Improved Priority Access, Bandwidth Allocation and Traffic Scheduling for DOCSIS Cable Networks

Wen-Kuang Kuo, Sunil Kumar, and C.-C. Jay Kuo

***Abstract*—The Data Over Cable Service Interface Specifications (DOCSIS) is intended to support IP flows over HFC (Hybrid Fiber/Coax) networks with significantly higher data rates than analog modems and Integrated Service Digital Network (ISDN) links for high quality audio, video and interactive services. To support quality-of-service (QoS) for such applications, it is important for HFC networks to provide effective media access and traffic scheduling mechanisms. In this paper, we first present a multilevel priority collision resolution scheme with adaptive contention window adjustment. The proposed collision resolution scheme separates and resolves collisions for different traffic priority classes (such as delay-sensitive and best effort streams), thus achieving the capability for preemptive priorities. Second, a novel MAC (media access control) scheduling mechanism and a new bandwidth allocation scheme are proposed to support multimedia traffic over DOCSIS-compliant cable networks. It is shown through simulation results that throughput and delay performance have been improved for the transmission of real-time VBR (variable bit rate) traffic as compared to current DOCSIS specifications.**

***Index Terms*—Cable modem (CM), collision resolution, data over cable service interface specifications (DOCSIS), MAC, QoS, scheduling.**

I. Introduction

RECENTLY, the rapid growth of the number of residential Internet users and the increased bandwidth requirements of multimedia applications have necessitated the introduction of an access network that can support the demand of such services. Community Antenna Television (CATV) networks seem to be in an important position for supporting these services, at least from an economic perspective. There are two major reasons for this statement. First, CATV infrastructures already connect a majority of homes. Second, the Hybrid Fiber Coax (HFC) used in CATV networks can be used to deliver broadband services without requiring costly upgrade of existing CATV network systems. We can thus foresee that CATV networks are in an important position to support broadband access networks in the near future. However, CATV networks have been traditionally used to provide analog audio and video broadcast programs from the headend (HE) to subscribers. In the downstream direction, CATV networks are characterized by a point-to-multi-point tree and branch topology with the broadcasting node at the root and recipients at leaves. The bandwidth is divided into several channels, most of them dedicated to downstream transmission (from the HE to CMs) while only a few are for upstream transmission (from CMs to the HE). Since all users connecting to the fiber node through the same coaxial cable have to share the upstream channel, it is necessary to identify an effective MAC protocol to make efficient use of CATV networks.

The Data Over Cable Service Interface Specifications (DOCSIS) [1], [2] is the dominant specification for carrying data over CATV networks. Other CATV standard activities include those in IEEE 802.14 [3], [4], the Internet Engineering Task Force (IETF) [5], the Digital Audio Visual Council (DAVIC), the Digital Video Broadcasting (DVB) [6], the ATM Forum Residential Broadband Working Group (RBWG) [7] and the Society of Cable Telecommunications Engineers (SCTE) [8]. DOCSIS has been developed by CableLabs and MCNS (Multimedia Cable Networks Systems), which is a group of major cable companies, to support IP flows over HFC networks. RBWG investigates the provision of ATM for media distribution within the CATV networks. IETF is contributing to IP delivery on top of CATV networks. DOCSIS 1.0 was accepted as an international specification by ITU SG9 in 1998. DOCSIS defines modulation and protocols for high speed bi-directional data transmissions over cable systems. It has also been accepted by most major vendors and is now a widely used specification to provide high-speed residential access. Subsequently, an enhanced specification, known as the DOCSIS 1.1, provides improved flexibility, security and quality-of-service features. Recently, CableLabs has finalized DOCSIS 2.0 that provides up to 30 Mbps throughput in upstream and 50 Mbps throughput in downstream directions respectively. This advanced specification will be compatible to previous DOCSIS versions.

One of the challenging issues for DOCSIS is to support the QoS requirements of delay sensitive interactive multimedia applications such as computer games, video telephony on the upstream channel. These applications demand a very stringent delay bound and may allow VBR traffic. As explained in Section III, an effective priority mechanism is needed to give a higher priority to these applications to minimize access delay during the period of high contention. The DOCSIS 1.0 specification does not employ a priority mechanism. Priority mechanisms have been implemented in other MAC protocols such as DQDB [9] and the token ring [10]. However, these priority mechanisms, which are collision-free, are not suitable for

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the contention-based CATV networks [11]. Lin and Gampbell [12] proposed the amendment of Extended Distributed Queue Random Access Protocol (XDQRAP) that adds an extra slot to each frame to identify priorities. However, this scheme only supports two priorities with a fixed frame format. Citta and Lin [13] implemented a priority scheme with variable probabilities incorporated with the p-persistence random access protocol. However, this scheme cannot be applied to DOCSIS since the collision resolution procedure is not random p-persistence. The preemptive scheduling scheme [14], which is widely used in time-sharing system, does not alleviate this problem either. If an HE uses preemptive scheduling schemes to assign data slots to CMs, CMs must contend for the channel to send their bandwidth requests before the HE receives bandwidth requests. If the high priority CMs cannot transmit their bandwidth requests in time, the access delay is still very high. Qiu and Li [15] proposed three types of multiple access schemes for wireless networks. These multiple access schemes only provide two priorities, *i.e.*, voice and data. Furthermore, these schemes are suitable for wireless networks but may not behave well for cable networks. In this paper, we propose two schemes to implement an effective priority mechanism. (i) The HE uses a multiple priority queue scheduler to allocate the bandwidth to CMs of different priorities. (ii) The MAC protocol uses a dynamic backoff window scheme to resolve collisions so that higher priority CMs are able to transmit bandwidth requests without interference from lower priority CMs. The proposed scheme can be easily integrated with DOCSIS, as explained in Section III.

Since real time VBR traffic such as MPEG video is bursty, it may sometimes be quite challenging to support its QoS requirements (*i.e.*, varying bandwidth and delay bound) over the upstream channel in cable networks. DOCSIS 1.1 defines six QoS classes to support requirements of various types of applications. As explained in Section IV, these classes are not adequate to meet the QoS requirements of real-time VBR traffic. To the best of our knowledge, there has been little research on the transmission of real-time VBR traffic over DOCSIS, especially in the QoS area except for work presented in [16]–[19]. Most of them are not compatible with the DOCSIS 1.1 specification or are designed only for IEEE 802.14. In this work, we present a novel QoS MAC scheduling mechanism by proposing a new QoS class called the Unsolicited Grant Piggybacked Service that provides QoS guarantees for real-time VBR traffic transmission over cable networks. The proposed scheme allocates a constant bit rate (CBR) bandwidth for a certain fraction of real-time VBR traffic. By exploiting the self-similar nature of real-time VBR traffic [20], the CBR bandwidth is predicted based on the past history. The VBR portion of the transmission bandwidth for the remaining fraction is allocated on demand. Our scheme ensures high bandwidth utilization and low latency for real-time VBR traffic. The remaining bandwidth not used by CBR and VBR services could be fairly shared by all CMs to transmit available bit rate (ABR) traffic.

The rest of this paper is organized as follows. Section II provides an overview of DOCSIS specifications and CATV networks. Section III describes the proposed dynamic backoff window scheme and the proposed scheduling algorithm with multiple priority queues. Section IV discusses the proposed new MAC scheduling service and the bandwidth allocation algorithm. Simulation results and the performance comparison are presented in Section V. Section VI provides concluding remarks.

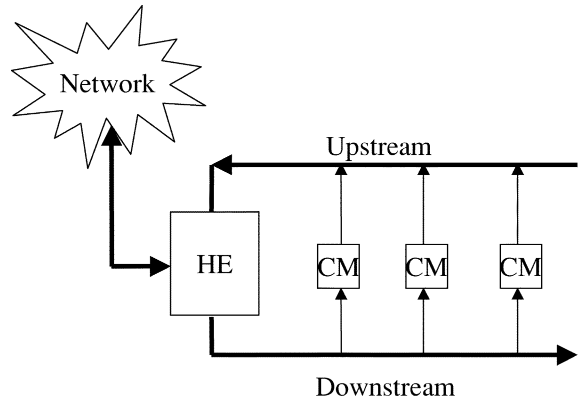

Fig. 1. The logical topology of a CATV network [23].

II. Overview of DOCSIS MAC Layer Operations and CATV Networks

Perkins and Gatherer [23] and Sherali *et al.* [24] gave a very good review of the DOCSIS specification. The logical topology of a CATV network is shown in Fig. 1. The downstream path (from the HE to all CMs) uses a 6 MHz channel selected by the cable operator. Since the HE alone transmits data in the downstream direction, no downstream MAC mechanism is needed. Each subscriber unit is assigned an individual ID that allows it to filter out any downstream data not addressed to it. Each upstream channel is shared by a number of subscriber units known as stations. It is therefore divided in time into individually numbered allocation units called minislots. The MAC mechanism is required to coordinate transmissions along the upstream channel. Since the physical equipment of the cable plant requires the isolation of signals in the upstream direction, upstream transmissions can only be heard by HE but not by other CMs. Thus, concurrent upstream station transmissions from CMs to the HE can collide while the individual CM may not be aware of it.

The MAC protocol allocates the available bandwidth of the shared upstream channel to CMs. The MAC operations are split between CMs and the HE controller. Most of bandwidth allocation and traffic scheduling operations are regulated by the HE. The upstream bandwidth is allocated in granular units called minislots, which are groups of eight octets and identified by a unique number. Control of minislot's type and number is done by the HE. As described below, both upstream timing and MAC scheduling are closely related to the service requirements of CMs, and the HE and CMs must work together to determine proper operations.

In DOCSIS, subscriber units can be located as far as 100 miles from the HE. Typical distances however are 10–15 miles [1]. According to the DOCSIS specification, the MAC protocol must handle propagation delays no greater than 800 ms in each direction. Such large delay can cause a problem in the upstream as

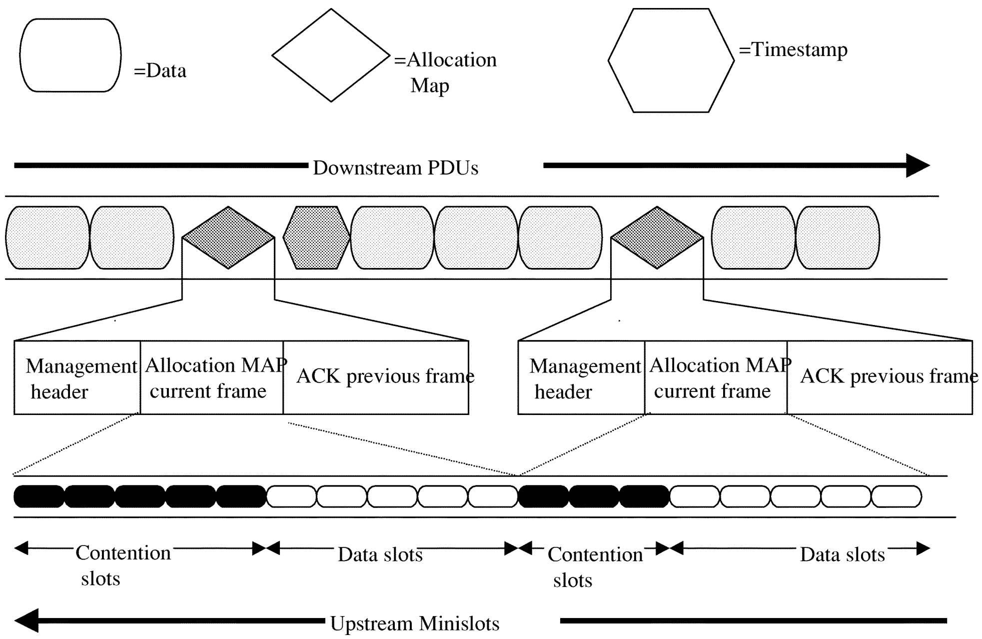

Fig. 2. The upstream minislots allocation.

the data from two widely separated CMs, which start transmission during consecutively numbered upstream minislots, could arrive at the HE out of order or in a collision state. To solve this problem, upstream timing is adjusted by each station so that if two stations transmit during the same minislot, both their transmissions will arrive at the HE at the same instance. At startup, each CM MAC determines its upstream timing adjustment value through a procedure known as "ranging".

During normal operations, the HE regularly sends control signals in the downstream channel, which contain MAP messages to describe the allocation of upstream bandwidth. Any station that desires to request an allocation must contend for access during periods specified in this MAP message with short minislot-sized messages that contain the station's id and the number of minislots needed. If successful, the HE will allocate a proper portion of the upstream bandwidth for the station in a future allocation MAP message. Once a subscriber unit receives a bandwidth allocation in the upstream MAP message, it has the opportunity to piggyback new allocation requests in its reserved upstream allocation. Piggybacking allows stations to request more bandwidth without reentering the contention-based request process. If, however, multiple CMs select the same allocation request minislot, a collision occurs and MAC initiates a collision resolution process. Fig. 2 illustrates the upstream mapping.

In DOCSIS 1.1, QoS is supported in both upstream and downstream flows by specifying more classes of service flows. Usually, data packets entering the HFC network are classified into service flows based on their QoS requirements. QoS may be guaranteed by shaping, policing, and/or prioritizing data packets at both CMs and the HE. DOCSIS defines the following six QoS service classes.

- Unsolicited grant service (UGS).

  The aim of UGS is to reserve guaranteed upstream transmission bandwidth for traffic flows. Upon receiving the request from a CM, the HE schedules fixed size grants at periodic intervals to the UGS flow. The CM needs to send the request only once, and then it is the responsibility of the HE to control the timing of allocated grants to satisfy the required delay and delay-jitter bounds. UGS can thus provide deterministic QoS guarantees. However, the reserved bandwidth may be wasted when a corresponding UGS flow is inactive.
- UGS with activity detection (UGS-AD).

  For UGS-AD flows, the HE employs an activity detection algorithm to examine the flow state. When an UGS-AD flow is active, the HE provides periodic grants to it. If the flow is in an inactive state, the HE only provides periodic request polling.
- Real-time polling service (rtPS).

  The rtPS flow is used to reserve the transmission opportunities for real time variable bit rate (rt-VBR) applications. Upon receiving the request from the rtPS flow, the HE polls the flow periodically so that the flow can send its bandwidth request even when the network traffic is congested. The rtPS traffic can provide statistical QoS guarantees and high network utilization.
- Non-real-time polling service (nrtPS)).

  Both nrtPS and rtPS flows are polled through periodic requests from the HE. However, nrtPS flows receive few request polling opportunities during network congestion while rtPS flows are polled regardless of the network load. The objective of nrtPS is to support nonreal-time applications such as FTP.
- Best Effort (BE).

  The BE service is used to provide transmission opportunities for the best effort traffic. For the BE service, a station must use the normal reservation mode or the immediate access mode to gain upstream bandwidth.

- Committed information rate (CIR).

  The CIR service can be defined by vendors in a number of different ways. For example, it could be configured by using the nrtPS service with a reserved minimum traffic rate.

To meet the QoS requirements, the HE must adopt an admission control mechanism and a scheduling algorithm among different services to reduce the QoS violation probability. Each QoS flow matches exactly one QoS service. If a station has a special bandwidth requirement not specified in the QoS service profile, it could dynamically request a service by sending a dynamic service addition request to the HE. Moreover, after a QoS flow is established, the payload header suppression mechanism can be adopted to efficiently utilize the bandwidth by replacing the repetitive portion of payload headers with a payload header index.

III. Multi-Priority Access Scheme

A. Motivation and Problem Description

The DOCSIS specification requires the subscriber unit to follow a truncated binary exponential backoff algorithm to access minislots in the group. The HE dynamically controls parameters necessary for the algorithms. The truncated binary exponential backoff algorithm requires two parameters for its operation, *i.e.,* the initial back-off window and the maximum back-off window. These values are set as a part of the bandwidth allocation MAP messages. To begin a request, the CM sets its window to the size of the initial back-off window. Then, it chooses a random value that is within the window range. Once a value is chosen, the station must let that many allocation request minislots pass before it makes its request. If a CM does not receive a response to its request before a timeout value, it assumes a collision. The CM then increases its window size by a factor of 2 (as long as it is less than the maximum back-off window size) and retries. This process can be repeated for a maximum of sixteen times.

The access delay, *i.e.,* the time it takes for a CM to successfully send a request to the HE, must be kept as low as possible for interactive rt-VBR flows, even during high contention periods. The time spent in the contention process includes that for collision, retransmission, etc. An effective priority mechanism is therefore needed in CATV networks to support QoS demands (delay constraints in particular) of real-time interactive services such as interactive computer games, video telephony, and so on.

Although the DOCSIS protocol supports six upstream QoS services as described in Section II, some problems still exist. First, DOCSIS has a provision to assign eight priority traffic classes to a service flow, but it does not specify an algorithm to implement it. Second, it does not separate and resolve collisions according to the priority order of traffic. In fact, DOSCIS treats all CMs equally during contention. As a result, new users of high priority traffic (*e.g.,* delay sensitive applications) may be blocked for an extended period of time during congestion, resulting in unacceptably large access delay.

To solve this problem, we introduce a scheme to support multi-priority access system for DOCSIS. In the proposed scheme, higher priority CMs are assigned contention slots more easily with a higher probability for initial access as well as retransmissions. This is discussed in detail below.

B. Dynamic Backoff Algorithm

We propose a new dynamic backoff scheme for multiple priority traffic over DOCSIS cable networks in this work. A CM with a new request transmits its request with probability 1 when a group of contention slots (*i.e.,* window) corresponding to its priority becomes available. The station randomly selects a contention slot in the window. As described in Section II. A, the DOCSIS protocol uses a truncated binary exponential backoff scheme to resolve collisions. We propose modifications to this collision resolution scheme by allocating a backoff value to CMs, which is inversely proportional to the priority. In other words, a lower backoff value is allocated to CMs of higher priority. For different priority CMs, the backoff value is governed by the reserved number of contention slots for their corresponding priority class. Furthermore, the number of contention slots for a given priority class is determined according to traffic conditions to decrease contention collision and achieve lower access delay for high priority traffic. Thus, the binary exponential backoff scheme can be improved by dynamically selecting the backoff value according to the estimate of the number of contending CMs with the measurements of the channel activity performed by the HE. The proposed dynamic backoff window scheme, in which the size of backoff window is chosen based on the traffic priority and the number of contending CMs, will be described and the optimal backoff value that depends on the number of contending CMs will be derived in this section.

Let us consider n greedy stations (*i.e.,* stations that always have packets to transmit) that are contending for requested minislots. For a given contention window W, the backoff value $b(t)$ at time slot t is randomly chosen in the range $(0, W-1)$, where W is the initial value of the contention window. The $b(t)$ is decreased by an unit value in each subsequent time slot. This can be modeled by the following Markov Chain,

$$\begin{aligned} \Pr\{b(t+1)=k\} &= Pr\{b(t)=k+1\} \\ &\quad + \frac{Pr\{b(t)=0\}}{W}, \\ 0 \le & k < W-1, \end{aligned} \tag{1}$$

$$\begin{aligned} \Pr\{b(t+1)=W-1\} &= \frac{Pr\{b(t)=0\}}{W}, \\ k =& W-1, \end{aligned} \tag{2}$$

where k represents backoff delay and takes integer values from 0 to $W-1$. The value $Pr\{b(t)=0\}/W$ accounts for the fact that, after one transmission, the new backoff value is uniformly chosen from the range between 0 and $W-1$. We express the steady state probability $\lim_{t\to\infty}\Pr\{b(t)=k\}$ as p_k. By using (1) and (2), and the fact that $\sum_{k=0}^{W-1} p_k = 1$, we can obtain the steady state probability as

$$p_k = \frac{2(W-k)}{W(W+1)}. \tag{3}$$

TABLE I

If $(CS > \sum_{i=1}^{M} \hat{W}_i)$ **then**
//Compute additional contention slots S_i to be allocated to traffic class i
    $S_i = \hat{W}_i(CS - \sum_{i=1}^{M} \hat{W}_i)/\sum_{i=1}^{M} \hat{W}_i$
    $CS = \hat{W}_i + \hat{S}_i$
Else
//Compute total No. of available additional slots
    $S = CS - \sum_{i=1}^{M} G_i$
    $CS_M = G_M + min(S, \hat{W}_M - G_M)$
    If $(S < \hat{W}_M - G_M)$ **then**
        $CS_i = G_i$, for $i = 1, 2, ..., M - 1$
    Else
//Allocate remaining slots to different traffic class in the decreasing order of their priority
        $CS_{M-1} = G_{M-1} + min(S - (\hat{W}_M - G_M), \hat{W}_{M-1} - G_{M-1})$
        If $(S - (\hat{W}_M - G_M) < \hat{W}_{M-1} - G_{M-1})$ **then**
            $CS_i = G_i, fori = 1, 2, ..., M - 1$
            repeat this procedure until the lowest priority class

Since a station transmits data in a time slot when its backoff value becomes zero, the transmission probability is $p_0 = 2/W + 1$. The probability that a transmission is successful, denoted by p_s, is given by the probability that only one station transmits in a given time slot. If n CMs are contending for requested minislot, then we have

$$p_s = np_0(1 - p_0)^{n-1}. \tag{4}$$

To optimize p_s, we set $dp_s/dW = 0$ and assume that $(1 - p_0)^n = 1 - np_0$. From this, we compute the size of the backoff window as $W = 2n - 1$. Although the HE does not know the exact number of contending CMs, it can however be estimated based on the number of observed collisions.

If c is the number of collisions observed by the HE over W requested minislots, the collision probability p_c is given as

$$p_c = \frac{c}{W} = [1 - (1 - p_0)^n] - np_0(1 - p_0)^{n-1}. \tag{5}$$

From (5), we obtain

$$\hat{n} = \frac{1 + \sqrt{\frac{1+c(W+1)^2}{W}}}{2}, \tag{6}$$

where $\hat{n}$ is the estimate of n based on the number of observed collisions. Using the value of $\hat{n}$ from (6), we can estimate the value of the backoff window as

$$\hat{W}(t) = 2\hat{n} - 1. \tag{7}$$

In order to provide a smoother behavior of the estimate, we can take a weighted average of $\hat{n}$ over time, *i.e.*,

$$\bar{n}(t + 1) = (1 - \alpha)\bar{n}(t) + \alpha\hat{n}(t + 1), \tag{8}$$

where α is a weighting parameter and $\bar{n}$ is the value used to compute the adaptive backoff window $\hat{W}$ for each priority.

The proposed dynamic backoff window algorithm for multiple traffic classes is summarized in Table I, where the following definitions are adopted

- CS: the total number of contention minislots in each frame
- i: traffic class $(i = 1, 2, \ldots, M - 1)$ with the priority in an ascending order, *i.e.*, the priority for class j is higher than that for class i, if $j > i$
- CS_i : contention minislots assigned to traffic class i
- $\hat{W}_i$: estimate of the backoff window for traffic class i obtained from (7)
- G_i : the number of guaranteed CS for each priority, which is set by the network operator to provide the minimum service access

It is assumed here that $G_i < \hat{W}_i$.

C. Multiple-Priority-Queue Scheme

Typically, the HE scheduler serves each CM on a first-come-first-serve basis. In this work, we employ the multiple priority-queue scheme [25], where the traffic at each CM is identified by its priority class. We use eight priorities $(M = 8)$ as specified in the DOCSIS protocol. The multiple priority-queue scheme consists of different queues at the HE depending on the priority of each traffic class at different CMs. When a new request arrives at the HE, it waits in the corresponding queue. When the time

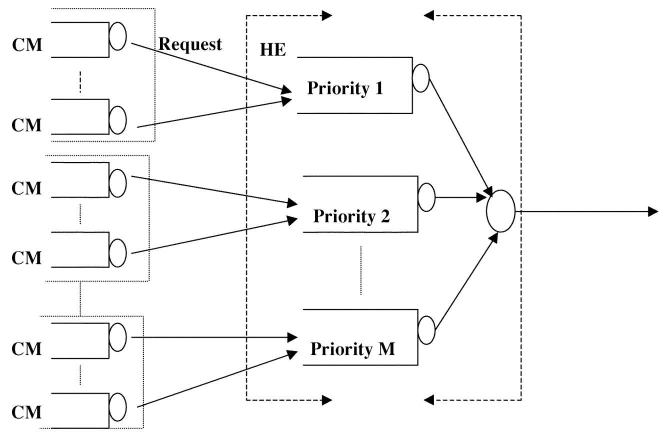

Fig. 3. The priority scheduler in DOCSIS.

comes for the next MAP message, the HE computes a horizon of all events in the time order that can be scheduled, and requests in the queues are then served in the priority order. Requests are granted completely or not at all. When a CM has data to transmit, it sends its request according to the proposed dynamic backoff window scheme and waits for the allocation of the requested bandwidth in a subsequent MAP message. If the request is successful, then the CM will be notified with the minislot number in which it can start transmission and the number of minislots assigned to it. Otherwise, the CM must repeat its request attempt based on the proposed adaptive backoff window scheme. Fig. 3 shows the proposed priority scheduler under the DOCSIS environment.

To implement the proposed scheme, we discuss modifications needed in the current DOSCIS 1.1 protocol and show that our scheme can be easily supported by the specification below. The proposed priority scheme has the following features. First, a CM is allowed to request a priority level during registration. We use a concept that is similar to service negotiation used during the connection setup in ATM networks. When a CM enters the network, after acquiring synchronization and completing the ranging and power leveling steps, it sends a registration request message to the HE. The DOCSIS protocol can support several user-defined registration request classes that allow the station to request different types of services through the "MAC management message header" as shown of the DOCSIS document [1, Table 6.17]. We use the reserved 'type value' field to implement the priority service. The HE can associate a priority level with a station according to the priority value specified in the registration request message. Second, the HE marks the contention slots with a priority value as shown in Fig. 4, according to the dynamic backoff algorithm described above. For each group of contention slots, the HE specifies a priority value, the start time of the group in the next frame and acknowledgment (ACK) for the contention slots transmitted in the previous frame. The fields of "Data Backoff Start" and "Data Backoff End" are used to specify the backoff value for each priority group. Thus, the HE can associate a CM with a priority according to the priority value contained in its registration request message.

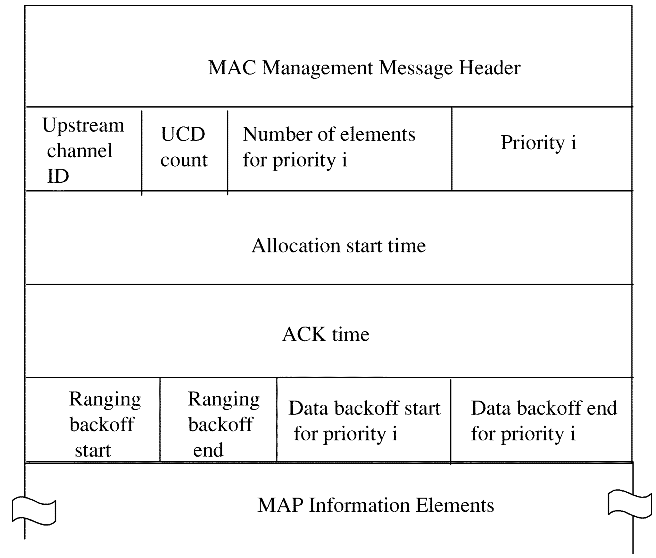

Fig. 4. The modified allocation MAP message.

IV. New Service Class and Bandwidth Allocation

A. Motivation

DOCSIS 1.1 uses two scheduling classes, *i.e.*, unsolicited grant service (UGS) and real-time polling service (rtPS), to support real-time upstream traffic transmission. The UGS scheduling service is intended to support the CBR traffic transmission over the upstream channel. For UGS services, a traffic flow will be granted the periodic data transmission opportunities without having to request for it. For example, a 64 kbps voice application (CBR) can be transmitted using a UGS service flow with the 10 ms nominal grant interval and the 80-byte grant size. The real-time VBR video stream is however not suitable to be served by UGS since its bandwidth request is variable and bursty.

DOCSIS defines the real-time polling service class (rtPS) to serve real-time VBR traffic. The rtPS flows receive periodic transmission opportunities regardless of network congestion. A service flow with rtPS scheduling is allocated periodic request opportunities, in which upstream bandwidth requests can be transmitted. Basically, each CM gets polled by the HE to find out about the instantaneous upstream bandwidth requirements for its data. For example, a real-time VBR video stream with an average bit rate of 6.4 Mbps can be transmitted in an rtPS flow with request opportunities granted every 10 ms. In this case, the CM will request a data transmission opportunity for 8 Kbytes on the average. The drawback of the rtPS class (or the delay sensitive real-time interactive service) is that a CM may experience relatively longer delay since it has to wait first for the HE to poll for the bandwidth request and then for the HE to respond for available slots.

To eliminate delay due to polling, we propose a new service class called the "unsolicited grant piggyback request service" (UGPS). UGPS provides very low delay, which is close to that of UGS, yet with good channel utilization, which is similar to rtPS.

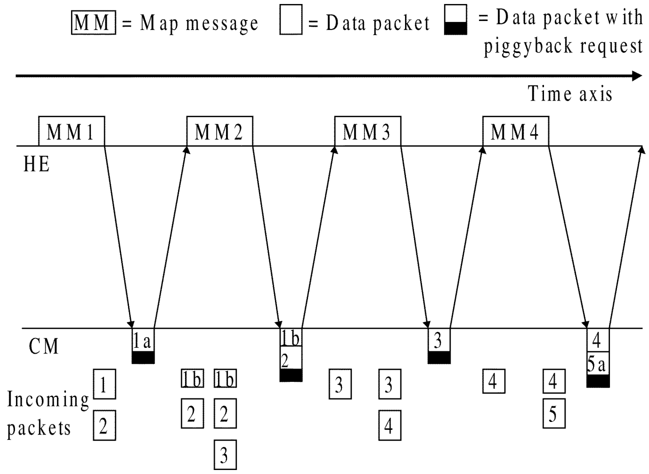

Fig. 5. The UGPS scheduling service time diagram.

B. New UGPS Class

In DOCSIS 1.1, the request/transmission policies of both UGS and rtPS prohibit piggyback requests. However, the specification adopts piggyback request mechanism in fragmenting frames. When fragmentation is enabled and the grant size is smaller than the request, the CM uses the piggyback field in the fragment extended header to request the bandwidth necessary to transfer the remainder of the frame. It is the responsibility of the CM to keep track of the remainder to send. The proposed UGPS service treats the real-time VBR traffic in a way similar to CBR [21] by employing the piggyback mechanism [22]. The proposed UGPS class reserves a fraction of the average VBR traffic's bit rate required. This reserved part, which is called unsolicited allocation, is allocated to the service flow using an algorithm discussed in Section IV-C by the HE at periodic time intervals. This is similar to the service of the UGS class. The bandwidth requests for the remaining VBR portion of the traffic are then piggybacked in the data grant slot. The HE will process these piggybacked requests and issue the corresponding data grants indicated by MAP messages.

The idea of UGPS can be illustrated by the time diagram as shown in Fig. 5. The first data unit (packets 1 and 2) are bigger than the available unsolicited grant. Here, a portion of the packet, denoted by (1a), is transmitted in the unsolicited data grant slot, along with a the piggybacked request for (1b) and (2). In the next MAP message *i.e.*, MM2, the HE allocates the unsolicited grant and an additional transmission opportunity in response to the piggybacked request. The fractions (1b) and (2) are then transmitted in the allocated data grant slots, along with a piggyback request for (3), and so on.

To give an example, a VBR MPEG stream with a 6.4 Mbps average bit rate could be transmitted in UGPS class by sending 3.2 Mbits/s using unsolicited grants (at a 20 ms nominal grant interval with an unsolicited grant size of 8 Kbytes), and piggybacked requests for the remaining VBR portion of data. The unsolicited grant portion here is kept lower than average bit-rate to get higher channel utilization. Low latency is achieved by using piggyback requests for the VBR portion of the stream instead of requesting the bandwidth from the HE during the nominal polling intervals.

C. Bandwidth Allocation Algorithm

Since the bandwidth demand of the VBR traffic varies with time, we propose a scheme in which the HE dynamically adjusts the unsolicited allocation based on the following update

$$\begin{aligned} &unsolicited\_allocation(n+1)\\ &= unsolicited\_allocation(n)\\ &\quad - unused\_bytes(n)\\ &\quad + piggyback\_request(n) \end{aligned} \tag{9}$$

where n is the index of the MAP message cycle, $unused\_bytes(n)$ is the average unused bandwidth of the unsolicited allocation portion in the previous N MAP message cycles, and $piggyback\_request(n)$ is the average value of the piggyback request portion transmitted over the previous N MAP message cycles. Real-time VBR data, such as coded MPEG video streams, exhibit strong long-range dependence and are bursty over multiple time periods. This is known as the "self-similar" traffic [20]. The bandwidth reservation in the current and N previous scheduling cycles can therefore be used to estimate the required bandwidth for the next scheduling cycle. Sometimes, the value of $unsolicited\_allocation(n+1)$ in (9) may be equal to zero. In this case, the value of $unsolicited\_allocation(n+1)$ is set to one time slot to avoid the contention and/or polling delay.

In each MAP message, the HE computes the total bandwidth requirement of all VBR streams. If the sum of $unsolicited\_allocation$, *i.e.*, the left hand side of (9), of all VBR streams exceeds the channel capacity, we propose the following bandwidth allocation algorithm.

Initially, the total residual bandwidth, $initial\_ResBW$, is set to the maximum available upstream bandwidth. The HE computes the initial average residual bandwidth, $initial\_Avg\_ResBW$, which is equal to the $initial\_ResBW$ divided by the total number K of VBR streams. If the $unsolicited\_allocationBW$ is more than $initial\_Avg\_ResBW$ for a VBR stream, then the HE only allocates Avg_ResBW to it. Otherwise, the $unsolicited\_allocation$ is allocated. In the latter case, the excess bandwidth allocated to each stream i is computed as the $initial\_Avg\_ResBW$ minus $unsolicited\_allocationBW$. These excess bandwidths for all streams are added to obtain a new value of residual bandwidth $(ResBW)$ available with the HE. The Avg_ResBW is then computed as the $ResBW$ divided by the number of VBR streams (*i.e.*, streams which demand $unsolicited\_allocation$ higher than Avg_ResBW). In the second iteration, the Avg_ResBW is allocated to each VBR stream in the same way. This process is performed repeatedly until $ResBW$ is equal to zero. The detailed algorithm is presented in Table II.

V. Simulation Results and Discussion

Cablelabs and OPNET have jointly developed a common simulation framework (CSF) for simulation studies related

TABLE II

```
For (each VBR stream i, i = 1, 2, ..., K) {
      If (end of each MAP message cycle) {
          measure average unused_bytes (of previous N MAP message cycles)
          measure average piggyback_request (of previous N MAP message cycles)
          unsolicited_allocation(n + 1) = unsolicited_allocation(n)
          −unused_bytes(n) + piggyback_request(n)
        If (unsolicited_allocation(n + 1) <= 0)
            unsolicited_allocation(n) = 1
      }
}
If (∑_{i=1}^{K} unsolicited_allocation(i) > channel capacity) {
      Step 1. k = K
                Set S = 1, 2, ..., K
      Step 2. initial_ResBW = max upstream bandwidth available for VBR streams
            initial_Avg_ResBW = initial_ResBW/K
            For (each VBR stream i) {
              allocated_BW(i) = Min(unsolicited_allocation(i), initial_Avg_ResBW)
              If (allocated_BW(i) < initial_Avg_ResBW) {
                S = S − i
                k = k − 1
              }
            }
      Step 3. ResBW = ∑_{i∈S}(initial_Avg_ResBW − allocated_BW(i))
            If (ResBW ≠ 0) {
              Avg_ResBW = ResBW/k
              For (VBR stream in S) {
                available_BW(i) = allocated_BW(i) + Avg_ResBW
               allocated_BW(i) = Min(unsolicited_allocation(i), available_BW(i))
                If (unsolicited_allocation(i) < available_BW(i)) {
                  S = S − i
                  k = k − 1
                }
              }
            }
            ResBW = ∑_{i∈S}(initial_Avg_ResBW − allocated_BW(i))
            If (ResBW ≠ 0)
              Repeat this process (step 3) until ResBW = 0
          }
```

to DOCSIS with the modeler simulation tools in OPNET. We have modified this simulation system slightly to meet our need. A summary of simulation parameters is given in Table IV.

If the estimated backoff window is smaller than G_i, G_i minislots are assigned to class i, and the remaining minislots are assigned to other priority traffic. In our simulation, the network consists of 60 CMs that are divided into three priority classes. Among them, 10 CMs generate real time traffic and are assigned the high priority, 20 CMs generate nonreal time traffic with the Poisson source and are assigned the medium priority and, finally, 30 CMs generate nonreal-time traffic with the Poisson source and are assigned the low priority. Simulation results are given below to demonstrate the effectiveness of the proposed priority-based scheme. Both the access delay and the throughput of the three classes of CMs are compared. Note that the access delay is the time a packet takes to reach the HE from the time it is initially requested by its CM.

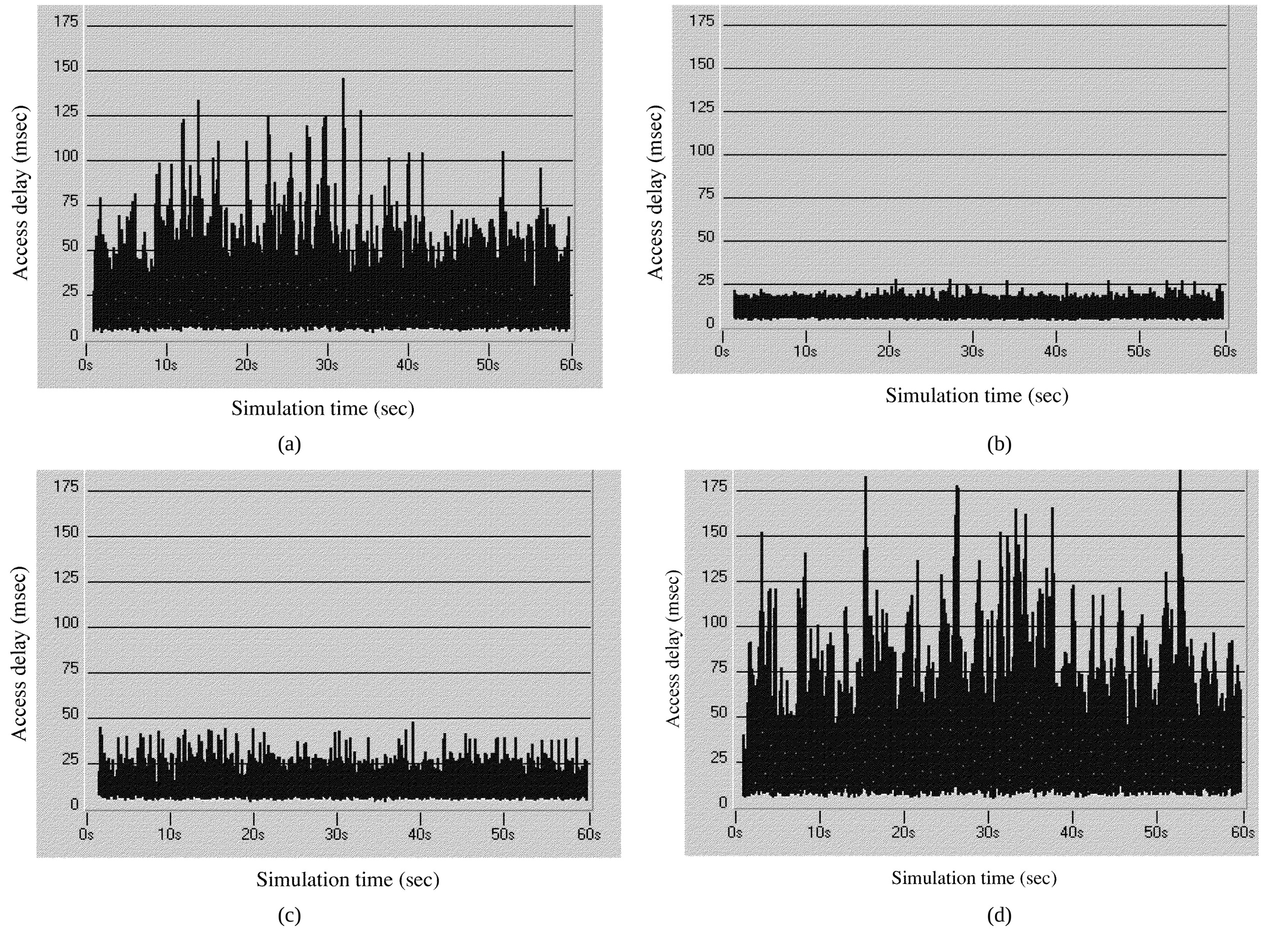

Fig. 6. The comparison of access delay performance for (a) DOCSIS system using no-priority, and (b) high-priority traffic, (c) medium priority traffic, and (d) low-priority traffic in the proposed scheme.

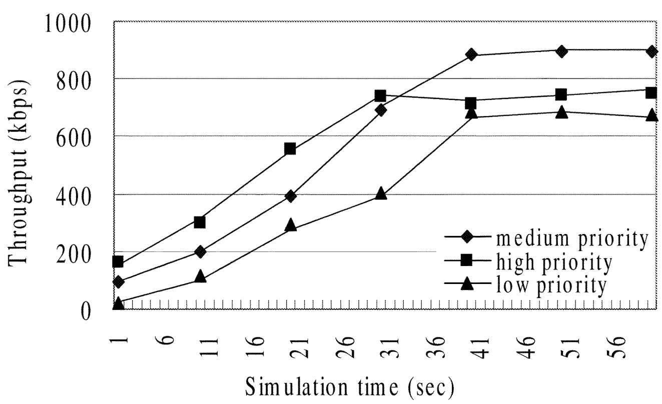

Fig. 7. The throughput performance of the proposed priority scheme.

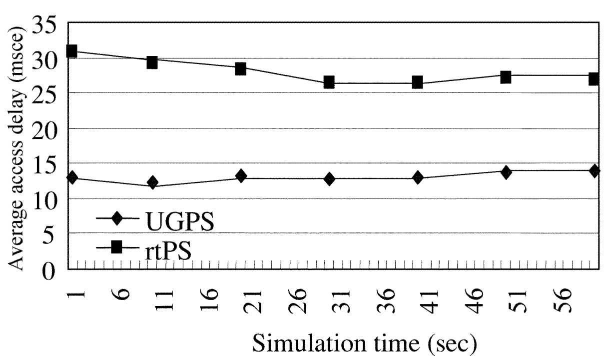

Fig. 8. Access delay for the UGPS and rtPS service classes.

TABLE III
SIMULATION PARAMETERS

upstream data rate	2.56 Mbps
Minislot size	16 bytes
Frame size	200 minislots
N (MAP message cycle)	5
Simulation run	60 sec
number of CBR CMs	10
number of VBR CMs	20
number of ABR CMs	30
CBR average bit rate	60 kbps
VBR average bit rate	80 kbps
VBR peak bit rate	120 kbps
ABR peak bit rate	40 kbps

A comparison of the mean access delay for the current DOCSIS system that does not use priority and the proposed scheme that employs priorities is shown in Fig. 6, where the X and Y axes represent the simulation time (in seconds) and the access delay (in ms), respectively. In the proposed priority-based scheme, although the access delay for low-priority traffic is higher than that for original DOCSIS system, the access delays for high and medium priority traffic are much lower (about 20 ms and 25–40 ms, respectively). As expected,

TABLE IV
SIMULATION PARAMETERS

upstream data rate	2.56 Mbps
number of contention slots/frame	32
number of high priority CMs	10
number of medium priority CMs	20
number of low priority CMs	30
minislot/contention slot size	16 bytes
frame size	500 minislots
number of contention slots guaranteed per priority	$G_h = 2, G_m = G_l = 1$
simulation run	60 sec
high priority CBR traffic (ON/OFF source) [26]	mean off time = mean on time = 1 sec, average bit rate = 80 kbps per CM
medium priority (Poisson source)	average bit rate = 50 kbps per CM
low priority (Poisson source)	average bit rate = 30 kbps per CM

the high priority traffic has the lowest access delay because it is assigned contention slots first. On the other hand, the low priority traffic has the highest access delay because it gets only the remaining contention slots and is treated with the best effort service.

The throughput performance of the proposed scheme is shown in Fig. 7. From this figure, we see that the high priority traffic throughput is more than 700 kbps during the steady state, which is close to the total high priority traffic load. Since the high priority traffic experiences fewer collisions during the contention phase, the HE successfully receives most of its bandwidth requests. The total throughput is about 2.3 Mbps in the steady state, which is about 90% of the maximum upstream bandwidth.

Fig. 8 compares the average access delay for the proposed UGPS (including bandwidth allocation algorithm) and the DOCSIS rtPS service classes. The UGS (CBR traffic), rtPS (real time VBR traffic) and Best Effort (non real-time traffic) classes are assigned priorities in a decreasing order. This means that the HE first allocates bandwidth to UGS followed by rtPS, and then by the Best Effort flow. The same priority order is also used in the proposed scheme that uses the UGPS service class instead of rtPS. In this case, the cable network is composed of 10 CMs that transmit CBR traffic and 20 CMs that transmit VBR and ABR streams. The simulation parameters are listed in Table III.

As expected, the proposed UGPS service class has much lower access delay (about 13–14 ms) than rtPS (about 25–30 ms). This is because the UGPS class uses piggyback instead of polling. Fig. 9 compares the throughput of UGPS and rtPS service classes. The proposed UGPS service class has a higher throughput than the DOCSIS rtPS class. The reason is that the dynamic bandwidth estimation scheme combined with the information about the piggyback request helps the HE to estimate the bandwidth demand of CMs more accurately. Thus, the proposed new UGPS service and dynamic bandwidth allocation scheme have a better performance than the existing DOCSIS 1.1 specification. However, the throughput for the lowest priority class, *e.g.*, the best effort service, in the proposed scheme may sometimes be

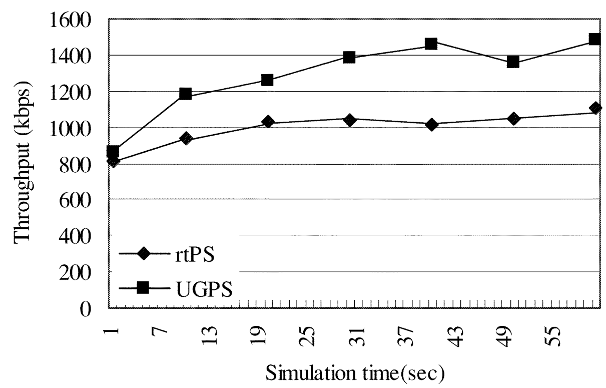

Fig. 9. The throughput performance of UGPS and rtPS service classes.

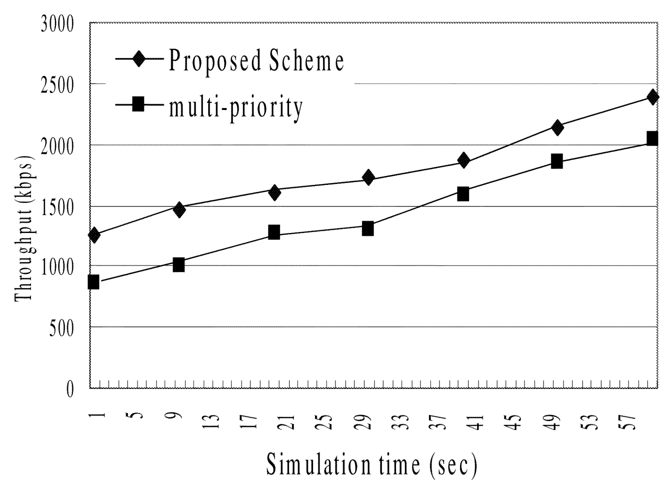

Fig. 10. The comparison of the total throughput.

lower than that in the DOCSIS system because the proposed scheme would allocate more bandwidth to real-time VBR traffic whenever required, thus reducing the transmission rate of the best effort service. Note also that the proposed scheme achieves a higher overall throughput as shown in Fig. 10, and thus higher bandwidth utilization than the multi-priority scheme specified in DOCSIS.

VI. CONCLUSION

In this research, we proposed a dynamic backoff window scheme that dynamically adjusts the backoff window sizes for different priority flows and employed a scheduling algorithm to handle the multiple priority queues. Efficiency is achieved using the priority ordering during contention access and scheduling. Simulation results showed that the proposed priority mechanism performs well and meets the requirements for different traffic types. Such a priority system improves the performance of the DOCSIS protocol. To further improve the QoS performance of the DOCSIS protocol, we also proposed a novel scheduling service class called the UGPS class and a corresponding bandwidth allocation scheme to support real-time VBR traffic. It was demonstrated by simulation results that the proposed scheme achieved a good balance in terms of a higher throughput and lower latency than the service types provided by the current DOCSIS specification.

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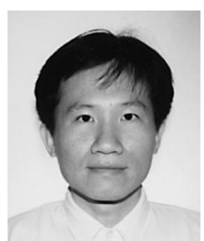

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