frame length surpasses $q_{BE,i}$, the incoming BE frames are dropped immediately; 4) when the buffer is full, an incoming frame with higher priority displaces a lower priority frame, while an incoming low priority frame is dropped immediately.

After classifying, frames are checked for their conformance, and unnecessary ones are dropped. The above traffic policing regulates the flow of higher priority traffic, ensuring that it conforms to its SLA. Without the maximum length limitation, the higher priority traffic (*e.g.*, EF traffic) may aggressively displace the low priority

traffic (*e.g.*, BE traffic). In the extreme case, when the EF traffic load is heavier than the agreed SLA, all enqueued AF and BE frames would be replaced by the EF frames, and thus penalizing the AF and BE frames with indefinite increase in delay, heavier loss, and uncontrollable access to the upstream channel.

5.2.3 Class-Based Queue Status Report

In service interval n, ONUi reports its local queue status by piggybacking a REPORT message at the end of timeslot n. Different from the LSTP mechanism, a REPORT message carries three queue status (EF queue, AF queue, and BE queue). The bandwidth arbitration is conducted by the centralized process in the OLT.

5.3 Service Differentiation at the OLT

5.3.1 Class-Based Traffic Prediction and Bandwidth Requirement

The predictor embedded in the OLT forecasts the incoming frames of different classes separately as

$$\widehat{b^w_{c,i}}(n+1) = \sum_{k=0}^{L-1} \alpha_{c,i,k}(n) b^w_{c,i}(n-k), \tag{5.1}$$

where $\widehat{b^w_{c,i}}(n)$ is the predicted class c frames in bytes arrived at ONUi during the waiting time of service interval n, $c \in \{EF, AF, BE\}$, and $b^w_{c,i}(n)$ is the arrived frames of class c in bytes at ONUi during the waiting time of service interval n . Accordingly, the weight factor of the LP is updated as

$$\alpha_{c,i,k}(n+1) = \alpha_{c,i,k}(n) + \mu_{c,i}(n)\frac{e_{c,i}(n)}{b^w_{c,i}(n)}, \tag{5.2}$$

where the prediction error $e_{c,i}(n)$ is defined as

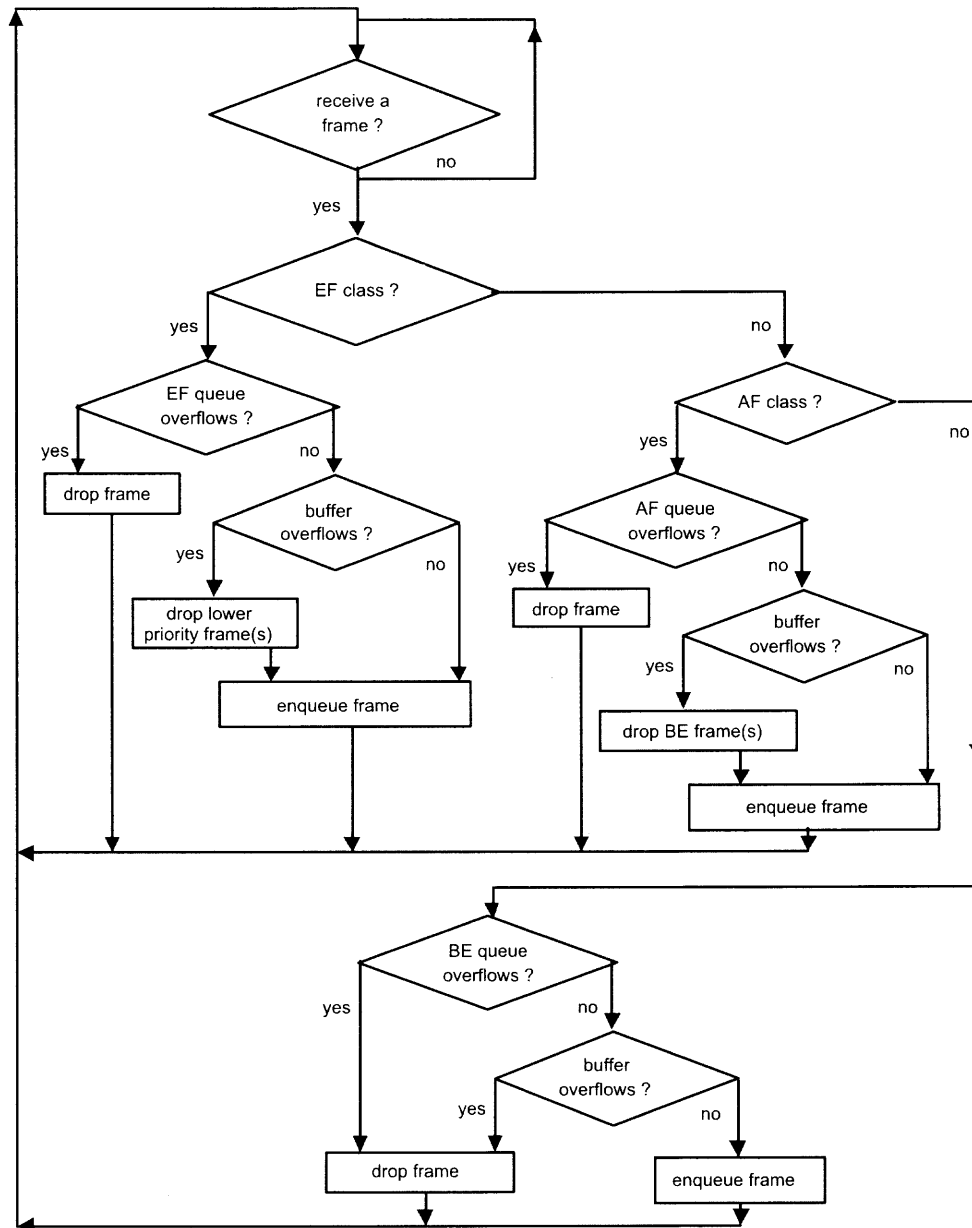$$e_{c,i}(n) = b^w_{c,i}(n) - \widehat{b^w_{c,i}}(n). \tag{5.3}$$

Figure 5.2 Priority queuing at an ONU.

After receiving the queue status report from ONUi, the OLT calculates its bandwidth requirement for service interval $n + 1$ in the class level. For each class of traffic, the bandwidth requirement is the already enqueued frames plus the prediction, i.e.,

$$b_{c,i}^r(n + 1) = b_{c,i}^q(n) + \widehat{b_{c,i}^w}(n). \tag{5.4}$$

In Eq. (5.4), $b_{c,i}^r(n + 1)$ is the bandwidth requirement of class c traffic at ONUi for service interval $n + 1$, and $b_{c,i}^q(n)$ is the enqueued class c frames in bytes at ONUi when sending the REPORT message. As an extension to the basic LSTP mechanism, the OLT predicts the incoming traffic arrived during the waiting time in the traffic class level. The intuition behind this prediction is again the traffic self-similarity. The correlation in one class of traffic does not decay rapidly, and one class of traffic is correlated between timeslots. Therefore, it is feasible to estimate the incoming traffic by the information of several previous service intervals.

5.3.2 Class-Based Bandwidth Arbitration

After receiving a REPORT message, the OLT calculates and processes the class-based bandwidth requirements immediately. The assigned bandwidth for one class of traffic is upper-bounded by its SLA as

$$b_{c,i}^g(n + 1) = min\left\{b_{c,i}^r(n + 1), B_{c,i}\right\}, \tag{5.5}$$

where $B_{c,i}$ is the maximum timeslot length of class c traffic at ONUi, and is specified by the SLA.

The bandwidth allocated to class c traffic at ONUi is upper-bounded by the smaller value of the bandwidth requirement $b_{c,i}^r(n + 1)$, and the maximum timeslot length $B_{c,i}$. $B_{c,i}$ is specified in the contract between the service provider and the customer.

When the bandwidth requirement is no more than the maximum timeslot length, class c traffic at ONUi is called "underloaded". The assigned bandwidth dynamically keeps track of the incoming class c traffic. A portion of the upstream bandwidth is pre-reserved to transmit the class c traffic arrived during the waiting time, thus dramatically alleviating the frame deferral phenomenon illustrated in Chapter 3.

When the requirement is more than the maximum timeslot length, class c traffic at ONUi is called "overloaded". In this case, this class of traffic violates the SLA. Therefore, $B_{c,i}$ is employed as an upper-bound, limiting the aggressive competition for the upstream bandwidth while ensuring data transmission of other classes of traffics.

5.4 Class-Based Data Loss Analysis

The objective of the class-based service is to enhance the basic LSTP with service differentiation. Therefore, the class-based data loss is one of the major criteria for performance evaluation. It is expected that better services are provided to the higher priority frames.

The LSTP enhancement employs the prioritized queuing mechanism illustrated in Fig. 5.2, with the EF frames having the highest priority, the AF frames having the medium priority, and the BE frames having the lowest priority. Each traffic has its own SLA parameter to upper-bound the queue length. All frames share a common physical buffer; when the shared buffer is full, the incoming higher priority frames replace the already enqueued lower priority ones. The class-based frame loss probabilities are analyzed in the following. For notational simplicity, the referencing of the service interval is omitted.

5.4.1 EF Frame Loss

Given that $q_{EF,i} \leq C_i$, the EF frames are lost if the sum of the enqueued EF frame length is larger than the SLA parameter $q_{EF,i}$. The EF frame loss probability $P_{EF,i}^{loss}$

is

$$
\begin{aligned}
P_{EF,i}^{loss} &= P\left\{b_{EF,i}^w + b_{EF,i}^q > q_{EF,i}\right\} \\
&= P\left\{b_{EF,i}^w - \widehat{b_{EF,i}^w} + b_{EF,i}^q > q_{EF,i} - \widehat{b_{EF,i}^w}\right\} \\
&= P\left\{b_{EF,i}^w - \widehat{b_{EF,i}^w} > q_{EF,i} - b_{EF,i}^q - \widehat{b_{EF,i}^w}\right\} \\
&= P\left\{e_{EF,i} > q_{EF,i} - b_{EF,i}^q - \widehat{b_{EF,i}^w}\right\}
\end{aligned}
\tag{5.6}
$$

As analyzed in Chapter 4, the traffic predictor employs the traffic correlation information, and the prediction error is thus approximately uncorrelated. Further assuming that the prediction error of the EF traffic is Gaussian with mean $m_{EF,i}$ and variance $\sigma_{EF,i}^2$, i.e., $e_{EF,i} \sim N(m_{EF,i}, \sigma_{EF,i}^2)$, the frame loss probability of the EF class can be deduced as

$$
\begin{aligned}
P_{EF,i}^{loss} &= P\left\{e_{EF,i} > q_{EF,i} - b_{EF,i}^q - \widehat{b_{EF,i}^w}\right\} \\
&= Q\left(\frac{q_{EF,i} - b_{EF,i}^q - \widehat{b_{EF,i}^w} - m_{EF,i}}{\sigma_{EF,i}}\right)
\end{aligned}
\tag{5.7}
$$

Since Q-function is monotonically decreasing, the following properties can be deduced from Eq. (5.7): First, the frame loss of the EF traffic is determined by the SLA parameter $q_{EF,i}$, and increasing the allowed EF queue length reduces the EF frame loss. Second, increasing the EF traffic load results in heavier EF frame loss. Third, the EF traffic predictor contributes to the frame loss control, and decreasing $m_{EF,i}$ and $\sigma_{EF,i}^2$ reduces the EF frame loss. Therefore, the EF frame loss is immune to the onslaught of AF and BE frames, implying that the EF frames have the highest priority to access both the local buffer and the upstream bandwidth.

As compared to the extreme case, when no traffic prediction is employed, $\widehat{b_{EF,i}^w} = 0$ and $e_{EF,i} = b_{EF,i}^w$, then $m_{EF,i}$ and $\sigma_{EF,i}$ are the mean and the standard deviation of $b_{EF,i}^w$, which are much larger than the ones with traffic prediction in the LSTP enhancement. According to the properties of Q-function, in Eq. (5.7), increasing $m_{EF,i}$ and $\sigma_{EF,i}$ results in a larger value of $P_{EF,i}^{loss}$. Therefore, an EPON without traffic

prediction suffers much heavier EF frame loss as compared to that with the LSTP enhancement.

5.4.2 AF Frame Loss

The AF traffic experience loss in the following two scenarios: First, if the buffer is fully occupied by either EF or AF frames, an incoming AF frame will be dropped; Second, when the sum of the enqueued AF frame length is larger than the SLA parameter $q_{AF,i}$, an incoming AF frame will be dropped immediately. The corresponding loss probability is

$$
\begin{aligned}
P_{AF,i}^{loss} &= (1 - P_{EF,i}^{loss})P\left\{b_{EF,i}^w + b_{EF,i}^q + b_{AF,i}^w + b_{AF,i}^q > C_i\right\} \\
&\quad + P_{EF,i}^{loss}P\left\{q_{EF,i} + b_{AF,i}^w + b_{AF,i}^q > C_i\right\} \\
&\quad + P\left\{b_{AF,i}^w + b_{AF,i}^q > q_{AF,i}\right\} \\
&= (1 - P_{EF,i}^{loss})P\left\{e_{EF,i} + e_{AF,i} > C_i - b_{EF,i}^q - \widehat{b_{EF,i}^w} - b_{AF,i}^q - \widehat{b_{AF,i}^w}\right\} \\
&\quad + P_{EF,i}^{loss}P\left\{e_{AF,i} > C_i - q_{EF,i} - b_{AF,i}^q - \widehat{b_{AF,i}^w}\right\} \\
&\quad + P\left\{e_{AF,i} > q_{AF,i} - b_{AF,i}^q - \widehat{b_{AF,i}^w}\right\}
\end{aligned}
\tag{5.8}
$$

Since the LSTP enhancement employs separate predictors to the EF and AF frames, $e_{EF,i}$ and $e_{AF,i}$ are independent. Borrowing the analysis of the EF traffic prediction error and further assuming that $e_{AF,i}$ is approximately Gaussian distributed with mean $m_{AF,i}$ and variance $\sigma_{AF,i}^2$, i.e., $e_{AF,i} \sim N(m_{AF,i}, \sigma_{AF,i}^2)$, the frame loss probability of the AF class is

$$
\begin{aligned}
P_{AF,i}^{loss} &= (1 - P_{EF,i}^{loss})P\left\{e_{EF,i} + e_{AF,i} > C_i - b_{EF,i}^q - \widehat{b_{EF,i}^w} - b_{AF,i}^q - \widehat{b_{AF,i}^w}\right\} \\
&\quad + P_{EF,i}^{loss}P\left\{e_{AF,i} > C_i - q_{EF,i} - b_{AF,i}^q - \widehat{b_{AF,i}^w}\right\} \\
&\quad + P\left\{e_{AF,i} > q_{AF,i} - b_{AF,i}^q - \widehat{b_{AF,i}^w}\right\} \\
&= (1 - P_{EF,i}^{loss})Q\left(\frac{C_i - \sum_{c=EF,AF}(b_{c,i}^q + \widehat{b_{c,i}^w} + m_{c,i})}{\sqrt{\sum_{c=EF,AF}\sigma_{c,i}^2}}\right) \\
&\quad + P_{EF,i}^{loss}Q\left(\frac{C_i - q_{EF,i} - b_{AF,i}^q - \widehat{b_{AF,i}^w} - m_{AF,i}}{\sigma_{AF,i}}\right) \\
&\quad + Q\left(\frac{q_{AF,i} - b_{AF,i}^q - \widehat{b_{AF,i}^w} - m_{AF,i}}{\sigma_{AF,i}}\right)
\end{aligned}
\tag{5.9}
$$

As formulated in Eq. (5.9), the AF frame loss is determined by the buffer size, the EF and AF traffic load, the EF and AF traffic predictors, and the SLA parameter $q_{AF,i}$. Increasing the allowed AF queue length could reduce the AF frame loss, increasing the AF traffic load results in heavier AF frame loss, and improving the predictor accuracy reduces the AF frame loss. Although the AF frame loss is immune to the BE traffic, the load of the EF traffic plays an important role in the AF frame loss control. Particularly, under the condition that the EF traffic behaves according to the SLA parameter $q_{EF,i}$, the AF frame loss increases as the EF traffic load gets heavier, implying that the EF frames are granted with higher priority than the AF frames.

5.4.3   BE Frame Loss

Similarly, an incoming BE frame is dropped in the following two cases: First, when the buffer is full, an incoming BE frame will be dropped; Second, when the BE traffic queue length is larger that $q_{BE,i}$, an incoming BE frame will be dropped by the dropping module in the ONU. Assuming the prediction error of the BE frames is Gaussian with mean $m_{BE,i}$ and variance $\sigma^2_{BE,i}$, $i.e.$, $e_{BE,i} \sim N(m_{BE,i}, \sigma^2_{BE,i})$, the frame

loss probability of the BE class traffic at ONUi is formulated in Eq. (5.10).

$$P_{BE,i}^{loss} = (1 - P_{EF,i}^{loss})(1 - P_{AF,i}^{loss})P\left\{b_{EF,i}^w + b_{EF,i}^q + b_{AF,i}^w + b_{AF,i}^q + b_{BE,i}^w + b_{BE,i}^q > C_i\right\}$$

$$+P_{EF,i}^{loss}(1 - P_{AF,i}^{loss})P\left\{q_{EF,i} + b_{AF,i}^w + b_{AF,i}^q + b_{BE,i}^w + b_{BE,i}^q > C_i\right\}$$

$$+(1 - P_{EF,i}^{loss})P_{AF,i}^{loss}P\left\{b_{EF,i}^w + b_{EF,i}^q + q_{AF,i} + b_{BE,i}^w + b_{BE,i}^q > C_i\right\}$$

$$+P_{EF,i}^{loss}P_{AF,i}^{loss}P\left\{q_{EF,i} + q_{AF,i} + b_{BE,i}^w + b_{BE,i}^q > C_i\right\}$$

$$+P\left\{b_{BE,i}^w + b_{BE,i}^q > q_{BE,i}\right\}$$

$$= (1 - P_{EF,i}^{loss})(1 - P_{AF,i}^{loss})Q\left(\frac{C_i - \sum_c(b_{c,i}^q + \widehat{b_{c,i}^w} + m_{c,i})}{\sqrt{\sum_c \sigma_{c,i}^2}}\right)$$

$$+P_{EF,i}^{loss}(1 - P_{AF,i}^{loss})Q\left(\frac{C_i - q_{EF,i} - \sum_{c=AF,BE}(b_{c,i}^q + \widehat{b_{c,i}^w} + m_{c,i})}{\sqrt{\sum_{c=AF,BE} \sigma_{c,i}^2}}\right)$$

$$+(1 - P_{EF,i}^{loss})P_{AF,i}^{loss}Q\left(\frac{C_i - q_{AF,i} - \sum_{c=EF,BE}(b_{c,i}^q + \widehat{b_{c,i}^w} + m_{c,i})}{\sqrt{\sum_{c=EF,BE} \sigma_{c,i}^2}}\right)$$

$$+P_{EF,i}^{loss}P_{AF,i}^{loss}Q\left(\frac{C_i - q_{EF,i} - q_{AF,i} - b_{BE,i}^q - \widehat{b_{BE,i}^w} - m_{BE,i})}{\sigma_{BE,i}}\right)$$

$$+Q\left(\frac{q_{BE,i} - b_{BE,i}^q - \widehat{b_{BE,i}^w} - m_{BE,i}}{\sigma_{BE,i}}\right)$$

$$(5.10)$$

The BE frame loss is thus dominated by the EF and AF traffics. When the high priority traffics are heavy, the BE frames are more likely dropped due to buffer overflow, and frame loss increase is thus inevitable. Similar to the EF traffic predictor, the contributions of the AF and BE traffic predictors are to follow the dynamics of the on-line data. Decreasing the means and variances of the prediction errors reduces the frame loss of each class accordingly.

5.5   Simulation Results

As in Chapter 4, simulations are conducted in the system with one OLT and 16 ONUs. The downstream and upstream channels are both 1.25 Gbps. Each ONU has a finite buffer of 20 Mbytes, and the distance from an ONU to the OLT is assumed to be from 10 km to 20 km. The length of Ethernet frames randomly varies from 64 bytes to 1518 bytes. Self-similar traffic is generated with the Hurst parameter of 0.8. The total traffic load of the entire network is changing from 0.1 to 0.8; 20%, 30%, and 50% of the total traffics are the EF, AF, and BE frames, respectively.
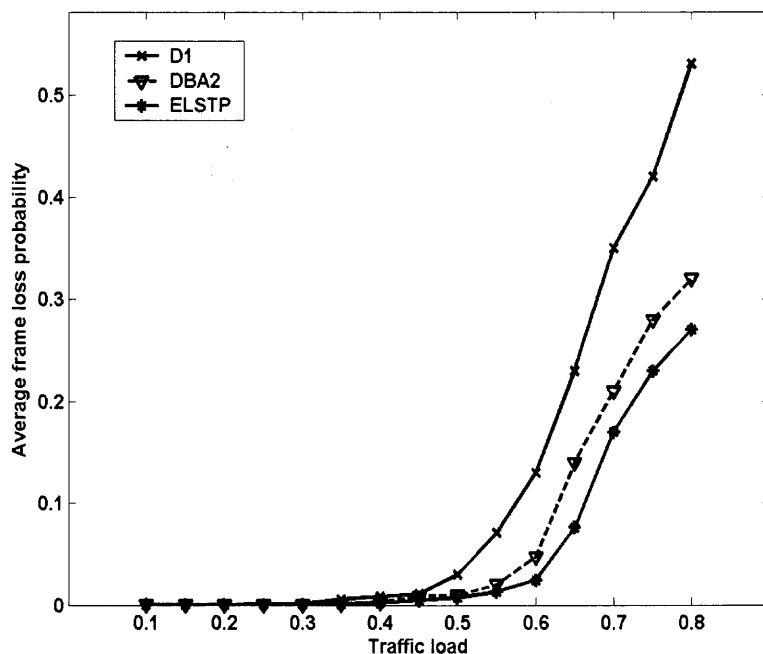
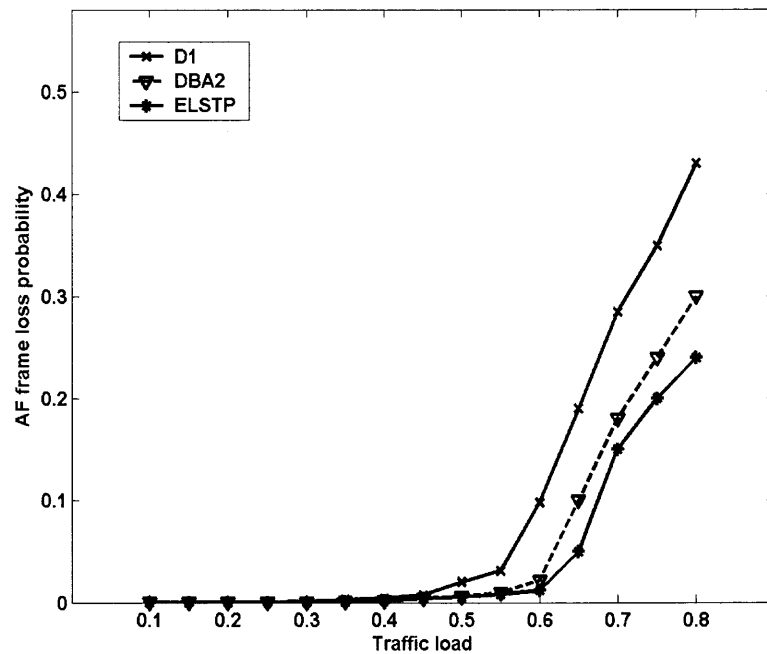**Figure 5.3** Simulations on average frame loss probability *vs.* traffic load.

The compared algorithms include D1 [37], DBA2 [31], and the enhanced LSTP (ELSTP). In D1, the class-based bandwidth allocation is handled by collecting the REPORT messages from all ONUs before making any decision. The OLT assigns a fixed bandwidth to the EF traffics from all ONUs regardless of their dynamics. The leftover bandwidth after having served the EF traffics is distributed among the AF traffics from all ONUs, and, if possible, the BE traffics. In the following simulations, the D1 algorithm reserves 1/3 of the upstream bandwidth to transmit the EF traffic from all ONUs. DBA2 employs EBR with the priority-based scheduling, classifying the incoming frames at an ONU into three classes as ELSTP. DBA2 only forecasts the EF frames, using the amount of actually arrived EF frames during the waiting time of the last service interval, *i.e.*, $\widehat{b^w_{EF,i}}(n+1) = b^w_{EF,i}(n)$. Unlike ELSTP, DBA2 serves the lightly loaded ONUs ahead of the heavily loaded ones, and the heavily loaded ONUs are served after the OLT redistributes the excessive bandwidth.
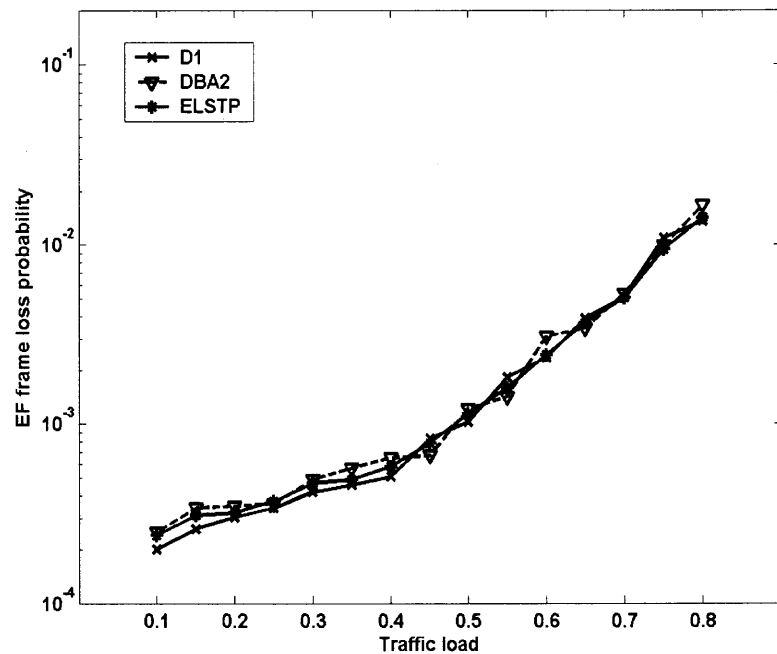
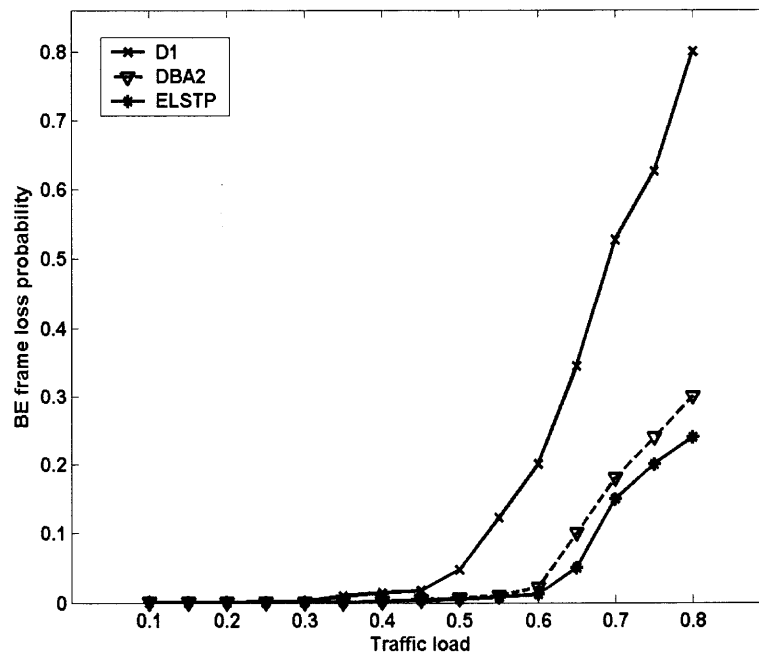**Figure 5.4** Simulations on EF frame loss probability *vs.* traffic load.

Fig. 5.3 illustrates the relationship between the average data loss and the network traffic load. The average frame loss probability is defined as the number of dropped Ethernet frames versus the total number of Ethernet frames. D1 experiences the heaviest data loss, which is attributed to the fact that D1 disregards the data arrived during the waiting time and allocates fixed bandwidth to the EF traffic. Since the allocated 1/3 upstream bandwidth in D1 is enough to transmit the EF data, the traffic load of which is the lightest one among all of the three classes, the EF data can be promptly delivered from the ONUs to the OLT. Hence, in Fig. 5.4, D1 achieves similar EF frame loss probability as compared to that of both DBA2 and SDLSTP. Without taking into account of the SLA parameters and the traffic dynamics, D1 suffers severe data loss of the AF and BE traffics, which are exhibited in Figs. 5.5 and 5.6, respectively.

Figure 5.5 Simulations on AF frame loss probability *vs.* traffic load.

The penalty of "blind" bandwidth allocation in D1 has been alleviated in DBA2 by employing a rough prediction of the incoming EF traffic arrived during the waiting time. The data loss reduction from D1 to DBA2 shows that traffic prediction plays a significant role in data loss control.

ELSTP outperforms both DBA2 and D1 with respect to the average frame loss, AF frame loss, and BE frame loss. The salient features inherited from the basic LSTP scheme contribute to the performance improvement in ELSTP. First, ELSTP predicts all classes of traffics instead of only one class in DBA2 and no traffic prediction at all in D1. Second, the OLT responds to the ONU bandwidth report instantaneously in ELSTP. In DBA2, the heavily-loaded ONUs are served after the lightly-loaded ones, and the deferred service for those heavily-loaded ONUs results in heavier data loss. Besides, the differentiated data process in ELSTP facilitates the achievement of service differentiation in GEPON. From the applicability

Figure 5.6 Simulations on BE frame loss probability *vs.* traffic load.

point of view, the bandwidth negotiation mechanism in ELSTP for reporting and granting three queues are seamlessly compatible with the standard, the priority-based queuing mechanism provides guaranteed service to the higher priority traffic, and the centralized traffic prediction at the OLT side effectively reduces the extra cost for system upgrade.

Figs. 5.7 and 5.8 illustrate the frame delay and queue size performance, respectively. ELSTP outperforms D1 and DBA2 with the shortest frame delay and the smallest queue size, verifying the contribution of the traffic predictors to delay reduction and queue management. As compared to DBA2, the prediction accuracy has been improved in ELSTP by employing the LMS-based predictor, which is suitable for adaptive on-line traffic prediction. Instead, DBA2 predicts the incoming EF traffic at an ONU in a very rough way by simply replacing it with the actual number of incoming EF frames during the last waiting time, ignoring the correlation with other
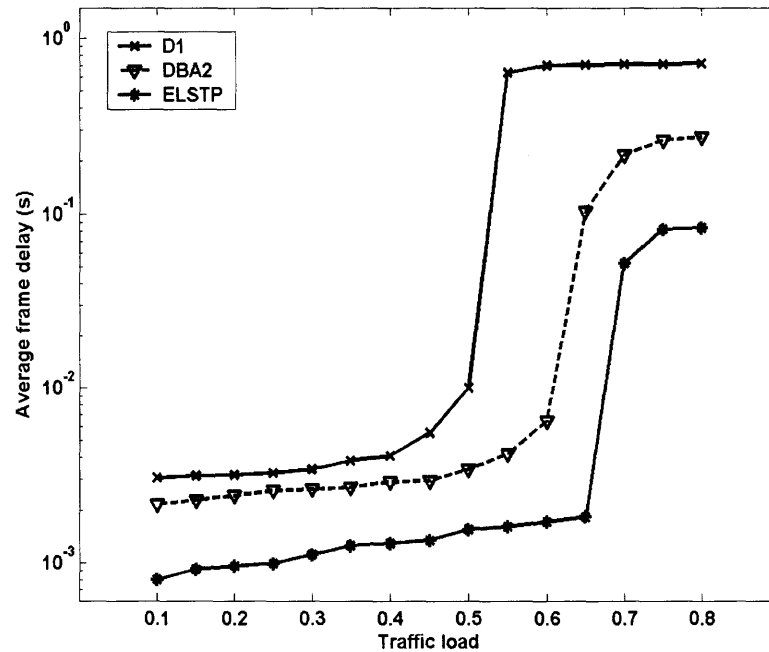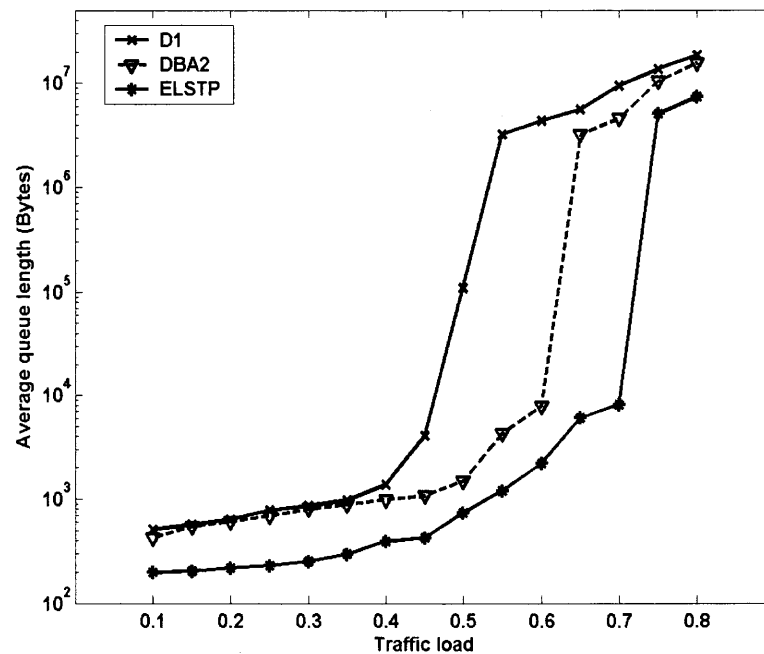
Figure 5.7 Simulations on classified average delay *vs.* traffic load.

previous timeslots. In addition, ELSTP implements the fixed ONU service order instead of the dynamic service order in DBA2, and reduces the drastic change of the service interval length in DBA2, thus facilitating the traffic prediction. Different from in both D1 and DBA2, the OLT responds to the ONU bandwidth requirement instantaneously in ELSTP. In D1, the EF traffics from all ONUs are served first, followed by the AF and BE traffics. In DBA2, the heavily-loaded ONUs are always served after the lightly-loaded ones, and the deferred service for those heavily-loaded ONUs results in longer delay of the incoming frames.

5.6 Chapter Summary

This chapter presents the enhanced LSTP (ELSTP) mechanism, which supports service differentiation over EPONs. Local traffics at an ONU are first classified into three classes according to their delay and bandwidth requirements. Different classes

Figure 5.8 Simulations on classified average queue length *vs.* traffic load.

are treated in different ways. With respect to the bandwidth negotiation, ELSTP includes multiple queue status reports from an ONU in one REPORT message and multiple bandwidth grants to an ONU in one GATE message. At the ONU side, priority queuing is employed, providing precedence to the higher priority traffic as well as upper-bounding the aggressive bandwidth consumption by the queue length limit. At the OLT side, the traffic prediction forecasts the incoming frames arrived during the waiting time per class, and a portion of bandwidth is pre-reserved to transmit them, reducing the delay experienced by these frames. The aggressive bandwidth competition among different classes of traffics is limited by adopting the SLA as the upper-bound. The analysis on class-based frame loss justifies the service differentiation among the EF, AF, and BE traffics. Experimental simulations have demonstrated the performance improvement in ELSTP over other proposals.

CHAPTER 6

CONCLUSIONS AND FUTURE WORK

6.1 Contributions

As an inexpensive, simple, and scalable solution to the access network, EPONs have the capability of delivering integrated broadband services by employing efficient medium access control in the upstream channel. As compared to the point-to-point network and the curb-to-switched network, the P2MP architecture between each OLT and its associated ONUs has the advantages of minimizing the number of optical transceivers and eliminating the intermediate powering.

In the downstream direction, the broadcasting nature of Ethernet perfectly matches the EPON transmission, facilitating downstream multimedia services like video broadcasting and tele-conferencing. In the upstream direction, only a single ONU may transmit during a timeslot in order to avoid data collisions. The bandwidth management of the upstream channel is essential for the successful implementation of EPONs. This dissertation sheds some lights on how to dynamically allocate the upstream bandwidth among ONUs, and how to provide service differentiation to the end users. Original contributions of this dissertation include the following:

- The innovative *limited sharing with traffic prediction* (LSTP) mechanism has been proposed to tackle the issue of dynamic bandwidth allocation over EPONs. Based on the network traffic dynamics, LSTP possesses the following desirable properties: 1) enables the dynamic bandwidth negotiation between ONUs and the OLT by utilizing the REPORT/GATE process, thus making the LSTP-enabled EPON compatible with the IEEE standard 802.3ah; 2) alleviates the delay of the deferred frames by adopting linear predictor for traffic prediction and incorporating a bandwidth reservation strategy for resource allocation;

and 3) avoids the aggressive bandwidth competition among multiple ONUs by employing the SLA parameter and limiting the allocated bandwidth to each ONU.

- Frame loss, frame delay, and queue length have been theoretically analyzed, demonstrating the contribution of the LMS-based traffic predictor. It has been proven that, besides network traffic load, the accuracy of the traffic predictor plays a key factor to determine the network performance. Specifically, the employment of traffic predictor renders explicit control on the loss reduction improvement and bandwidth usage efficiency, lightening the frame loss, lessening the frame delay, and reducing the queue size.

- In order to serve diverse applications, LSTP has been further enhanced with the provisioning of service differentiation. The guidelines include: 1) traffic classification at the ONU-side, which enqueues the incoming traffic with different QoS requirements into different queues; 2) priority queuing at the ONU-side, which works in tandem with a traffic conformance verifier to render the buffer management; 3) class-based traffic prediction at the OLT-side, which facilitates the latency reduction and bandwidth efficiency as in the basic LSTP; and 4) class-based bandwidth allocation, which regulates the aggressive bandwidth competition among different applications. It has been verified, via theoretical analysis and simulation results, that the enhanced LSTP scheme improves the differentiated services without deleterious system costs.

6.2 Future Work

In addition to the above contributions, this dissertation has created the following future research opportunities:

- Extension of the proposed architecture. The future research will continue along this dissertation trajectory of high-speed networking, and will be steered in: 1) the enhancement of the proposed MAC framework with the consideration of downstream data security; 2) the interoperability, contention resolution, and protection/restoration issues for multiple access broadband networks; and 3) the traffic aggregation mechanisms and SLA-based scheduling schemes as well as their impact on the traffic characteristics and network performance.

- Exploration of other broadband network frameworks. Efforts will be devoted to the applicability of the proposed techniques for other access networks, such as G-PONs and WDM-PONs. Specifically, the topics of interest include: 1) a generic infrastructure that accommodates QoS and DiffServ operations through the P2MP access framework; 2) access network reliability and protection/restoration; 3) optical and wireless service integration in the access network; and 4) performance evaluation and improvement.

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