Providing Deterministic Quality-of-Service Guarantees on WDM Optical Networks

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***Abstract*—A major challenge in the design of future generation high-speed networks is the provision of guaranteed quality-of-service (QoS) for a wide variety of multimedia applications. In this paper we investigate the problem of providing QoS guarantees to real-time *variable length* messages (e.g., IP packets) in wavelength division multiplexing (WDM) optical networks. In particular, we propose a systematic mechanism comprised of admission control, traffic regulation, and message scheduling that provide guaranteed performance service for real-time application streams made up of variable-length messages. We formulate an analytical model based on the theory of *max-plus algebra* to evaluate the deterministic bounded message delay in a WDM network environment using our proposed QoS guarantee mechanism to determine the "schedulability conditions" of multimedia application streams. We also conduct a series of discrete-event and trace-driven simulations to verify the accuracy of the analytical model. The simulation results will demonstrate that the analytic delay bound we obtained for our WDM optical network is valid and accurate.**

***Index Terms*—Admission control, max-plus algebra, multiple-access protocols, optical networks, quality-of-service, wavelength-division-multiplexing (WDM).**

I. Introduction

WITH THE proliferation of the world wide web (WWW) in all aspects of networking, current local and wide area networks can barely cope with the huge demand for network bandwidth. As a result, there is a worldwide effort in upgrading current networks with high-bandwidth fiber-optic links that can potentially deliver terabits/s. *Wavelength division multiplexing* (WDM) is an effective technique for utilizing the large bandwidth of an optical fiber. By allowing multiple messages to be simultaneously transmitted on a number of channels, WDM has the potential to significantly improve the performance of optical networks. The nodes in such a network can transmit and receive messages on any of the available channels by using and tuning one or more tunable transmitter(s) and/or tunable receiver(s). Several topologies have been proposed for WDM networks [1], [2]. Of particular interest to us in this paper is the single-hop topology where a WDM optical network is configured as a broadcast-and-select network to which all the inputs from the various nodes are combined in a passive star coupler, and the mixed optical information is broadcast to all destinations [3].

To unleash the potential of single-hop WDM passive star networks, efficient multiple-access protocols are needed to efficiently allocate and coordinate the system resources [1]. Multiple-access protocols in a single-hop WDM passive star network environment can be divided into two classes: namely preallocation-based protocols and reservation-based protocols. Preallocation-based techniques use all channels of a fiber to transmit messages. These techniques assign transmission rights to different nodes in a static and predetermined manner. Examples of preallocation-based protocols can be found in [3]–[6]. Reservation-based techniques allocate a channel as the control channel to transmit global information regarding messages to all nodes in the network. Once such information is received, all nodes invoke the same scheduling algorithm to determine when to transmit/receive a message and on which data channel. Examples of reservation-based protocols can be found in [7]–[11]. Reservation-based techniques have a more dynamic nature and assign transmission rights based on the run-time requirements of the nodes. In this paper, we focus our attention on reservation-based techniques.

Most of the protocols proposed for reservation-based techniques are designed to handle and schedule *fixed length* packets because the underlying network is assumed to support fixed-length packet transmission because it is cost effective. On the other hand, traffic streams in the real world are often characterized as bursty. Most of the application level data units (ADU) must be segmented into a sequence of fixed size packets to be transmitted over these networks. For example, files transferred by an FTP service or video frames or audio data generated by multimedia applications [12] are too large to be encapsulated into a single fixed size packet (e.g., 53-byte ATM cell). As a result, consecutive arriving packets in a burst are strongly correlated by having the same destination node. Our intuitive idea about this observation is that all the fixed size packets of a burst should be scheduled as a whole and transmitted continuously in a WDM network rather than schedule them on a packet-by-packet basis. Another way of looking at this is that we should not segment the ADU's. Rather we should simply try to schedule them as a whole without *interleaving*. The main advantages of using a burst-based (message) transmission over WDM networks are: 1) to an application, the performance metrics of its data units (i.e., ADU's) are more relevant performance measures than ones specified by individual packets; 2) it perfectly fits the current trend of carrying IP traffic over networking infrastructures that support fixed length packets such as ATM/WDM networks; and 3) message fragmentation and reassembly are gone. Recently, few researchers have relaxed the constrain of simply using fixed length packets by allowing their network protocols to manage variable length messages transmission [7], [9], [13]. In this paper, we use this line of thought and focus on handling variable length messages on WDM networks.

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One of the important issues of high-speed networks, such as WDM optical networks, is to provide guaranteed performance service for real-time applications such as multimedia applications with their QoS requirements. The significance of guaranteeing the QoS lies in that once an application stream is admitted, the network service should guarantee the delivery of all its messages within the desired QoS constraints. There are two kinds of guaranteed performance service. One is the statistical guaranteed service with which stochastic or probabilistic bounds are provided. The other is the deterministic guaranteed service with which all messages will meet their performance requirements even in the worst case. The problem of providing guaranteed performance service to real-time applications is tied to three key issues: 1) admission control of the application streams; 2) characterization of the application streams and policing the traffic; and 3) the design of efficient scheduling algorithms to manage the messages' transmissions and multiplexing.

There are vast research results on the problem of providing guaranteed performance service to real- time applications on packet-switched networks. The interested reader is referred to [16] and [17], in which the author showed that deterministic service guarantees can be achieved using various scheduling techniques provided that the traffic entering the network is well regulated. In addition, there are a lot of papers which address the traffic characterization and the traffic models to ensure guaranteed performance service [18], [19]. However, the focus of these traffic models, the characterization of traffic, and the design of the regulators to shape traffic are mainly suitable for fixed length packets rather than variable length packets or messages. There are also lots of papers on admission control policies and scheduling algorithms for multiplexers to achieve guaranteed performance service for networks [20], [21]. However, most of the scheduling algorithms are mainly suitable for fixed length packets. To the best of our knowledge, very little research effort has been dedicated to traffic models or traffic characterizations and scheduling algorithms aimed at providing guaranteed performance service to variable length packets or messages. In particular, there is very little research aimed at providing guaranteed performance service on WDM passive star networks.

The only paper that considered in detail how to model and serve variable length messages is [22]. The authors have modeled the traffic as a marked point process which consists of two sequences of variables: the message arrival times and message lengths. A new traffic characterization, called g-regularity, to characterize a marked point process has been proposed. Based on the new traffic characterization, the authors consider the network to consist of two basic elements. One element is the traffic regulator to generate g-regular marked point processes. The other is the g-server to provide QoS for marked point processes. Any composite networks can be considered to be made up of these two elements by various methods such as concatenation. This work has provided us a theoretical system modeling method that can handle the problem of providing guaranteed deterministic performance service to applications generating variable length messages on WDM networks.

In this paper, we propose a complete mechanism including the admission control policy, the traffic regulator, and the scheduling algorithm for reservation-based multiple-access protocols in a WDM optical network environment to provide guaranteed deterministic performance service to applications' streams composed of real-time variable length messages. Our WDM networking environment has the following characteristics to differentiate them from other systems. The topology and architecture of the WDM optical network is based on the single-hop passive star coupled topology proposed in [7] and [9]. The application traffic streams are composed of aperiodic and variable length messages which are defined as variable numbers of packets with fixed number of bits for each packet. In other words, we foresee our networking infrastructure to be supporting fixed length packets (e.g., ATM/WDM, slotted WDM). This is essentially the de facto trend in high-speed networking because it is more cost effective, easier to control and manage, and can be built at a very high speed. Our contribution in this paper lies in that we have proposed a complete mechanism for reservation-based protocols in WDM optical networks in order to provide deterministic performance service to variable length aperiodic messages with time constraints. We have also analytically evaluated the delay bound for the messages' delivery in WDM networks based on the *max-plus algebra* and the framework proposed in [22]. Moreover, we have conducted a group of discrete-event and event-driven simulations to verify the delay bound we have set up for our "schedulability conditions" and to evaluate the system utilization.

The remainder of this paper is organized as follows. Section II specifies our WDM network environment and our system model. Section III presents the key components of our mechanism for providing guaranteed performance service. Section IV provides our theoretical analysis on delay bounds for variable length messages on a WDM network by max-plus algebra. Section V shows experimental evaluation results of the guaranteed performance service of the system. Finally, Section VI concludes the paper with a summary of the results and a discussion on our future work.

II. Network and Service Models

In this section, we specify the logic structure of our WDM network architecture. Then, we formulate the problem of providing guaranteed service in the specified architecture. Finally, we set up a system model which maps the logic structure of the WDM network architecture into a simplified point-to-point network structure composed of a multiplexer and transmission links so that the delay bound of the message delivery can be easily determined using max-plus algebra.

A. Network Overview

As mentioned previously, we consider message transmission in a single-hop WDM optical network, whose nodes are connected via a passive star coupler. The star coupler supports C channels and N nodes in the network. The C channels, referred to as data channels, are used for message transmission. Another channel, referred to as the control channel, is used to exchange global information among the nodes regarding the messages to be transmitted. The control channel is the basic mechanism for

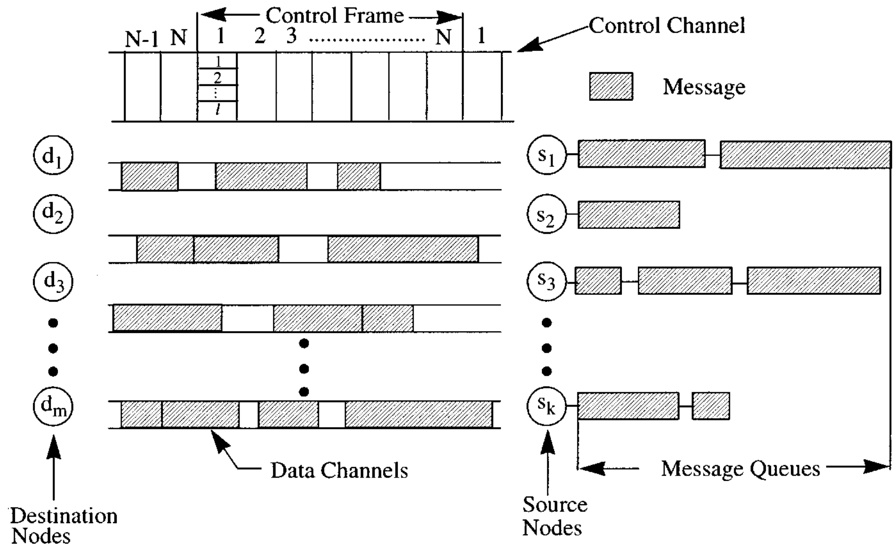

Fig. 1. Data and control channel configuration and message queues at nodes.

implementing the reservation scheme. Each node in the network has two transmitters and two receivers. One transmitter and one receiver are fixed and are tuned to the control channel. The other transmitter and receiver are tunable and can tune to any of the data channels to transmit/receive messages on those channels. This structure is similar to the optical network environment proposed in [7], [9], and [13].

The nodes are divided into two nondisjoint sets of source (transmitting) nodes s_i and destination (receiving) nodes d_j. However, any node can be a source node as well as a destination node at the same time, as there is a transmitter and a receiver at each node. A queue for the messages waiting to be transmitted is assumed to exist at each source node.

A time division multiple access (TDMA) protocol is used on the control channel to manage the transmission of control information. According to this protocol, each node can transmit a control packet during a predetermined time slot. N control packets make up one control frame on the control channel. Thus, each node has a corresponding control packet in a control frame, during which that node can access the control channel. The length of a control packet depends on the amount of control information pertained to each message, e.g., the address of the destination node message length. Fig. 1 illustrates some of the basic concepts used in our network environment.

The application streams in our WDM optical network are specified by three parameters. One parameter is the delay allowance of each message which is to indicate the time constraint of the application streams. The other parameter is the maximum length of the message in one application stream. Although the message length can be variable, it is bounded by a maximum message length. The last parameter is the traffic intensity which determines the bandwidth that an application requires.

The network service can be divided into two levels. The upper level is the flow level at which the application streams are managed and controlled by the network. The lower level is the message level at which individual messages are to be scheduled and transmitted. The flow level network service can be described as follows. When an application stream wants to connect to the network, the network evaluates its specification to decide whether it can be admitted or not. This is done to ensure that the QoS of the applications already connected to the network as well as the new coming application can be guaranteed. This network function is called admission control. Another function of the network service is to monitor the compliance of the admitted applications streams' with the streams' specifications. Once there is any specification violation, the network has to shape the traffic to make it obey some regularity principle. The goal of this shaping is to ensure that the QoS of the connected application streams can be maintained. This network function is called traffic policing.

The message level network service can be described as follows. When a source node has a message at the head of its queue to be transmitted, the source node s_i first sends a control packet during time slot i on the control channel to all other nodes. After $R+F$ time units, where R is the roundtrip propagation delay between a node and the star coupler and F is the time duration of a control frame, all the nodes in the network will have the information contained in a control frame regarding the messages to be transmitted. At this point, an identical copy of a distributed scheduling algorithm is invoked by all nodes, which assigns the messages represented in the control frame to the appropriate data channels to be transmitted at a given point in time. This scheduling algorithm has basically two functions: the resources assignment and the messages transmission ordering.

B. Problem Statement

Based on this WDM architecture and the network service, we can formulate the problem of providing deterministic bounded delay service as follows. We assume that there are N nodes and C data channels in our specified WDM network. There are also N application streams with variable length messages that request deterministic bounded delay service from the N transmission nodes. The problem we are facing now is to design an admission control policy, a traffic regulation function, as well as a message scheduling algorithm to provide transmission service

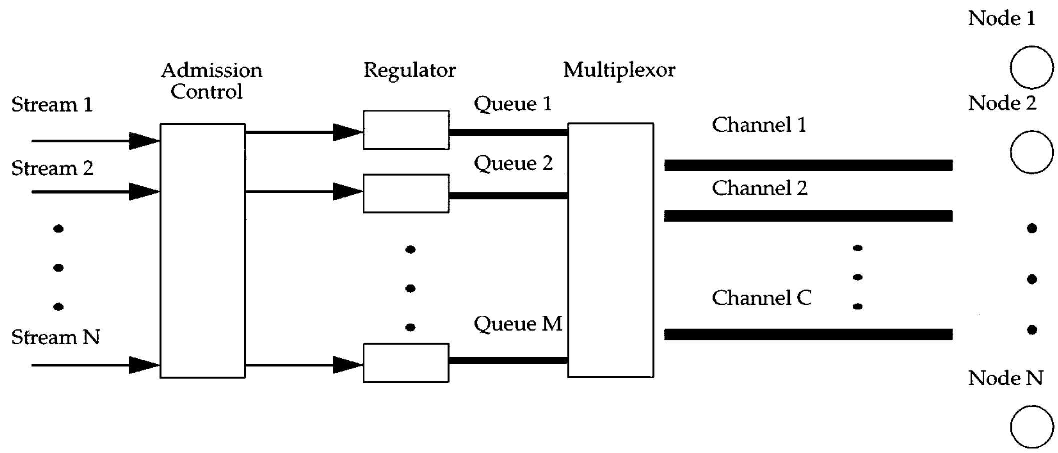

Fig. 2. System model of the specified WDM optical network.

to the application streams while ensuring that their QoS can be deterministically guaranteed.

C. System Model

Based on the above description, we can set up a system model to describe our specified WDM optical network and the network service. We map the specified WDM network architecture into a simple point-to-point connection-oriented message-switched multiplexer and C transmission links so that the problem of providing deterministic bounded delay service to the N application streams in the WDM network can be feasibly solved. The configuration of our system model is shown in Fig. 2.

We map each transmission node in our WDM network architecture to each head of the message queue. We then logically map the distributed scheduling algorithm to a centralized multiplexer on all transmission nodes. This multiplexer has the same function as the original scheduling algorithm. We also map the C transmission channels in the WDM network to the transmission links outgoing from the logical multiplexer. We specify that there are N destination nodes to receive the transmitted messages.

III. Network Service Scheme

In this section, we present service schemes for providing deterministic bounded delay service to the applications' streams in our system model. The key elements are the admission control policy, traffic characterization, and the scheduling algorithm of the multiplexer.

A. Admission Control Scheme

As mentioned previously, when an application stream wants to connect or enter into the network, the network evaluates the specification of the application stream to decide whether this application stream can be admitted or not. This is done to ensure that the QoS of the applications already connected to the network as well as the new coming application can be guaranteed. This network function is called admission control. We formulate the admission control scheme as follows. We assume that there are n application streams already connected to the network specified by $S_i = (d_i, m_i, t_i), i = 1, 2, \cdots, n$. There are m new application streams requesting guaranteed bounded delay service denoted by $S_j = (d_j, m_j, t_j), j = n+1, n+2, \cdots, n+m$. The admission control scheme employs a transmission bandwidth schedulability test to decide whether all/subset/none of the new application streams can be accepted.

Admission Control Scheme:
m new applications specified by S_j are requesting service, while n applications specified by S_i are being served;

Call resources schedulability test algorithm to consider the $n + m$ applications specified by S_p, $p = 1, 2, \cdots, n+m$;

If the delay bound of the $n + m$ applications can be ensured, accept all of the new applications,

If the delay bound of the $n+k$, $1 \leq k < m$, applications can be ensured, accept the subset of the new k applications, and reject the other subset of the new $m - k$ applications.

Otherwise reject all of the new applications

The key component of the admission control scheme is the resource schedulability test algorithm which is based on a traffic intensity-oriented test. Our traffic intensity-oriented resource schedulability test is designed based on the principle that the total rate of all the traffic admitted into the network be kept below a threshold at which the bounded delay of all application streams can be guaranteed. We formulate our problem and present our complete algorithm as follows. There is a pair (S, t), where S is a set $\{x_1, x_2, \cdots, x_i, \cdots x_n, x_{n+1}, \cdots, x_{n+m}\}$ of positive integers, and t is also an integer. In this set, x_i is the traffic intensity of application stream i, t is the maximum total traffic intensity the network can support under the condition that the bounded delays of all application streams can be met. The first n values are the traffic intensities of the currently connected

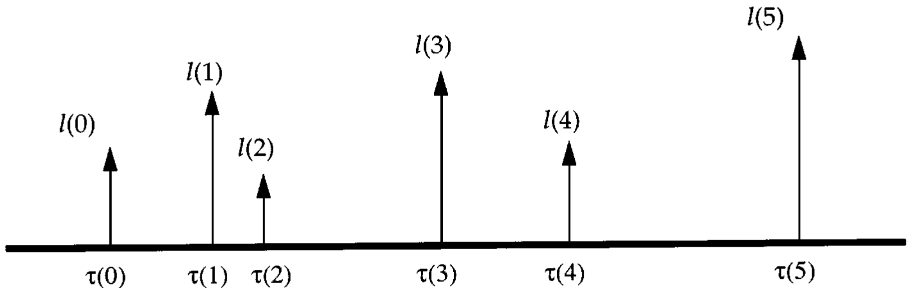

Fig. 3. Example of a marked point process.

application streams and the other m values are the traffic intensities of the new application streams. The solution to this problem is to get a subset of S whose sum is as large as possible but not larger than t. We name this algorithm the *sum-subset search* algorithm. The sum-subset search algorithm has two parts. The first one is to get a maximum value t' which is equal to or as close as possible to t. The second part is to search S to find out a subset of S whose sum is equal to t'.

```
Sum-Subset Search Algorithm:
Get t':
    L_0 ← (0);
    for (i = 1 to n + m) {L_i ← Merge-List
      (L_{i-1}, L_{i-1} + x_i)};
    Remove from L_i every element which is
      larger that t;
    Return the largest element in L_{n+m} as
      t'.
Search subset:
    S' = sort (S) in order of its value;
    for (i = 1 to n + m) {
      for (j = 0 to n + m - i) {
        substotaltraffic = substotaltraffic
          + S'(i + j)
        if substotaltraffic ≤ t'{mark
          S'(i+j); j++; if substotaltraffic =
          t' break;};
        if substotaltraffic > t' {j++;
          substotaltraffic =
          substotaltraffic − S'(i + j)}
                }
                    }
      search (S) for marked elements in S';
      Return subset of S.
```

The possible largest value of maximum total traffic is 1. This value could be calculated from the delay bound expression according to different scheduling algorithms as will be shown later.

B. Traffic Characterization

After a stream connection is established, the network must monitor the application stream's traffic with a traffic policing scheme to ensure that all traffic complies with its original specification. The traffic policing scheme either drops or delays messages which do not conform to the traffic specification in order to avoid having excessive traffic into the network. In our traffic policing scheme design, we take the policy of delaying messages which do not conform to their traffic specification. In this case there are special buffers required to temporarily store the delayed messages.

Since the traffic policing scheme is based on the traffic characterization, we introduce our traffic characterization, g-regularity, which is aimed at traffic with variable length messages. We start by introducing our traffic model since it is important that the traffic characterization conforms to a parameterized traffic model. We use a marked point process to model the application streams' traffic. A marked point process $\psi = (\tau, l)$ consists of two sequences of variables $\tau = \{\tau(n), n = 0, 1, 2, \cdots\}$ and $l = \{l(n), n = 0, 1, 2, \cdots\}$, where $\tau(n)$ and $l(n)$ are the arrival time and the message length of the $n + 1$th message, respectively. Fig. 3 shows a typical sample path of a marked point process.

Let $L(0) = 0$ and $L(n) = \sum_{m=0}^{n-1} l(m)$ be the sum of the message lengths of the first n arrivals. The sequence $L = \{L(n), n = 0, 1, 2, \cdots\}$ is an increasing integer-valued sequence with $L(0) = 0$. Based on this traffic model, a new type of traffic characterization called g-regularity can be introduced. The g-regularity can be defined as follows.

Definition 1: A marked point process $\psi = (\tau, l)$ is said to be g-regular for some g, if for all $m \le n$ then $\tau(n) - \tau(m) \ge g(L(n) - L(m))$ holds.

With $g(0) = 0$, this traffic characterization can be rewritten as

$$\tau(n) = \max_{0 \le m \le n} [\tau(m) + g(L(n) - L(m))]$$

or simply

$$\tau = \tau^*_L g.$$

Based on the definition of the g-regularity, we can construct a traffic regulator so that its output is g-regular. The following theorem[1] shows the possibility and feasibility of constructing a g-regulator.

Theorem 1: Suppose that the sequence g is superadditive. Consider a marked point process $\psi = (\tau, l)$, let $\tau^1(n) = \max_{0 \le m \le n}[\tau(m) + g(L(n) - L(m))]$ for all n.

Construct the marked point process $\psi^1 = (\tau^1, l)$.

1) ψ^1 is g-regular.

[1]Because of space limitations (and as suggested by the reviewers) we are omitting the proofs for the theorems and lemmas in this paper. The interested reader is referred to [27] for more details about the proofs.

2) For any g-regular marked point process $\psi^* = (\tau^*, l)$ with $\tau^* \geq \tau$, then $\tau^* \geq \tau^1$.
3) ψ is g-regular if and only if $\psi^1 = \psi$.

The construction of ψ^1 is called a minimal g-regulator.

Theorem 1 shows that a minimal g-regulator generates an output which is g-regular. The best construction that can be implemented is the minimal g-regulator that minimizes the departure times.

In order to show that a superposition (multiplexing) of K g-regular marked point processes is g-regular, we first introduce the concept of inverse functions and their properties as follows:

Lemma 1: Given a marked point process $\psi = (\tau, l)$. Let $l_{\max}$ be the maximum service requirement and $V(t)$ be the cumulative service requirement by time t.

1) If ψ is g-regular, then for all $s \leq t$ yields

$$V(t) - V(s) \leq g_u^{-1}(t-s) + l_{\max}$$

where $g_u^{-1} = \inf\{s \geq 0\colon g^*(s) \quad > \quad t\}$ is the upper inverse function of any function g^*, with $g^*(n) = g(n)$ for all $n = 0, 1, 2, \cdots$

2) If $V(t) - V(s) \leq f(t-s)$ for all $s \leq t$ and some nonnegative and increasing function $f(t)$, $t \geq 0$, then ψ is f_L^{-1}-regular, where $f_L^{-1}(n) = \inf\{t \geq 0\colon f(t) \geq n\}$ is the lower inverse function of f.

Lemma 1 leads to the following theorem on the superposition.

Theorem 2: Given K marked point processes, $\psi_k = (\tau_k, l_k)$, $k = 1, 2, \cdots, K$. Let $\psi = (\tau, l)$ be the superposition of these K marked point processes. If ψ_k is g_k-regular, then ψ is g-regular with

$$g(n) = \inf\left\{t\check{S} \geq 0\colon \sum_{\kappa=1}^{K} g_{u,k}^{-1}(t) + l_{k,\max} \geq n\right\}$$

where $g_{u,k}^{-1}$ is the upper inverse function of g_k and $l_{k,\max}\check{S}$ is the maximum service requirement of l_k.

Theorem 2 not only shows that the superposition of K g_k-regular marked point processes is a g-regular marked point process, but also provides some insight on the service time of the multiplexer when the K g_k-regular marked point processes are superposed to a g-regular traffic.

These theorems form the foundation of the traffic characterization of g-regularity by a g-regulator. Our traffic policing scheme is based on the traffic characterization of g-regularity, and we implement the traffic policing by using a g-regulator.

C. Scheduling Algorithm

The scheduling algorithm is the crucial part of network service schemes. It is the strategy for the multiplexer to efficiently manage individual messages from multiple application streams to be transmitted within their deterministic delay bounds. The algorithm should handle two issues of the scheduling problem. One is the message sequencing, which determines the sequence of messages' transmissions. The other is channel assignment, which determines which channel and what time slots on that channel can be used to transmit the selected message [7], [13]. We designed a scheduling algorithm in our network model to handle the above two issues. We propose an *adaptive round-robin and earliest available time scheduling* (ARR-EATS) algorithm to provide guaranteed deterministic bounded delay service, in conjunction with our admission control and traffic policing schemes.

One of the characteristics of our ARR-EATS algorithm is that it is dedicated to scheduling variable- length messages. It has two parts to handle the message sequencing problem as well as the channel assignment problem. The first part is the adaptive round-robin algorithm which determines the message transmission sequence. The basic idea is that every application stream admitted to the multiplexer can get equal opportunity to have its message transmitted. More specifically, the scheduler arranges the message transmission sequence in the following way: the multiplexer transmits the first message in the queue of the first application stream, then it takes the first message in the queue of the second application stream and so on. When the first message in the queue of the nth stream (the last queue) has been transmitted, it will return to the queue of the first application stream to schedule the second message. If there is no message in the current queue, the multiplexer will skip to the next queue. The ARR algorithm is different from the original round-robin algorithm in that the ARR algorithm switches to serve the next queue when the current message has been successfully transmitted. Since the length of each message is different, the service time for each queue is also different; while the original round-robin algorithm serves different queues with equal time slices.

The technique to assign a data channel and transmission time slots to the selected message may vary based on different WDM optical network models. Examples of such techniques that are currently receiving attention are proposed in [7], [9], and [13]. One such assignment algorithm is called *earliest available time scheduling* (EATS). It is an efficient channel assignment algorithm for selecting a channel and time slots on that channel to the transmitted messages. In our scheduling algorithm, we adopt EATS as our basic channel assignment mechanism.

The basic idea of the EATS algorithm is to assign a message to a data channel that has the earliest available time slot among all other channels. In order to keep record of the channels and receivers usage and their states, there are two tables residing on the multiplexor which are termed *receiver available time* array (RAT) and *channel available time* array (CAT). RAT records the nonavailable time of the receiver of each node from the current time in the packet slot unit. CAT records the nonavailable time of each channel from the current time. With this global information on the multiplexer, the EATS algorithm works as follows: sort the channels based on the information of CAT; choose a channel with earliest available time slot, that is the channel with the smallest CAT value; calculate the transmission time of a message based on the two tables; and update the two tables according to the newly scheduled message.

Combining this channel assignment algorithm with our ARR message transmission sequencing algorithm, we can form our new scheduling algorithm as ARR-EATS algorithm. We formally present our ARR-EATS scheduling algorithm as follows.

We assume that there are N transmission nodes and C channels in our WDM network. The index of the node is i, $i = 0, 1, 2, \cdots, N-1$. The CAT table has C items.

```
  i = −1;
START: i++;
  The new message to be transmitted from
  the queue i%N;
  If there is no message in the queue i%N,
  go to START to seek next message; Other-
  wise send a control packet on the con-
  trol channel to inform all nodes; Sort
  CAT[k] in nondecreasing order by the
  value of CAT[k] to form CAT′[h]; Use
  the channel k to transmit the message,
  where CAT′[h] = CAT[k] and h = 0; Calcu-
  late r = RAT[j] + T, t1 = max(CAT[k], T), t2
  = max(t1 + R, r); where T is the transmit-
  ters' tuning time, R is the propagation
  delay.
  Schedule the message transmission time
  at t = t2 − R;
  Update RAT[j] = t2 + m, CAT[k] = t2 − R + m,
  where m is the message length.
  Go to START to schedule next message;
```

IV. Delay Bound Analysis

The analytical evaluation of the guaranteed deterministic delay bound for our proposed network service schemes including the traffic characterization and the scheduling algorithm is based on the theory of max-plus algebra [24].

Our evaluations are done under the following assumptions.

1) The tuning times of the transmitters and receivers are constant with same value.
2) The propagation time for either the control packet or message transmission is considered.
3) A message transmitted by a node is destined to every other node with equal probability.
4) In each application stream, the traffic is specified by the three elements, (d_i, m_i, t_i).
5) In each application stream, the message length can be any value but bounded by m_i.
6) In each application stream, the average traffic intensity or the maximum intensity is t_i.
7) In each application stream, the delay allowance for each message is d_i.
8) In each application stream, the traffic is not initially characterized.

We present the theoretical analysis of the guaranteed deterministic delay bound for our network model by first introducing further analytical results on the g-regulator and g-server. Our theoretical analysis to the system model is based on the max-plus theory.

Definition 2: A server is called a g-server for an input marked point process $\psi = (\tau, l)$ if its output marked point process $\psi^1 = (\tau_1, l)$ satisfies $\tau^1(n) \leq \max_{0\leq m\leq n}[\tau(m) + g(L(n) - L(m))]$ for all n.

Obviously, a g-regulator is a g-server as it satisfies this definition, too. Now we present a theorem on the performance bounds of the g-server, which provides the foundation to achieve the delay bound of our system service scheme.

Theorem 3: Consider a g_2-server for a marked point process $\psi = (\tau, l)$. Let $\psi^1 = (\tau^1, l)$ be the output. Also let $d = \sup_{n\geq 0}[\tau^1(n) - t(n)]$ be the maximum delay at the server. Suppose that ψ is g_1-regular, then we have the following.

1) Maximum delay: $d \leq \sup_{n\geq 0}[g_2(n) - g_1(n)]$.
2) Maximum queued service requirements. The total amount of service requirements queued at the server is bounded above by $g_{u,1}^{-1}(d) + l_{\max}$, where d is the maximum delay in (1), $g_{u,1}^{-1}$ is the upper inverse function of g_1 and $l_{\max}$ is the maximum service requirement.
3) Output characterization. If $\tau^1 \geq \tau$, then ψ^1 is g_3-regular, where $g_3(0) = 0$ and

$$g_3(n) = \inf_{m\geq 0}[g_1(m+n) - g_2(m)], \qquad n > 0.$$

We have now reached the point where we can evaluate the delay bound for the general g-server. Furthermore, we present a special kind of g-server and their properties in the following lemma. This type of g-server plays a key role in evaluating the delay bound of our system service scheme.

Lemma 2: Let O_d be a sequence with $O_d(n) = d$ for all $n = 0, 1, 2, \cdots$.

1) A server guarantees maximum delay d for a marked point process $\psi = (\tau, l)$ if and only if it is an O_d-server for ψ.
2) Suppose that ψ is g-regular. Then a server is an O_d-server for ψ if and only if the server is g^*O_d- server for ψ.

   Suppose that the server is a g^*O_d-server for ψ. Since ψ is assumed to be g-regular, it follows that the maximum delay is bounded above by d. Thus, we have that the server is an O_d-server for ψ.

Since the O_d-server has constant value of service delay, it is important to evaluate the other kinds of g-servers. Most computer communication networks can be modeled as several simple g-servers such as O_d-servers concatenated together. The following theorem is on the concatenation of g-servers.

Theorem 4: A concatenation of a g_1-server for a marked point process $\psi = (\tau, l)$ and a g_2-server for the output from the g_1-server is a g-server for ψ, where $g = g_1^* g_2$.

Based on the above lemma and theorems, we can introduce the other two kinds of g-servers which are very important to the analysis of the delay bound of our system service scheme.

The first server is the $G/G/1$ queue. Let $\psi^2 = (\tau^2, l)$ with $\tau^2(n)$ being the departure time of the n+1th customer. As $\tau^1(n)$ is the time that the $n + 1$th customer is served, we have

$$\tau^2(n) = \tau^1(n) + l(n)/r \leq \tau^1(n) + d = (\tau_L^{1*} O_d)(n)$$

where $d = l_{\max}/r$ and $l_{\max}$ is the maximum service requirement of all customers. Thus, a $G/G/1$ queue with rate r can be viewed as a concatenation of a g_1-server and an O_d-server with $g_1(n) = n/r$ and $d = l_{\max}/r$. So it is a g_2-server with $g_2(n) = (n + l_{\max})/r$ according to the concatenation theorem.

Now we consider another kind of g-server, which is named a $G/G/1$ queue with *vacation*. Suppose that the server takes a vacation every time the queue is empty. When a vacation ends, it will take another vacation if the queue is still empty. Otherwise, it starts to serve the first customer in the queue. The vacation time is bounded by $v_{\max}$, a $G/G/1$ queue with vacation can be

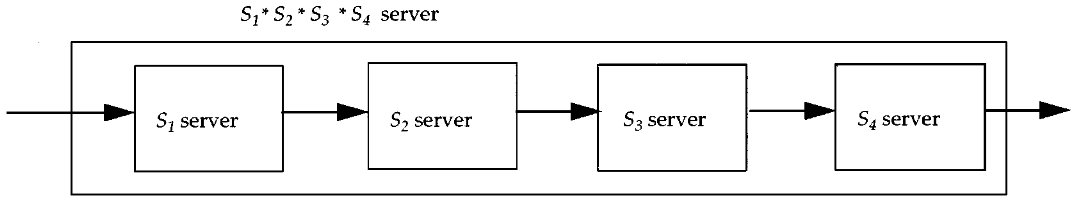

Fig. 4. Model of system service as concatenation of four servers.

viewed as a concatenation of a g_2-server and an O_d-server with $g_2(n) = (n + l_{\max})/r$ and $d = v_{\max}$. A $G/G/1$ queue with bounded vacation is a g_3-server with $g_3 = (n+l_{\max})/r+v_{\max}$.

The above analysis on the g-server which is based on the max-plus algebra has laid a theoretical foundation for us to evaluate the guaranteed deterministic delay bound of our network service schemes. Based on this, we can map each service of the multiplexer to each queue of the application stream as a $G/G/1$ server with vacation. We label this $G/G/1$ server of the service provided to each queue by the multiplexer as S_4. We assume that the service by the multiplexer to each queue of the application stream is interrupted by a vacation to that queue. The service time to each queue is bounded by the maximum message length in that application stream. The vacation time to the queue of that stream is bounded by the sum of the service time. We map the g-regulator as an O_d server with its service delay. We label the O_d server of the g-regulator as server S_1. We map the propagation of either control packet or message transmission as an O_d server, too. We label the O_d server of the propagation of control packet and message transmission as server S_2. We also map the waiting time which is the time for a message to wait for a transmission channel available as an O_d server with certain service delay. We label this O_d server of a message as server S_3. We notice that all these O_d servers and the $G/G/1$ server can be concatenated as a whole server, which is the model of our whole network service scheme. We show the whole concatenated server to represent our model of the whole network service scheme in Fig. 4.

The delay bound of the service time of server S_2 is equal to the propagation time of messages. To evaluate the delay bound of server S_3 relies on analysis of the execution of the EATS algorithm. The delay bound of the server S_4 is composed of two parts. One is the time a message spends in the queue to receive the service of the multiplexer. The other part is the time when the service to the current queue is at vacation. Hence, the whole service scheme of our network is modeled as a concatenation of a set of O_d servers and one $G/G/1$ server with vacation so that the evaluation of the bounded-delay to the whole concatenated server is simply to add the bounded delay of each server together according to the theorem of g-server concatenation. The total delay bound of our system service schemes to individual application streams should be the sum of the service time of each server as they are concatenated. We present the total delay bound in the following formula:

$$\text{delay bound} = S_1 + S_2 + S_3 + S_4$$
$$\text{delay bound}_i = P + M_i^* I_i + M^* N/C + \Sigma_i^N M_i^* I_i/C$$

where P is the propagation time for the message to propagate in the network; N is the number of application streams admitted into the network; C is the number of the channels; M is the maximum message length from all application streams; and M_i is the maximum message length from the application stream i; I_i is the maximum traffic intensity for admitted application stream i.

V. EXPERIMENTAL EVALUATION

In this section, we present the results from a set of simulation experiments to evaluate the performance of the proposed network service schemes. We conduct two groups of simulation experiments. In the first group of experiments, a generic type of real-time data traffic model is employed to conduct a discrete event simulation studying the performance of the networks with our systematic scheme. In the second group, we use real MPEG traffic to conduct a trace-driven simulation. We also evaluate the parameters of the real MPEG traffic to form two modeled MPEG traffic and use them in a discrete event simulation. We present the experiment designs as well as the experiment results of these two groups of simulations in the following subsections.

A. Simulation with Real-Time Data Traffic

In this experiment, the effect on the WDM performance by varying the traffic intensities from all the admitted application streams is studied. We also compare the delay bound from theoretical analysis with the maximum message delay received through the simulation experiments to validate each other.

1) Experiment Design: In this experiment, we assume that there are three application streams admitted to get transmission service. And there are two data channels to be used for transmission. We describe the real-time application traffic based on the marked point process. The parameters used in our experiments are as follows. Message length is a random variable following an exponential distribution with mean value of 50 time units but bounded by maximum message length of 100 time units for each application stream. A Poisson message arrival process for each of three application streams is considered whose mean interarrival time can be determined by the maximum message length in each application stream divided by the average traffic intensity of the corresponding stream. The average intensity changes from 0.1 to 0.325 for each application stream and from 0.3 to 0.975 for the total traffic in the network. Tuning latency is considered as 10 time units and the propagation delay is set to 100 time units in the experiments. Destination nodes for messages are chosen according to a uniform probability distribution. The

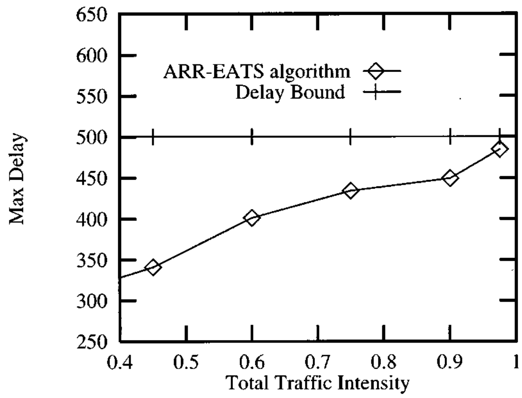

Fig. 5. Max delay versus traffic intensity.

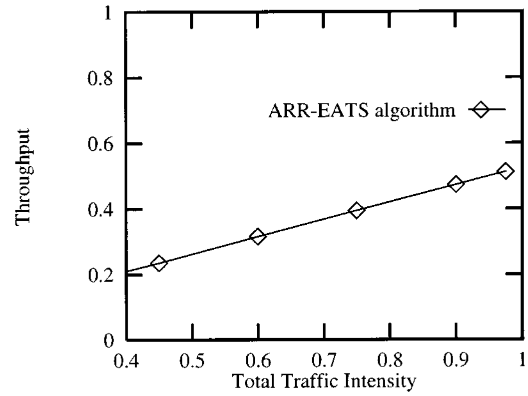

Fig. 6. Network throughput versus traffic intensity.

message delay allowance is set to be the same as the delay bound. The behavior of the network service schemes is observed over a simulation period of 1 000 000 time units. The metrics of the performance used in the experiments are the *maximum message delay* for comparison to the delay bound which is obtained from the formula in the previous section. This metric is rather a statistical metric than an experiental metric which means that once a maximum value has appeared it will be recorded. The *system throughput* is used as a general metric to describe the efficiency of the network.

2) Experiment Results: We present the experimental results in two figures. One of the figures shows the maximum message delays the messages experienced during a simulation period of 1 000 000 time units and the delay bound which is evaluated by the formula we set up in the previous section. The other figure shows the changes of the total system throughput when the total traffic intensity is increased during the simulation.

In Fig. 5, we present the maximum message delay and the delay bound of the ARR-EATS algorithm. It shows the relationship between the maximum message delay and the total traffic intensity. It also displays the relationship between the delay bound and the total traffic intensity. In this figure, we can see that the maximum message delay increases when the total traffic load increases. The reason for this is that when the traffic is heavy, messages come into the network at a higher rate, hence, the waiting time for a message gets longer. This figure also reveals that when the total traffic intensity is under 1, the maximum message delay can always be kept under the delay bound. When the total traffic intensity approaches 1, the maximum message delay is close to the delay bound. This fact can verify that the formula to evaluate the delay bound we received in the last section is reasonable and correct. If the delay allowance of any of these three application streams is set to be the same as the delay bound, the message tardy rate of those application streams should always be 0 when the total traffic intensity is under 1. If the delay allowance of any of these application streams is less than the delay bound, the message tardy rate of those streams can be changed to be larger than 0, even if the total traffic intensity is less than 1.

Fig. 6 shows the relationship between the total system throughput and the total traffic intensity from all the application streams. In this figure, we can see that the total network throughput increases when the total traffic load increases. The reason for this lies in that when the input traffic load increases, the output traffic intensity of the network will also increase if the network is not blocked. The total network throughput is actually a metric to measure the output traffic intensity. This figure also reveals that the throughout of the network is not high. It is just a bit more than 50% when the input traffic load is near 1. And this reflects the fact that the utilization of the network is degraded when guaranteed deterministic bounded delay is required.

B. Simulation with MPEG Video Traffic

In this group of experiments, we employ real MPEG video traffic, which is available from [25], to investigate the effectiveness of our proposed admission control algorithm, and to evaluate the performance of our g-regularity traffic characterization scheme and ARR-EATS scheduling algorithm.

1) Experiment Design: In this group of experiments, we still have two data channels for message transmission. But we do not limit the number of admitted application streams to be served because herein we are going to examine the effectiveness of our admission control algorithm. The number of application streams admitted into the network is determined dynamically by our admission control algorithm. At [25], there are around 20 files containing frame size traces from MPEG encoded video sequences. We pick 13 different traces from the above collection. We assume that these 13 application streams are requesting guaranteed deterministic bounded-delay transmission service from our WDM optical network.

A frame size trace from MPEG encoded video sequence is a series of data, each item of which is the size of a frame in bits. Each of the selected frame size traces has the following properties: 1) there are 39 996 frames in each trace file; 2) the frames are arranged in a deterministic periodic sequence in a pattern of IBBPBBPBBPBB; 3) we assume that in every second, there are two group of pictures (GOP). It means that the interarrival time between frames is constant and it is 1/24 second. Each of the trace file contains 27.8 minutes of real-time full-motion video. The contents of all the selected 13 frame size traces include several different video types such as TV news and talks, episodes from comedies and cartoons, cable TV of soccer games and races, and movies.

We assume that the bandwidth of each data channel is 3.0 megabits. We map the time unit in our simulation system as

TABLE I
PARAMETERS OF FRAME SIZE TRACES

Trace No.	1	2	3	4	5	6	7	8	9	10	11	12	13
Max Size	26.5	44.3	39.9	35.6	46.9	41.6	44.7	80.1	49.1	57.6	76.4	83.8	63.3
Ave. Load	0.09	0.14	0.11	0.12	0.19	0.07	0.06	0.15	0.18	0.11	0.14	0.16	0.12
Max. Load	0.27	0.45	0.42	0.31	0.57	0.30	0.31	0.56	0.72	.0.45	0.58	0.97	0.74

TABLE II
ADMITTED TRAFFIC STREAMS UNDER DIFFERENT THROUGHPUT

Throughput	0.6	0.7	0.85	0.91	1.0
Admitted Streams	2,5,8,13	5,7,8,9,13	2,3,5,8,11,13	5,8,9,10,12,13	4,5,8,9,11,12,13
Number of Admitted	4	5	6	6	7

1 millisecond in real world. Hereby, the size of a frame is converted into the time period, in which a frame is transmitted through the WDM optical data channels. In this way, a frame size trace from an MPEG encoded video sequence can be mapped as a real-time application stream composed of variable-length messages.

In the simulations, we consider the tuning time of the transceivers to be 10 unit times and the propagation delay is 50 unit times. We study the relationship between the maximum message delay and the network throughput. The network throughput changes from 0.6 to 1.0 in the experiments. The number of admitted application streams varies as the network throughput changes. Destination nodes for application streams are chosen according to a uniform probability distribution. The message delay allowance is set to be same as the delay bound. The behavior of the network service schemes is observed over a simulation period of 1 666 500 time units, which is the same as the period of the available frame size trace has been transmitted. The metrics of the performance used in the experiments are the *maximum message delay*, which is the same as that used in the previous group of simulation experiments. The *system throughput* is also used as a general metric to describe the efficiency of the network.

In this simulation, the admission control scheme analyzes all of the application streams. The scheme gets the maximum frame length, the average traffic intensity, and the maximum traffic intensity as parameters of each application stream. Then the admission control algorithm, *sum-subset search* algorithm, is invoked to determine which application streams to admit into the network for transmission service from 13 requesting streams based on the average traffic intensity of each of stream. The traffic of each admitted stream will then be shaped by the characterization scheme, which is based on the principle of g-regularity. The shaped traffic of each admitted stream will get the transmission service under our variable-length message transmission scheduling algorithm—the ARR-EATS algorithm.

2) Experiment Results: First, we present a table to show the effectiveness of our admission control algorithm, which admits various MPEG video streams among those requesting transmission service as the throughput changes.

a) Effectiveness of admission control algorithm: We have 13 frame size traces from the MPEG encoded video sequences as application streams requesting guaranteed deterministic bounded-delay service. After being mapped to the real-time application streams composed of variable-length messages, they have their parameters in Table I:

Our admission control algorithm will determine which of the MPEG encoded video sequences are admitted into the network among the above 13 requesting sequences according to the parameter of average intensity of each of the sequences under different normalized network throughput. Table II shows the result of executing our admission control scheme.

As an example, Table II shows that when the network throughput is 0.6, the admitted MPEG traffic has numbers 2, 5, 8, 13, which are the parameters shown in Table I. Under different network throughput, the admitted MPEG traffic as well as the total number are different.

b) Network performance by trace-driven simulation: In Fig. 7, we present the maximum message delay and the delay bound of the ARR-EATS algorithm when trace-driven simulation is conducted. It shows the relationship between the maximum message delay and the network throughput. In this figure, the delay bound is calculated according to the formula proposed in Section IV. The value of the delay bound is increasing because the number of the admitted MPEG traffic increases when the network throughput gets larger. The maximum message delay, which is the largest delay any message has ever experienced in the network, increases when the total traffic load as well as network throughput get to 1. However, even when the throughput is 1, the maximum message delay has not exceeded the calculated delay bound. This shows that our analysis on the delay bound of message transmission is correct and valid. It can also be used to evaluate whether a real-time application stream can be transmitted within its deadline or not under certain network throughput if the parameters including time laxity (delay allowance) of the stream can be provided to the network scheme. Suppose one application stream has set its delay allowance to be 400 time units, which means any message in this stream cannot experience a delay exceeding 400 time units. From Fig. 7, we can say that if the total network traffic is less than 0.8, our net-

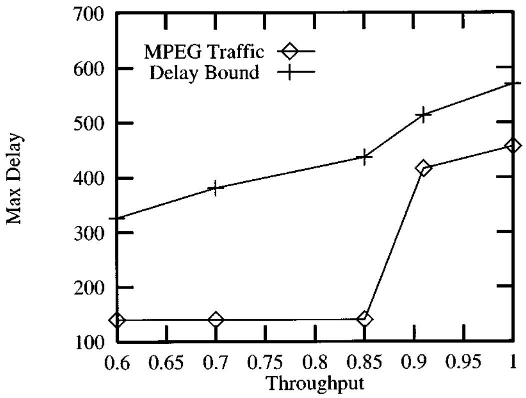

Fig. 7. Max delay versus network throughput (real traffic).

work scheme can ensure this deterministic bounded delay when the messages in this stream are scheduled and transmitted. Otherwise, it cannot be guaranteed. Another important point shown by this figure is that the delay bound guaranteed by our network scheme to the MPEG encoded video traffic is around 500 time units when the throughput is almost equal to 1. Let us convert this delay bound into the real world. We have set that one simulation time unit is equal to 1 millisecond before we proceeded with the trace-driven simulation under the condition that the bandwidth of the data channel is 3.0 megabits. The guaranteed delay bound of 500 time units means that the maximum delay imposed on the MPEG encoded video streams by the WDM optical networks, with bandwidth of each data channel being 3.0 megabits under the control of our systematic scheme, will not exceed the time of transmission of one GOP or one picture. This implies that at the destination nodes, limited capacity of buffer can be allocated for temporal buffering of the information of only one GOP, while the quality of the video cannot be impaired.

Having presented the results of simulations regarding the transmission of MPEG encoded video traffic, we have the following points to summarize.

1) The admission control algorithm has been verified to be an effective algorithm in limiting the number of application streams served so that deterministic real-time service can be guaranteed.
2) Our proposed systematic scheme has been tested to have the ability to provide guaranteed deterministic bounded-delay service in the specified WDM optical networks to whatever real-time application streams composed of variable-length messages.
3) Our developed formula to evaluate the delay bound of the message transmission delay has been validated by either theory of max-plus algebra or extensive simulation experiments.

VI. CONCLUSION

In this paper, we investigated the problem of providing guaranteed deterministic bounded-delay service to application streams composed of variable-length messages with time constraints in a single-hop WDM optical network. We proposed a set of network service schemes which ensure deterministic QoS guarantees to real-time variable-length messages. These schemes include admission control, traffic characterization, and scheduling algorithms. Our main contribution in this paper lies in that: 1) we have proposed a new admission control policy to decide on which application streams (*made up of variable length messages*) can be admitted when more than one new application stream requests admission into the network; 2) we have proposed a new scheduling algorithm called ARR-EATS which is dedicated to scheduling variable-length messages and can be used in the specified WDM optical network; and 3) we formulated a mathematical evaluation model on the delay bound in our specified WDM optical network based on the max-plus algebra and the theory of g-regularity. We also evaluated the performance of the proposed techniques in a number of experiments and compared their results to those from the mathematical evaluation.

We plan to extend this framework of providing QoS guarantees for variable length packets to wavelength routed networks (multihop WDM networks). The issue of admission control in conjunction with wavelength assignment and routing would make the problem very challenging and of considerable interest in QoS-oriented wide area networks.

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