# Efficient Protocols for Multimedia Streams on WDMA Networks

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***Abstract*—This paper introduces a new approach to integrate different types of medium access control (MAC) protocols into a single wavelength-division-multiplexing (WDM) network system. The WDM network is based on a passive star coupler, and the purpose of integrating different MAC protocols is to efficiently accommodate various types of multimedia traffic streams with different characteristics and quality of service demands. Our integrated MAC protocol is termed *multimedia wavelength-division multiple-access* (M-WDMA). Three types of multimedia traffic streams are considered in this paper: constant-bit-rate traffic and two classes of variable-bit-rate traffic. Accordingly, three tunable transmitters and one fixed home channel receiver are used in the design of each WDM node. The transmitters transmit the three types of multimedia traffic streams in a *pipeline* fashion so as to overcome the tuning time overhead and to support parallel transmissions of traffic streams that emerge simultaneously. We further incorporate a *dynamic bandwidth allocation* scheme that dynamically adjusts the portions of bandwidth occupied by the three types of traffic streams according to their demands. Consequently the M-WDMA protocol achieves high utilization and efficiently adapts to the demands of the multimedia streams so as to guarantee their QoS. The performance of the M-WDMA is evaluated through a simple analytical model and extensive discrete-event simulations. It is shown that the M-WDMA can satisfy the QoS requirements of various mixes of multimedia traffic streams even under very stringent requirements. Moreover, we show that the M-WDMA outperforms conventional MAC protocols for WDM networks. As a result, we expect M-WDMA to be a good multimedia MAC candidate protocol for future-generation WDM networks.**

***Index Terms*—Analytical modeling, MAC protocols, multimedia streams, performance evaluation, WDM networks.**

## I. Introduction

It is estimated that a single optical fiber has a capacity of at least 30 THz in its low-loss region (1.2–1.6 $\mu$m). This huge bandwidth can be used for data transmission well beyond the terabits per second (Tb/s). At the present time, however, only a fraction of this huge capacity can be used due to well known *optoelectronic* bottleneck problems. These problems can be reduced by dividing the total bandwidth of the fiber into a number of noninterfering channels. These channels are of relatively lower bandwidths but with the advantage that they operate at the peak speed of the electronic processing devices. This technique is known as *wavelength-division multiplexing* (WDM). In a WDM architecture, a transmitter and a receiver have to tie onto the same wavelength (i.e., channel) so that transmission can be carried out successfully. Due to the limitation of the number of transceivers at each node, the complete connectivity of a WDM network requires the use of *tunable transceivers*, which can tune their working wavelength across all channels in the network. Unfortunately, one has to make a compromise between the tuning time, which can range from tens of nanoseconds to hundreds of milliseconds, and the tuning range, which can be from a few channels to hundreds of channels. Using the state-of-the-art technology of optical fibers, fast tuning time implies small tuning range, and vice versa. From a networking perspective, it was shown that increasing the number of channels can be quite beneficial. As a result, dealing with the overhead that can be associated with the tuning time becomes a crucial design problem in the WDM networks [5].

Several topologies have been proposed for the WDM networks [17], [18], a popular one being the single-hop p*assive star-coupled* topology. Each transmitter in this single-hop network broadcasts its data to all other nodes through a *passive star coupler*. The receivers may then filter out a coming packet at a certain wavelength for their proper reception. This single-hop network is also referred to as a *broadcast-and-select* network. However, this WDM network comes with various challenges, the most important challenge being the design of the medium access control (MAC) protocols for these networks. One reason for this challenge is that all higher layer services (e.g., such as those shown in Fig. 1) are built on the fundamental packet transfer service which is provided by the MAC sublayer, and it is the MAC protocol that determines the characteristics of this fundamental service. Hence, improvements to MAC services result in improved system performance, while the provision of new MAC services means that new applications and services can be developed [10], [11].

A plethora of MAC protocols have been proposed for star-coupled WDM networks (for more details, see [17]). However, most of these MAC protocols are not suited for an integrated services environment because they have been designed with just one *generic* traffic type in mind. As a result, they perform quite well for the traffic streams they have been designed for but poorly for other traffic streams with different characteristics. The purpose of this paper is to combine the advantages of

Manuscript received May 7, 2003. A preliminary version of this paper has won the Best Paper Award among 150 accepted papers at the 12th International Conference on Information Networking, Tokyo, Japan, 1998.

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Digital Object Identifier 10.1109/JLT.2003.816815

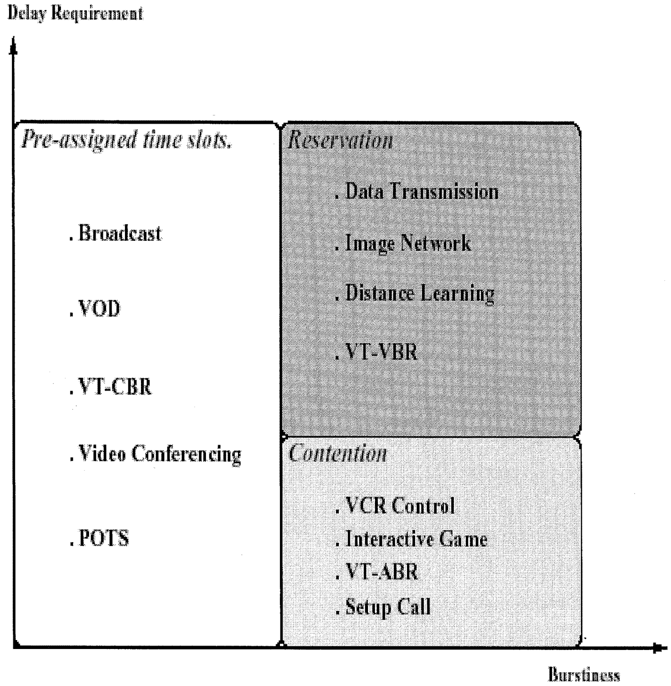



Fig. 1. Multiple traffic stream features and their appropriate MAC protocols.

these various MAC protocols within a single framework so as to be able to efficiently serve a wide range of traffic streams typical in multimedia applications. In particular, we propose a novel WDM MAC protocol that can integrate different types of MAC protocols into a single physical WDM network to efficiently support multimedia traffic streams. This MAC protocol is termed *multimedia wavelength-division mutliple-access* (M-WDMA).

The motivation behind integrating different MAC protocols into a single WDM network comes from the fact that different types of traffic streams have different characteristics. As a result, the traffic streams have different transmission and quality of service (QoS) requirements. We can classify multimedia traffic streams as a function of their data burstiness and delay requirements, as shown in Fig. 1. For example, video/audio streams and plain old telephone service (POTS) have small data burstiness but require almost constant transmission delay and almost fixed bandwidth in order to guarantee their QoS. On the other hand, applications such as image networking and distance learning are less stringent in terms of their delay requirements, but their traffic streams are very bursty. Finally, there are other applications that require a very low delay while their traffic streams are bursty. Examples of this type of applications include control messages for video-on-demand systems or interactive games and network control and management.

These different traffic streams are better served by different MAC protocols. Video/audio data streams and other constant-bit-rate (CBR) traffic streams benefit best from prescheduled MAC protocols since they can guarantee that each node has a cyclic and fixed available bandwidth. The best MAC protocols for this purpose would be a simple round-robin time-division-multiplexing access (TDMA) scheme. In a WDM network, a TDMA MAC protocol is applied to each of the channels. The scheduling principle can be represented in a scheduling table that is arranged in such a way that every channel is cyclically allocated to every node in a prefixed order, and every channel is used by a pair of source and destination nodes at a time. This protocol is usually referred to as *interleaved time-division multiple-access* (ITDMA) protocols [3], [4], or WDM/TDM protocols [16]. A typical node configuration for the ITDMA protocols uses a fixed receiver and a tunable transmitter (FR-TT) or a tunable receiver and a fixed transmitter (TR-FT). One drawback of this class of protocols is that if the input load is nonuniform or bursty, the TDMA protocol performs rather poorly.

On the other hand, reservation-based MAC protocols are very well suited for applications where the traffic streams are bursty or the traffic load of the nodes is unbalanced, since reservation-based MAC protocols schedule the transmission according to a particular transmission request. Before a transmission takes place, a node has to send a transmission request to the destination node or to a central controller. Then, the transmission can only start after the source node gets a positive acknowledgment (e.g., grant). Since a lot of detailed information about the transmission can be obtained before the transmission takes place, the transmission schedule can be very efficient and precise. As a result, these MAC protocols can be easily designed to consider tuning time, propagation time, packet processing time, deadline, transmission cost, etc.

The disadvantages of the reservation-based MAC protocols are that the reservation mechanism requires extra bandwidth and time to exchange the request and the acknowledgment information. In addition, the scheduling methods can be relatively complicated, hence, requiring more processing time. These drawbacks cause considerable additional delay and delay variation for packet transmission. Therefore, this type of MAC protocols usually does not perform well for CBR traffic streams or for applications requiring very small delays (for more details about reservation-based protocols, see [13], [16], and [25]).

Finally, random access (contention) MAC protocols usually do not consider the status of the channels or the destination nodes. Once a transmission request emerges, the node starts the transmission almost immediately. This, however, leads to potential collisions between packets. That is, packets being transmitted in the same time slot will collide, and then they have to be retransmitted again. On the other hand, if the transmission succeeds, the packet transmission delay can be extremely low. Thus, random access MAC protocols have the potential of meeting the delay requirements of very urgent messages. Some examples of these urgent messages (e.g., call setup) are listed in Fig. 1. These applications may not generate a lot of traffic data when compared with the other applications. However, once a certain traffic (message) is generated, they require a very low delay. A typical random access MAC protocol that has been proposed in WDM networks is *Interleaved Slotted ALOHA* (ISA) [4]. These protocols perform very well, especially with respect to packet delays when the traffic load is low. However, if the traffic load gets high, then packet collisions start to occur more frequently. As a result, the performance can be quite poor.

As can be seen, none of these MAC protocols serves all types of traffic well. However, each one of them is ideal for certain

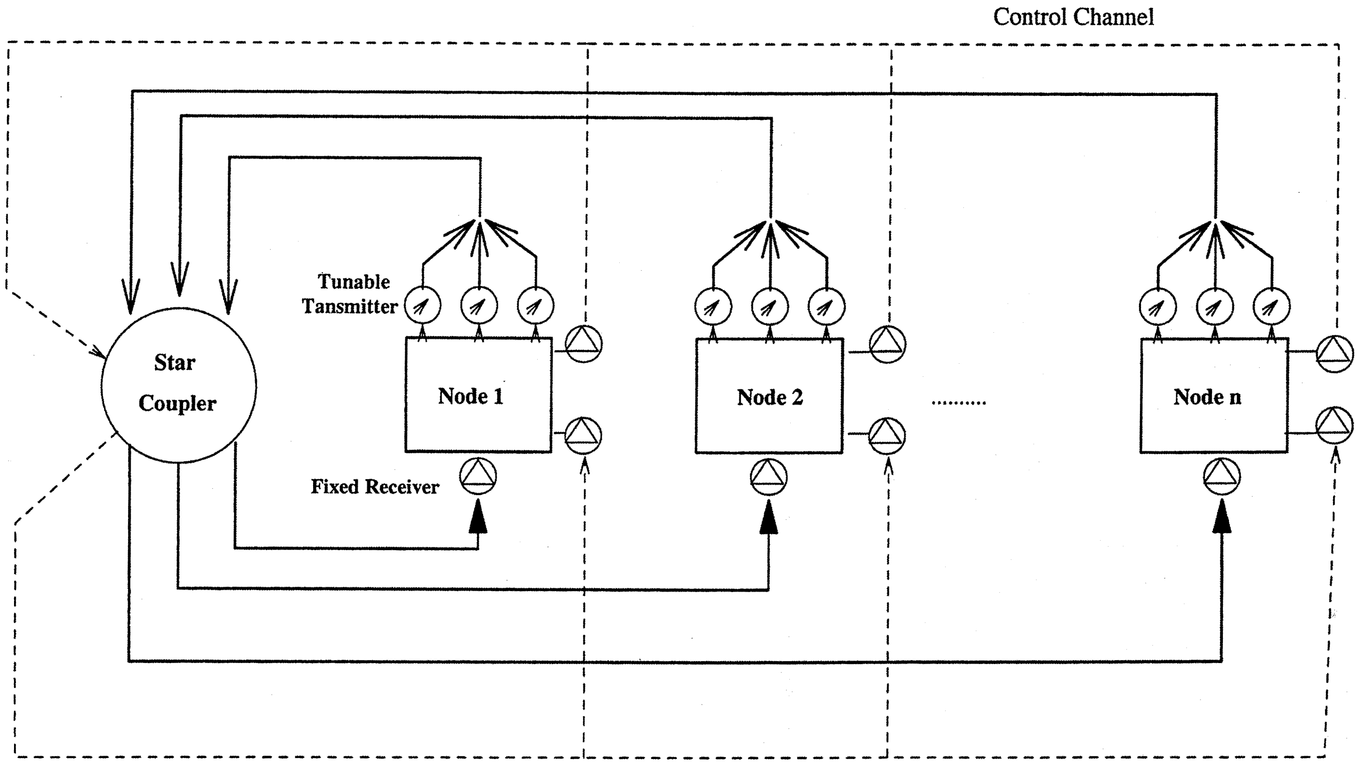



Fig. 2. M-WDMA network architecture.

types of traffic streams. This observation leads us to propose an efficient scheme of integrating all these MAC protocols into a single MAC protocol for WDM networks. We denote this integrated WDM MAC protocol as M-WDMA. In an M-WDMA MAC protocol, an ITDMA-like MAC protocol, a reservation-based (using token passing) MAC protocol, and an ISA protocol are integrated into a single physical WDM network in an efficient way so that different types of traffic streams can use the most appropriate protocol. In our M-WDMA implementation, we consider the following as pects of the environment:

1) three types of traffic streams: a CBR, a variable bit rate (VBR) with large burstiness (VBR1), and a VBR with longer interarrival times (VBR2);
2) the importance of the deadline associated with each traffic stream;
3) a nonzero tuning time of the transmitters/receivers.

In addition, we design a dynamic bandwidth allocation strategy to further improve the utilization of our M-WDMA, which we call M-WDMA+ to distinguish it from the case where the channel bandwidth is allocated in a static fashion.

This paper is organized as follows. Section II introduces our M-WDMA architecture. Section III details the operation of our M-WDMA MAC protocol. In Section IV, we present a simple mathematical model for M-WDMA architecture in order to assess its performance. Section V presents the performance evaluation of M-WDMA networks using extensive computer simulations. Section VI concludes this paper.

## II. The M-WDMA Architecture

Consider an M-WDMA network with $N$ nodes which are connected by a star coupler and having $C$ channels, as shown in Fig. 2. Each node has a fixed channel, a *home channel*, with wavelength $\lambda_i (i = 1, 2, \ldots, C)$. The home channels are intended for the destination nodes to receive packets. In case the number of available channels $C < N$, several nodes ($\lceil N/C \rceil$) may share one single home channel. The destination nodes can then accept or discard the packets by checking the addresses associated with these packets.

The M-WDMA MAC protocol has three integrated *MAC subprotocols*, namely, a TDMA protocol, a reservation-based protocol, and a random-access protocol to better serve all types of traffic streams. In our proposed M-WDMA protocol, each node has three tunable transmitters. The three tunable transmitters are used to serve three different classes of traffic streams according to the corresponding *subprotocols*. The transmitters are named TDM transmitters (for time-division multiplexing, meaning that these transmitters are driven by a TDM-like protocol), RSV transmitters (for reservation-based protocol), and CNT transmitters (for contention-based protocol).

There are two main reasons why we proposed to use three transmitters and how we specified their operation for our M-WDMA protocol: to reduce penalties of the transmitter tuning time and to combine mutliple subprotocols within M-WDMA. Each one is discussed in detail in the following subsections.

### A. Reduce Penalties of the Transmitter Tuning Time

An important parameter that affects the performance of WDM networks is the tuning delay of transmitters. The tuning delay is the amount of time required for a transmitter to tune from one wavelength to another. If we consider fixed-size packets, then we express the tuning delay $\Gamma$ in units of packet duration. The value of $\Gamma$ depends on the transmission rate $B$ (in bits per second), the packet size $L$ (in bits), and the actual tuning delay $D$ (in seconds) through the relationship

$$\Gamma = \frac{DB}{L}. \tag{1}$$

Advances in tunable optical devices decreases $D$, while the trend toward higher rates and smaller packet sizes (e.g., ATM cells) increases $B/L$. It is difficult to give an exact value of $\Gamma$ for future networks due to this "push–pull effect of technology." However, if we assume that the proportional decrease of $D$ is proportional to the increase of $B/L$, then the range of $\Gamma$ in future networks will be more or less the same as for current networks. As a result, it is safe to predict that the tuning delay of transmitters will also be a problem for future WDM networks. Let us give some example values for $D$, $B$, and $L$. The values that we use hereafter are values frequently used by many researchers in the area [5], [24] and have been used throughout this paper as well: $D = 10$ $\mu$s, $B = 1$ Gb/s, and $L = 424$ b (an ATM cell). In this case

$$\Gamma = \frac{DB}{L} = \frac{10^{-5} \times 10^{9}}{424} \approx 20.$$

Obviously, the value of $\Gamma = 20$ is quite large and can lead to a low throughput and large packet delays in WDM networks if nothing is done to reduce its effect. This same value $\Gamma = 20$ has been used by other researchers when designing their WDM protocols [5], [24]. Most of the existing work to reduce or eliminate the tuning penalty considers a small tuning latency, which is less or equal to 1 (i.e., $\Gamma < 1$) [2], [7], [12], [19]. These solutions do not work well in an environment with large latencies. Some protocols consider arbitrary (small and large) tuning latency [5], [20]. However, they only provide adequate performance for small tuning latencies but not for large tuning latencies, especially when the ratio of the number of stations over the number of channels is not too large. This is not to diminish the novelty and contributions of these protocols as it is impossible to come up with scheduling algorithms that work perfectly well under all conditions (i.e., under any ratio of the number of stations over the number of channels or under any ratio of the tuning time over the packet transmission time).

In order to completely eliminate the tuning penalties, some researchers proposed the use of multiple tunable transmitters at each node [15], [23], [24]. By appropriate scheduling of the tunable transmitters, data packets can be continuously transmitted on all wavelengths. Namely, while one transmitter is transmitting, the others are being tuned to the appropriate channels. This type of transmission is termed *pipelining* [24]. The number of transmitters per node that would eliminate the tuning penalties depends on the ratio of the number of stations over the number of channels and on the ratio of the tuning time over the packet transmission time. As a result, under certain network conditions, to eliminate all the tuning penalties, we are required to use a large number of transmitters per node. In this paper, our approach for reducing or eliminating the tuning penalty follows, in part, the last approach. That is, we proposed to use multiple transmitters at each node. However, we wanted to limit the number of transmitters per node while eliminating the tuning penalty.

The key question now is what is an appropriate number of transmitters per node that would eliminate the tuning penalty. The authors of [24] have eloquently analyzed the maximum number of transmitters that need to be used per node in order to eliminate the tuning penalty as a function of the ratio of the number of nodes to the number of wavelengths and also the ratio of the tuning time over the packet transmission time.

### *B. Reduce Penalties of the Transmitter Tuning Time*

In our M-WDMA access protocol, we are using three subprotocols (preallocation protocol, reservation protocol, and contention protocol) to access the same optical medium. Each of these MAC protocols accesses the optical medium differently. As a result, they have different threads of control. Our intuition behind this is to allocate one tunable transmitter for each subprotocol in order to simplify the hardware design of the whole interface board (between the WDM network and the terminating equipment) and, in turn, simplify the design of the whole WDM network. As a result, we proposed to use three tunable transmitters per node. Since we are using a constant three transmitters per node independent of the number of nodes, we are not faced with a scalability problem.

Hence, given that we use three tunable transmitters per node (i.e., one transmitter for each subprotocol) and given the operating parameters that we use for $D, B$, and $L$, then we have to devise a proper strategy to eliminate the tuning penalties of the transmitters. Our basic strategy to achieve that is to use a frame that consists of multiple number slots. Then, we divide each frame into three segments where each segment is allocated to one of the three subprotocols used in our M-WDMA protocol. In particular, we perform pipelining at the frame level rather than from slot to slot. That is, transmitters perform tuning between frames instead of between slots. With this strategy, channel performance is not affected by the relatively long tuning delays. In fact, the tuning time determines the minimum size of the frame that would allow us to eliminate the tuning penalty (as is shown in Section III-A). In our paper, as we showed previously and as has been used by various other papers [5], [24], we assumed that

$$\begin{aligned} D &= 10\ \mu\text{s} \\ B &= 1\ \text{Gb/s} \\ L &= 424\ \text{b}. \end{aligned}$$

Then, as was calculated previously the value of the transmitter tuning time, $\Gamma = 20$ transmission slots. In order for us to eliminate this 20-slot tuning penalty, using the M-WDMA protocol, under any value of the ratio of the number of nodes to the number of channels, the frame size has to be equal to 30 slots, which is the value that we used in our paper. In this case, while a transmitter is transmitting using its allocated segment, the other two transmitters are tuning to the appropriate wavelength in time to be able to transmit in their allocated segments.

However, we must emphasize three points in order to put the paper in a better perspective.

1) The main contribution of this paper is the integration of different MAC protocols into a single WDM network and **NOT** on dealing with the tuning delays. The problem of dealing with tuning delays has been investigated thoroughly by various researchers [5], [24]. The purpose of the integration of different MAC protocols into a single WDM network is to efficiently accommodate various types of traffic streams with different characteristics and QoS demands. This topic, even though extremely important, has received little attention from the research community as far as the design of WDM networks is concerned.
2) While we propose to use three tunable transmitters per node under the intuition that it might lead to the simplification (and hopefully lower cost) of the whole interface board (between the WDM network and the terminating equipment), it is difficult for us to quantify that. The main reason for that is the fact that fast tunable transmitters such as the ones envisioned in this paper are purely experimental at this stage. As a result, we cannot accurately estimate their costs, and it would be even more difficult to approximate the portion of costs of an interface card that can be attributed to the tunable transmitters.
3) It is very possible to design our M-WDMA protocol using fewer/more than three transmitters per node depending on the given set of parameters (e.g., $\Gamma$, $D$, $L$, or $B$). However, doing so will put the paper out of focus. The interested reader can refer to [24] for such research.

In addition to the transceiver configuration, there are many other issues that need to be considered in the M-WDMA architecture design. Because the M-WDMA is a multiple-protocol scheme, using one single queue to store all types of packets causes a severe *head-of-line blocking* problem. To eliminate the problem, we have adopted three types of queues for three differenttypes of traffic. Each type of queue has $C$ queues at each node. As a result, the total number of queues at each node is $3C$. To reduce the cost of these multiple queues, they can be easily designed using *random-access memories* (for more details, the reader is referred to [1]). In addition to the $C$ channels, another channel $\lambda_0$ is required as a control channel in which networkwide control information can be exchanged. Correspondingly, an extra pair of transmitter-and-receiver at each node is constantly tuned to this channel (the dashed line in Fig. 2). When a packet arrives at a node, it is inserted into the appropriate queue according to its traffic type (e.g., CBR or VBR). When a transmitter is ready, it de-queues the packets from the corresponding queue at a proper time.

In an M-WDMA network, all packets are assumed to have fixed size $L$ b and are transmitted in a networkwide synchronous mode. The transmission time of a packet is refereed to as a *time slot*. In other words, the transmission of a packet takes a *slot* time. The slot time is denoted as $T_{\text{slot}} = L/B$, where $B$ is the data transmission rate.

## III. The M-WDMA Protocol and Its Frame Structure

The M-WDMA MAC protocol is designed to support multiclass traffic streams through the integration of multiple MAC protocols. The implementation issues of the M-WDMA protocol are not simply to have a number of subprotocols within a single network, but also to efficiently utilize the bandwidth by considering the effect of tuning time overhead.

### A. The M-WDMA Frame Structure

To complete a packet transmission, wavelength coordination has to be established in a WDM network. In an M-WDMA network, the transmitters have to tune to the home channel of the destination node before the transmission takes place. However, since there is only one receiver at each node, if more than one transmitter sends data to the same receiver, that would produce what is known as *receiver collision*. When two or more receivers share the same wavelength (the case when $N > C$), only one of them can be active during the communication;, otherwise, it may cause *channel collision*. Finally, if at a prescheduled transmission time, the transmitter is occupied by another transmission at the same node, this may result in a *transmitter collision*. Consequently, with the occurrence of any of these collisions, the transmission would be unsuccessful. The major function of the MAC protocol is to properly avoid these collisions and efficiently transmit the packets.

Different types of WDM MAC protocols use different schemes to avoid or resolve these collisions. In a TDMA protocol, packet transmissions avoid collision by properly arranging the schedule maps. In a reservation-based protocol, packet collisions are avoided by actively arranging the packet transmission coordination according to the transmission requests. In a contention-based MAC protocol, when packets collide, they are required to be retransmitted again using, for example, a binary back-off algorithm.

As described in Section II, each node in an M-WDMA network is equipped with three tunable transmitters. Thus, the three transmitters can operate in a *pipeline* fashion. That is, when one transmitter is transmitting a packet, the other transmitters can start their tuning process. As a result, the three types of transmissions (TDM, CNT, and RSV) from a node cannot take place at the same time. This has led us, among other considerations, to organize the three types of transmissions into a time *frame*. A frame consists of three *segments*: a TDM segment with length $L_{\text{TDM}}$ slots, an RSV segment with length $L_{\text{RSV}}$ slots, and a CNT segments with length $L_{\text{CNT}}$ slots. The TDM transmissions can only be started in a TDM segment, and analogously for CNT and RSV transmissions, respectively. To make sure that the TDM segments appear in periodic constant intervals, the frame length $L_{\text{frame}}$ is fixed (i.e., $L_{\text{frames}} = L_{\text{TDM}} + L_{\text{RSV}} + L_{\text{CNT}}$ is set as a fixed parameter). A transmission example of our M-WDMA MAC protocol is illustrated in Fig. 3.

In this figure, the horizontal direction denotes time and the vertical direction denotes spatial location of the backlogged nodes (the number in the vertical axis represents the location of the nodes, say, 1 means the location of node 1). The white segments denote the TDM segments, the light-shaded segments are the CNT segments, and the dark segments are the RSV

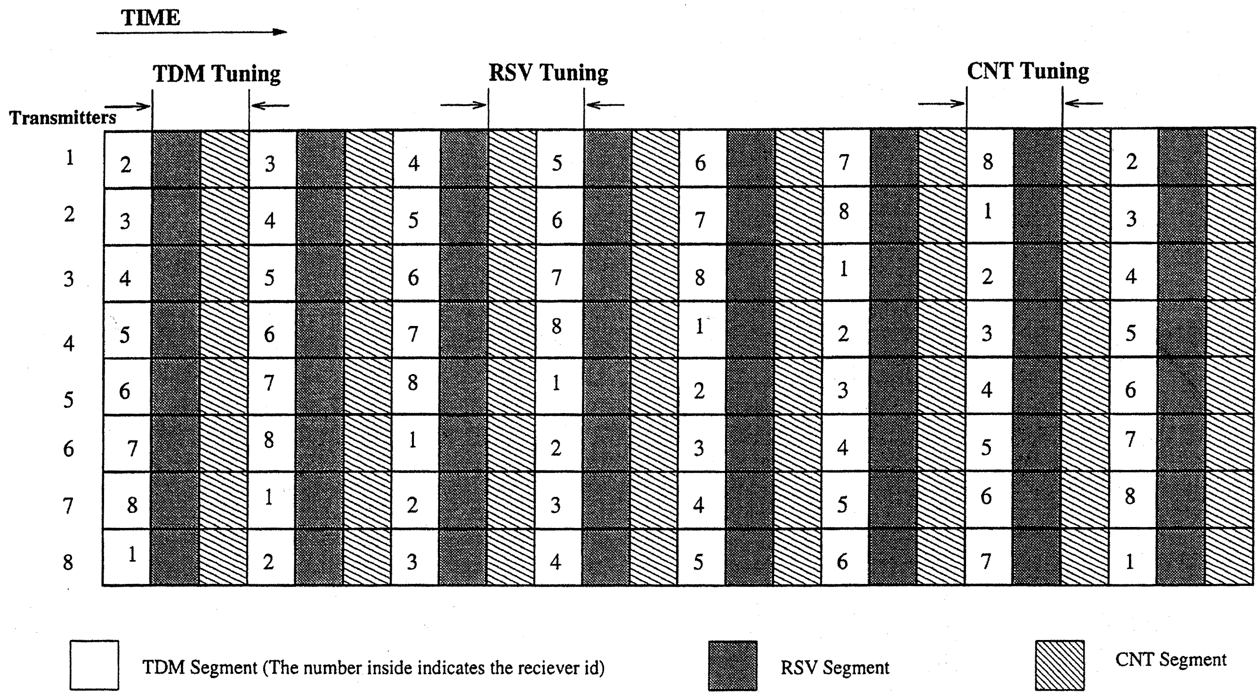



Fig. 3. M-WDMA transmission schedule example.

segments. The numbers in the TDM segments identify the home channel number that the underlying TDM segment ties to, according to the TDM protocol. For example, in the second row and fourth frame, the frame *ID* is 6. It means at that time, the current TDM segment is transmitted on channel 6. Next, in the fifth frame, the frame will be transmitted on channel number 7. All these transmissions happened at node 2, i.e., the TDM transmitter on node 2 is tuned to channel 7 at the time the fifth frame comes. From the figure, we can see, ideally, all channels can be utilized and all nodes can be transmitting even though some of their transmitters may be in a nontransmitting mode (tuning to a certain channel).

The frame length can have an implication on the operation and performance of an M-WDMA network. Hence, it should be properly chosen so that the tuning time can be efficiently masked. That is

$$\begin{aligned} L_{\mathrm{TDM}} + L_{\mathrm{RSV}} &> \Gamma \\ L_{\mathrm{TDM}} + L_{\mathrm{CNT}} &> \Gamma \\ L_{\mathrm{CNT}} + L_{\mathrm{RSV}} &> \Gamma. \end{aligned}$$

Since $L_{\mathrm{frame}} = L_{\mathrm{TDM}} + L_{\mathrm{RSV}} + L_{\mathrm{CNT}}$, then $L_{\mathrm{frame}} > 3\Gamma/2$.

In particular, the length of the frame directly affects the bandwidth allocated to TDM segments. Hence, the total TDM bandwidth is given by

$$B_{\mathrm{TDM}} = \frac{L_{\mathrm{TDM}} B}{(N-1) L_{\mathrm{frame}}}.$$

When $L_{\mathrm{frame}}$ increases, $B_{\mathrm{TDM}}$ decreases.

In an M-WDMA MAC protocol, a frame segment is further divided into slots (with length $L$ b) as shown in Fig. 4 so that multiple nodes can share the same segments. Since the TDM segments and the RSV segments are used in a predetermined manner, then at any one time, only one source node will be using a specific segment, while the CNT segments of a frame can be used by multiple nodes.

The control channel is a shared-access channel that is used by all nodes, which is controlled by a pure TDMA scheme. Its cycle time is $T_{\mathrm{ctrl}}$. A control channel cycle $T_{\mathrm{ctrl}}$ consists of $N$ minislots, each of which is assigned to a node. Every node can put on its reserved minislot any necessary information that needs to be known by all other nodes. At the end of a cycle, all nodes receive global information about the status of the whole M-WDMA network. Based on this information, all control procedures can be done by the nodes locally and networkwide synchronously. The control procedures include: 1) reservation requests which are used by the reservation subprotocol; 2) collision acknowledgments; and 3) bandwidth reallocation and notification which is used in an improved version of M-WDMA, named M-WDMA+ (which is discussed in Section III-D). We will discuss the protocol and operation of the control channel in more detail after we depict the three subprotocols.

### *B. The M-WDMA Protocol*

The M-WDMA MAC protocol is an integrated protocol. It includes three subprotocols: A TDM subprotocol, an RSV subprotocol, and a CNT subprotocol. Under the regulation of an M-WDMA frame format, these three subprotocols operate independently.

*1) The TDM Subprotocol:* The operation of the TDM subprotocol within our M-WDMA network is basically an *interleaved TDMA* MAC protocol [4]. The only differences between our TDM subprotocol and TDMA are that in an M-WDMA network, we take tuning time into consideration,

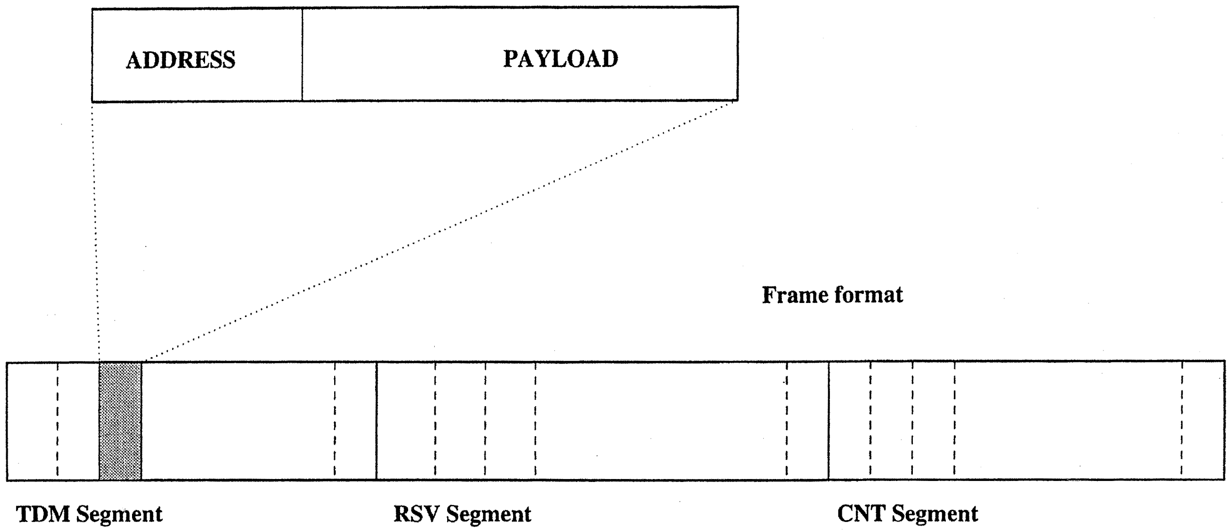



Fig. 4. M-WDMA frame and slot formats.

and the TDM segment length may change dynamically. Using the M-WDMA protocol, at the border between a TDM segment and an RSV segment, the TDM transmitter starts to tune to the next channel. Since the tuning order is prefixed, no extra information exchange is needed in the control channel. The segment length adjustment is made according to the *dynamic bandwidth allocation algorithm*, which is discussed subsequently.

*2) The RSV Subprotocol:* In an M-WDMA MAC protocol, the RSV packet transmission is controlled using a *multiple token* method [25]. Each channel is associated with a "token." A node can send its packets onto the destination channel only if it holds the corresponding token. An obvious advantage of using a token-based scheme in our RSV subprotocol is that it can efficiently support very bursty traffic streams, and its implementation can be simple. The disadvantage of this scheme is that the channel efficiency may not be high when compared with a perfect scheduling scheme, since in a frame, only one single node is allowed to access the RSV segment at a time.

In our implementation of the multiple-token mechanism, the tokens are not explicitly issued. They are in fact *virtual* tokens. In each cycle of the control channel, the M-WDMA nodes broadcast their transmission requests to all nodes. At the end of the cycle, all nodes synchronously execute a *token rotation algorithm* to determine the token distribution in the coming frame. The token rotation algorithm is depicted in Fig. 5.

Here, a prefixed order can adopt any order. However, we usually use an order of node *ID* so that all nodes can obtain identical results when the algorithm is executed in a distributed manner. If an M-WDMA network has an unbalanced load (e.g., client–server traffic), we can change the token order to favor the loaded nodes. In this way, more transmission chances are allocated to those nodes. It is worth noting that there are no transmission conflicts among nodes when using the token rotation algorithm since each node can only send one transmission request, and there is only one token available for any particular channel. The last part of the token rotation algorithm is to arbitrarily allocate the spare token to some idle nodes. To ensure that

```
[Token Rotation Algorithm]
 Put all the tokens with their holder into
a token list;
 Put all the nodes with a request into
checking list;
 For every token do {
   Check if the token holder still needs
the token
   If (it is true and does not exceed the
token holding time)
   then {
    allocate the token to the current
holder;
    remove the node from the checking
list;
    remove the token from the token list;
    continue to the next token;
   };
  Find another node waiting for the token
in a pre-fixed order;
  if (Found) {
    Allocate the token to that node.
    remove the node from the checking
list;
    remove the token from the token list;
     }
      }
 For each of the token remaining in the
token list, do {
 Find a node in the checking list, which
has no token;
 Allocate the token to the node;
 remove the node from the checking list;
 remove the token from the token list;
      }
```

Fig. 5. Token rotation algorithm.

the algorithm gets identical results, the allocation is also carried out in a prefixed order.

The token rotation is executed according to the control channel timing. Thus, the rotation results have to be synchronized with the data transmission (i.e., the frame timing). Hence, any rotation results are effective for the first coming RSV segment after each execution of the token rotation algorithm.

*3) The CNT Subprotocol:* The CNT subprotocol of our M-WDMA MAC protocol is similar to the *interleaved slotted ALOHA* [4]. The active nodes compete for the slots in the current CNT segment. In case there is a collision, retransmission is scheduled according to a binary back-off algorithm. In the context of the M-WDMA, two problems have to be solved. The first one is how to handle the acknowledgments. In M-WDMA, there are two receivers at each node: one is the home channel receiver, and the other is the control channel receiver. Although we can detect the collision at the receiver side, the notification of collision can consume a portion of the control channel bandwidth. In our design, the collision notification is issued in the early part of the control channel cycle. Considering the round-trip propagation delay, the total acknowledgment delay is the propagation time plus the control channel cycle time. In case the propagation delay is large, we can adopt a larger *sliding window* scheme to reduce its affect.

The second problem that needs to be addressed is the retransmission of collided packets. When a collision occurs, retransmission is inevitable. However, the scheduled retransmission should not cross the border of a CNT segment; otherwise, it will collide with other types of transmissions. However, if the back-off is limited to within one CNT segment, it may cause even more collisions. Thus, a CNT segment counter is set at each node to resolve this problem. When a node encounters the last slot of a CNT segment, its CNT segment counter stops counting. When the first slot of a CNT segment arrives, the counter starts to tick again. In this way, the retransmission time is calculated in terms of the counter rather than the real slot numbers in a frame. As a result, retransmission can be carried out across the frames.

### *C. Control Channel Configuration*

The control channel operates in a TDMA manner independently from the data transmission. One cycle consists of $N$ *minislots*, each of which is designated to a node. The cycle length is determined by the amount of information required and the number of nodes involved. Since the TDM subprotocol is a preallocated protocol, there is no need to exchange control information before transmission through the control channel. The RSV subprotocol is a multiple-token passing protocol, so the information needed to process the token rotation is only the states of the queue. We can use a bit map to represent the reservation request, each node taking a bit. If the bit is "1," that means that the queue is nonempty, and there is at least one packet waiting for transmission; otherwise, when it is "0," that implies the queue is empty and there are no transmission requests. Therefore for each node, one bit is sufficient. If there are $N$ nodes in the network, each minislot of the control channel needs $N$ b to indicate the requests information (including itself for simplicity). For the CNT transmission acknowledgment, we can use the bit map, again, to indicate the success or failure during the contention. Since there are $L_{\mathrm{cnt}}$ slots in a frame for contention, then $L_{\mathrm{cnt}}$ bits are enough for a CNT transmission acknowledgment. If a bit is "1," it implies the corresponding slot is successful. Since this information is broadcast to all nodes through the control channel, the source node can obtain this information and start to prepare the next transmission (or tuning). If a bit is "0," then there is a collision in the corresponding slot; thus, the source nodes involved in this collision have to start the retransmission process. Consequently, the length of a minislot is determined by $N + L_{\mathrm{cnt}}$. As a result,

$$T_{\mathrm{ctrl}} = \frac{N(N + L_{\mathrm{cnt}})}{R}$$

where $B$ is the channel bit rate. To realize the frame-by-frame reservation and collision detection, the control cycle has to complete within a frame time, that is,

$$\frac{N(N + L_{\mathrm{cnt}})}{l_{\mathrm{slot}}} < L_{\mathrm{frame}}$$

where $L_{\mathrm{slot}}$ is the slot length in bits. Solving the inequality, we get

$$N < \frac{L_{\mathrm{cnt}}}{2}\left(\sqrt{\left(1 + \frac{4 l_{\mathrm{slot}} L_{\mathrm{frame}}}{L_{\mathrm{cnt}}^2}\right)} - 1\right).$$

This leads to a network scale limitation. For example, if we choose, $L_{\mathrm{cnt}} = 10$ slots, $L_{\mathrm{frame}} = 30$ slots, $l_{\mathrm{slot}} = 424$ b, then the number of nodes should be $N < 107$.

By adding another receiver to do the collision detection, the $L_{\mathrm{cnt}}$ can be eliminated, then the scale limitation becomes $(l_{\mathrm{slot}} L_{\mathrm{frame}})^{1/2}$ (which is 11, as in the previous example). Alternatively, by releasing the requirement of frame-by-frame reservation, the square-root item can increase considerably, as a result, the restriction becomes more relaxed.

### *D. Dynamic Bandwidth Allocation: M-WDMA+*

Due to the randomness of the traffic streams, the bandwidth required by a particular traffic stream cannot be accurately estimated. It is possible that some of the segments are overloaded, while some segments are partially filled, resulting in a low efficiency. Although some of the prediction methods based on long-term statistics [14] are useful in this context, dynamic and adaptive solutions seem to be more appropriate, since they can match the transmission requirements more accurately. In an M-WDMA network, the frame length is fixed, which implies that utilizing all the frame slots must result in the full utilization of all available bandwidth. Therefore, for a given set of traffic streams, if we can properly allocate the portion of bandwidth that the three types of segments should take, then all available bandwidth would be properly utilized. Fortunately, in an M-WDMA, different classes of transmissions are grouped into a single frame, so the adjustment can be easily realized by adjusting the segment sizes.

In order to accomplish this dynamic adjustment, we have to be able to detect the traffic load. That is, we should be able to determine the segment size required by the traffic streams so

[Dynamic Bandwidth Allocation Algorithm]
Given $\hat{l}_{\rm TDM}$, the number of TDM requests
$\hat{l}_{\rm RSV}$, the number of RSV requests
$l = L_{\rm frame}$
1. if $\hat{l}_{\rm TDM} < Q_{\rm TDM}$ then $L_{\rm TDM} = \hat{l}_{\rm TDM}$;
else $L_{\rm TDM} = Q_{\rm TDM}$;
$l = l - L_{\rm TDM}$;
2. if $\hat{l}_{\rm RSV} < Q_{\rm RSV}$ then $L_{\rm RSV} =_{\rm RSV}$;
else $L_{\rm RSV} = Q_{\rm RSV}$;
$l = l - L_{\rm RSV}$;
3. $L_{\rm CNT} = l$;

Fig. 6. Dynamic bandwidth allocation algorithm.

as to meet their QoS requirements. By looking into the characteristics of the different types of the subprotocols within an M-WDMA network, we can find the solution. The CBR traffic is connection oriented in nature. Hence, once a CBR connection is established, the connection is usually kept for a relatively long time. During the connection time, the bandwidth required for the connection is known. Then, the number of slots required in each frame can be derived before the transmission takes place. For the VBR traffic streams using the reservation-based subprotocol, the required bandwidth varies a lot because of the burstiness of the data. However, the reservation subprotocol requires the source node to submit its transmission request before the transmission can proceed. In an M-WDMA network, the transmission requests are broadcast to all other nodes through the control channel. Therefore, every node can obtain the number of slots required for the RSV segments in every frame, before the transmission is started. Having determined the TDM and the RSV segment lengths, the remaining slots of the frame are left to the CNT segment. As a result, before a frame is used, we already know the sizes of all its segments.

Once the segment sizes required are known, bandwidth allocation becomes straightforward and more flexible. Our strategy is to assign priorities to the subprotocol segments in the following order: the TDM segments, the RSV segments, and then the CNT segments. That is, the TDM requirements are considered first (we allocate CBR traffic first). Next, we serve the VBR traffic streams through the RSV segment if there is bandwidth left. Finally, the remaining bandwidth, if available, is all given to the CNT segment. This allocation algorithm is described in Fig. 6.

In this algorithm, $Q_{\rm TDM}$ and $Q_{\rm RSV}$ are system parameters that indicate the maximum number of slots that a TDM segment and a RSV segment can have. These two parameters reflect the bandwidth allocation strategy. Generally, these two parameters should satisfy

$$\begin{aligned} Q_{\rm TDM} &> 0 \\ Q_{\rm RSV} &> 0 \\ L_{\rm frame} - Q_{\rm TDM} &> \Gamma. \end{aligned}$$

Fig. 7 shows one example of applying our dynamic bandwidth allocation algorithm. The histogram is obtained using the following parameters: $L_{\rm frame} = 30$, $Q_{\rm TDM} = 14$, $Q_{\rm RSV} = 25$, $\Gamma = 16$, and the normalized traffic load is 30%. From the histogram, we can see that more bursty data can be efficiently accommodated and more available bandwidth can be exploited by the CNT subprotocol when compared with the static allocation scheme (i.e., M-WDMA). The subprotocol with the highest priority, the TDM subprotocol, just uses the bandwidth it needs.

To determine the segment configuration for each frame in a distributed manner, every node has to record the current traffic load in each frame and in each channel. In this case, the information required for reservation in the RSV segment is not only the transmission of the request, but also the number of slots requested. This results in a longer control cycle and longer transmission delay. To solve the problem, we split the request part into two alternating stages. In first stage, the minislot is the same as in M-WDMA where a minislot consists of a request bit map and a CNT acknowledgment bit map. At the end of this cycle, the token rotation is determined. Then, the RSV transmitter can start to tune to the destination channel. In the next cycle, the second stage, the minislot consists of *number of requests* and the CNT acknowledgment bit map. Because the number of slots requested is needed only at the beginning of the frame in which these requested slots are loaded, and the second stage is executed in parallel with the transmitter tuning since the end of first stage, the necessary information for the RSV segment can all be available on time *without* extra delay due to the extra stage in the control channel. The number of requested slots is expressed in $l_{\rm req}$ bits. In the worst case, where all tokens are requested, the total number of nodes that can be supported is $1/l_{\rm req}$ of that in M-WDMA. Hence, there is a tradeoff between the network scalability and utilization of the network. If the $l_{\rm req}$ bits takes $l$ slots ($l > 1$) as the unit, then for the same range of burst length, fewer bits are needed, but the utilization would be degraded because the allocation is carried out in the units of $l$, no matter whether it is necessary or not. On the other hand, if $l = 1$, no wastage is incurred but longer $l_{\rm req}$ bits are required for the same range of bursty length. In the remainder of the paper, we will eliminate the boundary effects and assume the control cycle is sufficient for a given network size with frame-by-frame dynamic allocation.

## IV. Mathematical Modeling of the M-WDMA Network

This section investigates the performance of the M-WDMA MAC protocol through simple mathematical models. We focus our attention on determining the packet transmission delays using our M-WDMA MAC protocol. The dynamic allocation of bandwidth is not considered here. This is left for a sequel of this paper. In an M-WDMA MAC protocol, three subprotocols coexist together, but they are not active simultaneously. When one of the subprotocols is in use, the other subprotocols are idle from the point of view of the channels and the nodes. Given that the bandwidth (subprotocol segment sizes) is statically allocated in the M-WDMA, the three subprotocols can be thought of as if they operate using three different networks. These three *virtual* networks have a bandwidth equal to the bandwidth portion that is allocated to that subprotocol in an M-WDMA network. This is reasonable under the assumption that the three classes of traffic streams are independent from

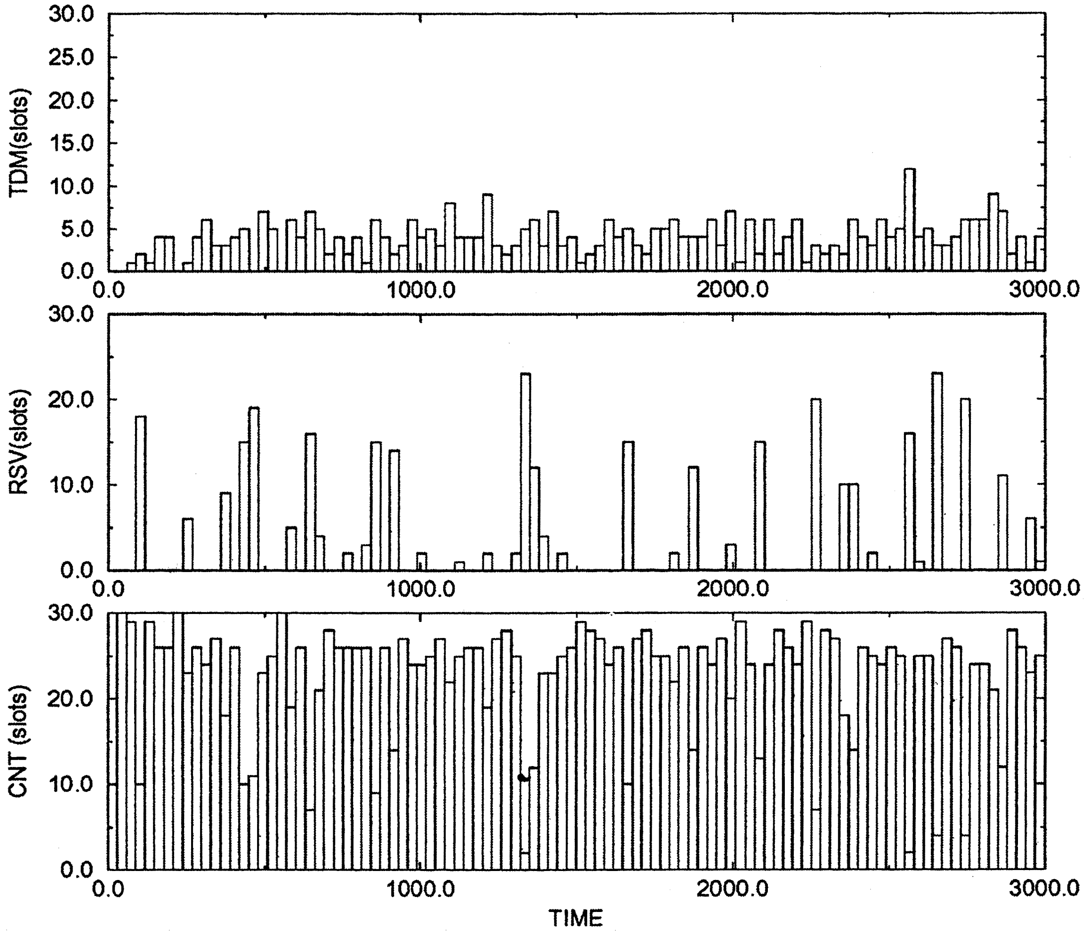



Fig. 7. Histogram of an M-WDMA+ bandwidth occupation.

each other. We term this assumption as *protocol independence assumption*. With this assumption, the three subprotocols can be modeled separately.

A packet has to wait in its queue until it is its turn to be served, no matter which subprotocol would serve it. That is, the server is not always available. It simply alternates between the states of *idle* and *busy*. In other words, a server (transmitter) would poll the queues containing the packets each time it can serve a packet. As a result, it is reasonable to apply a polling system analysis in our case [8], [9], [21], [22] for modeling our M-WDMA network. We assume that all the nodes in the M-WDMA network have equal probability to generate packets and also have equal probability to be the destination of a packet. Let $\lambda$ be the network normalized traffic load, and $\lambda_{\rm TDM}$, $\lambda_{\rm RSV}$, and $\lambda_{\rm CNT}$ be the mean traffic loads for the individual segments of the subprotocols, respectively. The relationships between these values are

$$\lambda_{\rm TDM} = \frac{L_{\rm TDM}\lambda}{L_{\rm frame}}$$
$$\lambda_{\rm RSV} = \frac{L_{\rm RSV}\lambda}{L_{\rm frame}}$$
$$\lambda_{\rm CNT} = \frac{L_{\rm CNT}\lambda}{L_{\rm frame}}.$$

Without loss of generality, we use $\lambda_i$ to denote the mean traffic load that arrives into queue $i$. The service time for queue $i$ is called the *visit time* of queue $i$, a generally distributed random variable $B_i$ with mean $b_i$ and second moment $b_i^{(2)}$. Since the server visits the queues in a cyclic order, the reference to the queue indexes is done *modulo* $N$. After visiting queue $i$, the server switches to queue $i+1$. The period during which the server switches from queue $i$ to queue $i+1$ is called the *switch-over period*, denoted by a random variable $S_i$, which is assumed to be a generally distributed random variable with a mean $s_i$ and a second moment $s_i^{(2)}$. It is assumed that the packet arrival times, the packet service times, and the switch-over processes are mutually independent. For a particular queue $i$, the period between two consecutive visits to queue $i$ is named $i$ *cycle*. The duration of an $i$ *cycle* is denoted by $Y_i$. Let $R_i$ be the residual time in an $i$ *cycle*. Our purpose here is to get the mean packet delay $E[W_i]$ in queue $i$ in steady state. Let $X_i$ denote the number of packets present in queue $i$ at an arbitrary moment. Under the assumption of exhaustive service, the waiting time of an arbitrary packet at queue $i$ consists of the following two components:

1) the time since the packet arrives until the next visit of the server to queue $i$, that is, $R_i$;
2) the time required for the server to serve the $X_i$ packets found at the queue during the last visit time.

Fig. 8 illustrates the relationship between these time components.

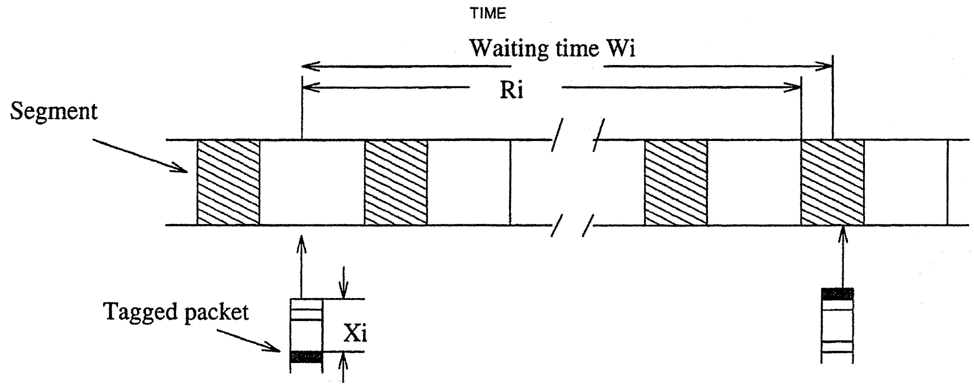



Fig. 8. Transmission delay components of a tagged packet.

By observation, we have

$$E[W_i] = E[R_i] + E[X_i]. \tag{2}$$

By Little's law, we get

$$E[X_i] = \lambda E[W_i]. \tag{3}$$

This is true according to the Poisson arrivals see time averages (PASTA) [9] property. When an arbitrary packet arrives at an arbitrary time, the number of packets ahead of it in the queue is also considered the mean number of packets in the queue.

We approximate $E[R_i]$ as [9]

$$E[R_i] = \frac{E[Y_1]E[R_1]}{E[Y_i]}. \tag{4}$$

Consequently, we can obtain a general form for the mean packet delay which is given by

$$E[W_i] = \frac{E[R_i]}{(1-\lambda_i)}. \tag{5}$$

Next, we apply this general form onto the individual subprotocols of the M-WDMA network to finalize our mathematical model.

### A. The TDM Subprotocol Model

In the TDM subprotocol, the transmission is organized according to a pre-fixed schedule table, as shown in Fig. 3. In addition, the cycle time is fixed to $(N-1)T_{\text{frame}}$. Since the schedule for transmission operates in a round-robin scheme, each queue has a fair chance (i.e., preallocated bandwidth) to transmit packets and the TDM segment is of fixed length $L_{\text{TDM}}$. Hence, when an arbitrary packet arrives at an arbitrary time, the expected residual time is given by

$$E[R_i] = \frac{(N-2)T_{\text{frame}}}{2}. \tag{6}$$

Note that the residual time excludes a frame time that is occupied by the underlying node. Using (5), we can get the mean packet delay using the TDM subprotocol as

$$E_{\text{TDM}}[W_i] = \frac{(N-2)T_{\text{frame}}}{2(1-\lambda_i)}. \tag{7}$$

### B. The RSV Subprotocol Model

Using the RSV subprotocol of the M-WDMA network, the $i$ *cycle* is no longer fixed. It depends on the elapsed time the token stays on the other nodes under the control of the RSV subprotocol. When a node gets a token, it may occupy all the slots of the RSV segments in the current frame and may continue to use the RSV segment in the next frames. In case few or no other nodes use this channel, the token may come back again. Hence, the $i$ *cycle* could have large variations. According to the suggestion given in [9], the $E[Y_i]$ can be derived using the recursive formula

$$E[Y_i] = b_i + \sum_{j \neq i} \min(1, \lambda E[Y_j])b_j + s \tag{8}$$

where $s$ is the overall switch-over time. Here, we assume $s = 0$, since when $L_{\text{frame}}$ is used to calculate the cycle, the tuning time (switch-over time) has already been included.

Since we have a fixed frame length, $E[R_1]$ can be chosen in the same way as in the previous subsection. Combining $E[R_i]$ and $E[Y_i]$ into (5), we can get the mean delay of packets $E_{\text{RSV}}[W_i]$.

### C. The CNT Subprotocol Model

The CNT subprotocol in an M-WDMA network is a random access protocol. Every node can access the CNT segment of each frame, so the $i$ *cycle* time is $T_{\text{frame}}$. Due to collisions, however, packets' retransmissions are needed. As a result, the network load is no longer equal to just $\lambda_{\text{CNT}}$. Many mathematical models have been proposed for such a protocol [4]. In our case, we reasonably follow Dowd's model, which is a *semi-Markov process* model. According to [4], the average number of packets in the system is given by

$$E(N) = \sum_{i=0}^{2(B+1)} E[N_i]P_i$$

where $P_i$ is the probability of being in state $S_i$, where the notation of system state $S_i$ implies that there are $i$ packets residing in the buffer. $B$ is the capacity of the buffer. By Little's law, the mean delay is given by

$$E[X_i] = \frac{E[N]}{\Gamma_s} \tag{9}$$

where $\Gamma = \gamma \sum_{i=1}^{B+1} P_i$, and $\gamma$ is the probability of successful transmission.

If we substitute (9) into (2) and given $E[R_i] = T_{\text{frame}}/2$, we can obtain the mean packet delay using the CNT subprotocol of our M-WDMA network.

### D. Maximum Throughput of the Subprotocols

Since the TDM and the RSV subprotocols are collision-free MAC protocols and all the slots in their corresponding segments could be fully utilized, the maximum throughput should be

$$S_{\text{TDM,max}} = \frac{CL_{\text{TDM}}}{L_{\text{frame}}} \tag{10}$$

$$S_{\text{RSV,max}} = \frac{CL_{\text{RSV}}}{L_{\text{frame}}}. \tag{11}$$

Using the CNT subprotocol, we can have packet collisions, and packet retransmission would be needed. Thus, the maximum throughput cannot exceed $1/e = 36.8\%$ [4]. As a result,

$$S_{\text{CNT,max}} = \frac{0.368CL_{\text{CNT}}}{L_{\text{frame}}}. \tag{12}$$

## V. Performance Evaluation of M-WDMA Through Simulation

In this section, we evaluate the performance of our M-WDMA using extensive discrete-event simulations. The purpose of the simulations includes the validation of the mathematical models discussed in the previous section, the quantitative evaluation of the improvement gained by our M-WDMA+ when compared with M-WDMA, and the comparison of the performance of M-WDMA+ with conventional WDMA MAC protocols.

### A. Network Parameters

Table I lists the M-WDMA network parameters that have been used in our simulation.

Note that in the table, the frame time is 12.72 $\mu$s. This implies that if a connection can get one slot in each frame, the bandwidth obtained would be 33.3 Mb/s. For guaranteed bandwidth allocation (TDMA), if the number of nodes is $N = 10$, and $C = N$, the rate of one slot per frame corresponds to a 3.3-Mb/s data rate (ten frames per cycle). This rate can already support two MPEG-I data streams (each around 1.6 Mb/s). Similarly, for an 8-KHz phone call, a one-slot-per-frame rate means that we can support about 200 pairs of phone conversations simultaneously. Higher bandwidth can be obtained in multiples of 3.3 Mb/s by allocating one more slot per cycle.

The peak rate can be obtained through the allocation of slots using the RSV segment. In case the RSV segment size is ten slots, the peak rate can reach 333 Mb/s. Even larger peak rates can be obtained by further extending the RSV segment. In an M-WDMA network, the theoretical available peak rate is 1 Gb/s. In such a case, the whole frame becomes the RSV segment.

TABLE I
M-WDMA Network Parameters Used in Our Simulation

| Parameters | Value |
|---|---|
| Channel data rate | 1 Gb/s |
| Slot size | 53 Bytes |
| Frame size | 30 slots |
| Slot time | 424 ns |
| Frame time | 12.72 $\mu$s |
| Number of Channels | $C$ |
| Tuning Time | 20 slot time |
| Number of nodes | $N$ |
| Maximum buffer size | 200 slots |

As mentioned in the introduction of this paper, urgent messages requiring very small delay would be sent through the CNT segment where a random access protocol is applied. In case no collision occurs, the response time can be expected to be equal to the round-trip propagation delay. Theoretically, when the traffic load for the slotted ALOHA is less than about 20% of the total CNT bandwidth, the collision rate can be very small. As the contention slots can be used at any time, the available effective throughput in the case of one slot per frame can be expected to reach about 7 Mb/s. It is worthwhile to mention again that the benefit from the CNT access is for a very small delay and simple access control, not high throughput.

### B. Traffic Generation Models

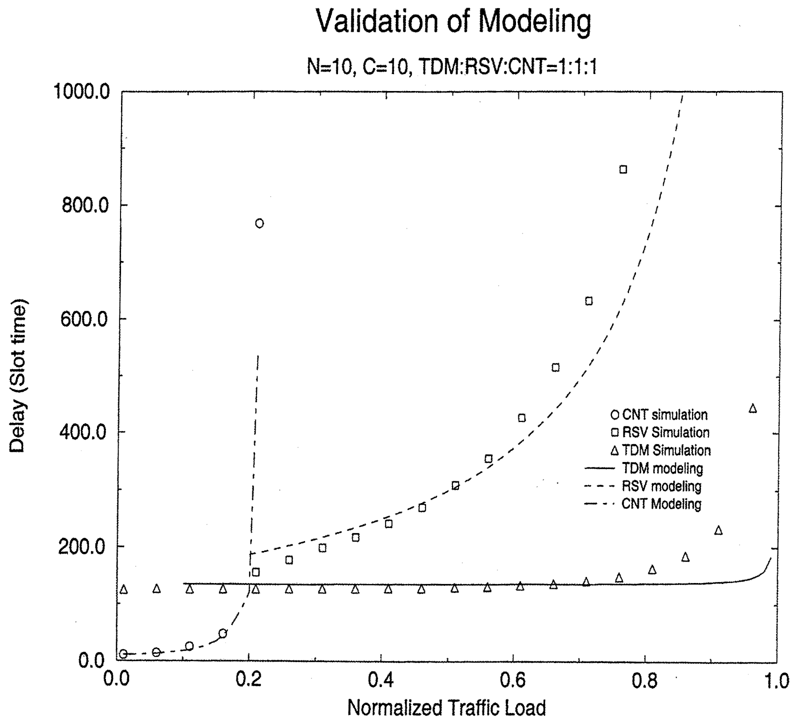
In our simulation study, three types of traffic streams are generated: the CBR sources, the VBR sources with small burstiness (VBR1), and the VBR with large burstiness (VBR2). For the CBR packet sources, the packets are generated periodically. The period is determined according to the traffic load percentage which corresponds to the system parameter $\lambda$. The sources and destinations are randomly chosen in a uniform distribution. The unit of the traffic generation is slot time.

The VBR1 data sources are in the class of connectionless traffic streams with small amounts of data, but expecting quick response, e.g., game control signaling and network management and signaling. The traffic load from these interactive applications is light but may be bursty. In our simulation, the generation of VBR1 traffic follows a Poisson arrival process. In addition, the packets are generated at each node in a batch as a message. The length of a message is short.

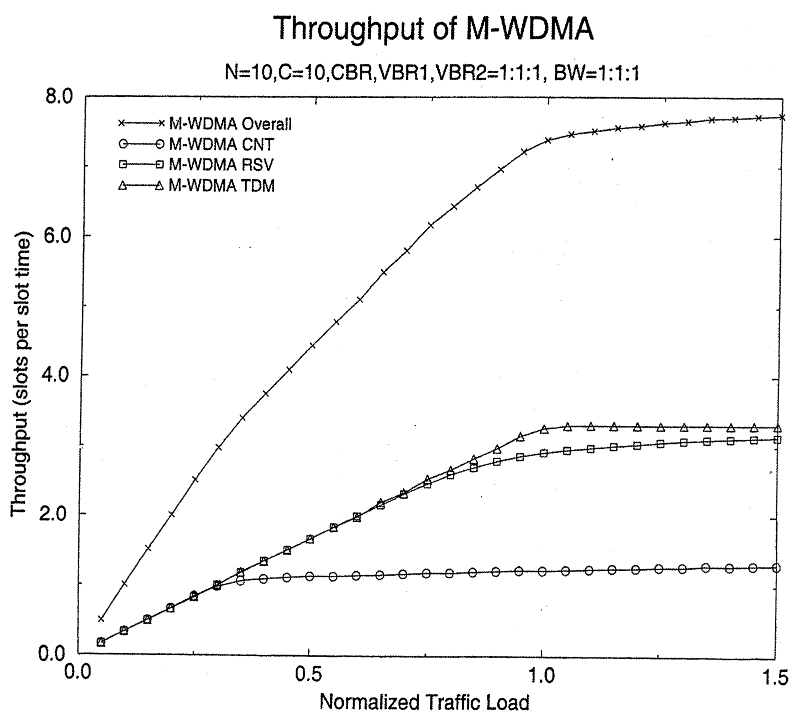
The VBR2 data are generated using an ON–OFF state machine with a Poisson arrival process and a uniformly distributed random ON state duration. The burstiness length of the traffic is a system parameter. This is used to emulate heavy data transfers with less time restrictions, such as data or image transfer and offline video/audio transmission.

### C. Model Validation

Fig. 9(a) and (b) shows the mean delay and throughput of an M-WDMA network. These results are obtained using our mathematical model and using simulation. For simplicity, in both the simulation and mathematical model, we assume the propagation delay to be zero. The number of nodes is $N = 10$ and the




(a)




(b)

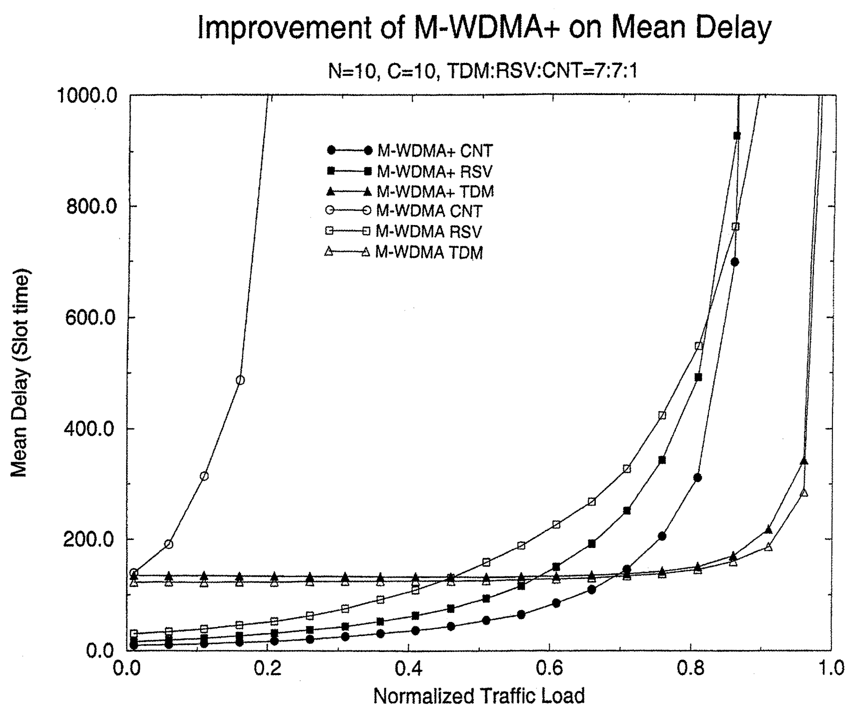
Fig. 9. Performance of the M-WDMA protocol with the fixed bandwidth allocation. (a) Mean delay. (b) Throughput.

number of channels $C = 10$. We also assume that the frame size is $L_{\rm frame} = 30$, and the subprotocol segment sizes are $L_{\rm TDM} = 10$, $L_{\rm RSV} = 10$, and $L_{\rm CNT} = 10$.

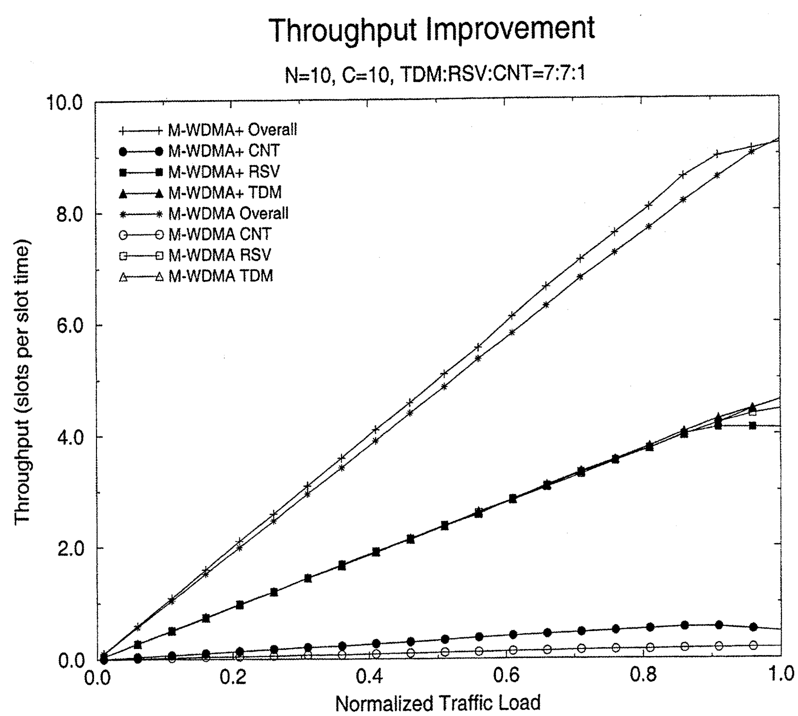
As can be seen from the figures, our simulation and mathematical model results are close to each other. This is an indication of the accuracy of our simple mathematical models for the M-WDMA MAC protocol.

## D. Performance Evaluation

*1) Improvements With M-WDMA+:* By dynamically allocating the M-WDMA bandwidth to different types of traffic streams, the network efficiency and performance can be improved as discussed in Section III. The numerical simulation quantifies this improvement as illustrated by Fig. 10(a) and (b).




(a)




(b)

Fig. 10. Performance improvement of M-WDMA+ over M-WDMA. (a) Mean delay. (b) Throughput.

In our simulation, the three different types of traffic streams are generated: CBR, VBR data streams (VBR1), and low-arrival-rate traffic data (VBR2). The CBR traffic is controlled by varying the stream length and the mean traffic load. The VBR1 traffic is controlled by varying the burstiness of the data (e.g., from 1–20 slots). The VBR2 traffic is similar to the VBR1, but its mean load is smaller. More precisely, the three types of traffic streams are such that 47% of the network load is the CBR data, 47% is VBR1 data, and 6% is VBR2 data. As a result, the bandwidth allocation in the M-WDMA also takes exactly the same proportion for the three subprotocols, respectively.

By applying the dynamic bandwidth allocation algorithm, the channel bandwidth is utilized more efficiently. Hence, the performance can be improved tremendously. From these figures, we can see that the performance of the CBR streams using the M-WDMA and the M-WDMA+ MAC protocols are more or less the same since both of them guarantee the bandwidth required for the CBR traffic streams. However, the performance of the VBR1 traffic streams is improved since the M-WDMA+ can accommodate longer data burstiness than in the "static" M-WDMA. The most significant improvement is with regard to the VBR2 traffic streams, where a random access subprotocol is applied. This is because all the unused bandwidth can be fully utilized by the VBR2 traffic data. For a given traffic load, a larger bandwidth implies lower collision rates and thus better efficiency.

*2) Performance Comparison of M-WDMA+ With ITDMA:* In this subsection and the following ones, we want to illustrate the superiority of integrating various protocols into a single one (M-WDMA) when compared with any individual conventional protocol for a WDM network. We now compare the performance of the M-WDMA+ with the ITDMA protocol. The ITDMA protocol can be found in [4], and it is exactly the same as the TDM subprotocol in our M-WDMA+ network. In other words, the ITDMA can be looked at as our M-WDMA, where all the frame size is a TDM segment. For a fair comparison, exactly the same traffic patterns and proportions are loaded onto both networks, and the same node configurations are also applied, i.e., in ITDMA, the transmitters are also pipelined. The mean delay and loss rate for both networks are shown in Fig. 11(a) and (b).

The CBR traffic performance is more or less the same for both networks, since our M-WDMA+ MAC protocol takes advantage of the ITDMA MAC protocol for this type of traffic. But for other types of traffic streams, the VBR1 and the VBR2, they are much better served by our M-WDMA+ network. In Fig. 11(a), we can see the three different traffic schemes obtain very similar performance using the ITDMA, i.e., the delay is almost constant. This is not good because some urgent messages may unnecessarily wait too long. On the other hand, the M-WDMA+ can better adapt to the traffic characteristics. You may see that when the traffic load is not very high, the delay can be quite low for the RSV and the CNT transmissions. Correspondingly, the advantage is also obvious in the loss rate comparison, as shown in Fig. 11(b). In the high traffic load, we can observe that the performance of the RSV traffic of the M-WDMA+ is a bit worse than that of the ITDMA. This is because the ITDMA does not need to do reservation. While for the TDM traffic, the M-WDMA+ is better than the ITDMA because the head-of-line problem is more severe in the ITDMA. The most significant advantage of the M-WDMA+ over the ITDMA protocol is the CNT part because the M-WDMA+ supplies much more bandwidth for the traffic contention, thus reducing collision.

*3) Performance Comparison of M-WDMA+ and MT-WDMAC:* We now compare the performance of our M-WDMA+ protocol with a reservation-based MAC protocol, namely, MT-WDMA [25]. The MT-WDMA stands for multiple-token WDMA. This protocol has been described previously and is similar to the one we use as part of our integrated MAC protocol for the M-WDMA. The only difference is that all the traffic streams are handled by a unique MT-WDMA protocol in this case, as opposed to multiple subprotocols for the case of the M-WDMA.

By examining Fig. 12(a) and (b), it is obvious that the M-WDMA+ performs better for all types of traffic streams because the MT-WDMA protocol has extra time overhead for the reservation. The RSV part in the M-WDMA+ is better than in the MT-WDMA, since the RSV protocol in the M-WDMA+ is less affected by the head-of-line blocking problem. The TDM part is better because the CBR traffic in the M-WDMA+ avoids the reservation operation. Comparing the CNT parts using the MT-WDMA and the M-WDMA+, we can see that the M-WDMA+ supplies much better performance because the M-WDMA+ gives more bandwidth for packet contention. Again, this is an affirmation of the superiority of using an integrated approach when designing MAC protocols for multimedia applications.

*4) Performance Comparison of M-WDMA+ With ISA:* Here, we compare the integrated MAC approach for our M-WDMA+ with that of a single random access protocol, ISA [4].

ISA is implemented using the same network configurations as our M-WDMA+ protocol. Again, the difference is that in ISA, all types of traffic streams are processed under the same MAC protocol, unlike the M-WDMA+ MAC protocol, where each type of traffic stream is allocated to the proper subprotocol. The results shown in Fig. 13(a) and (b) clearly illustrate the advantages of our integrated MAC approach in M-WDMA+. Note that only when the traffic load is very light, the ISA can achieve lower delay and similar loss rate as the TDM part in M-WDMA+.

*5) Performance Comparison of M-WDMA+ With Conrad:* CONRAD is a state-of-the-art MAC protocol that is proposed for a passive-star-based single-hop WDM local lightwave network [26]. It can efficiently provide services to support data traffic with tight delay constraints, i.e., real-time traffic, directly in the optical layer. In the original paper of CONRAD protocol, only two types of traffic are generated to evaluate its performance. In order to compare its performance with that of M-WDMA+ protocol, we have included three types of traffic, i.e., CBR traffic, VBR1 traffic, and VBR2 traffic. Under the same environment with the same system parameters, we compare the performance of both protocols.

The parameters of the simulation experiments for comparison purposes are as follows. The number of nodes in the system is ten. The number of channels is also ten. The time unit in the simulation is the same as the time slot for the transmission time of one packet. Propagation time is assumed to be zero in order to compare the silent feature of both protocols. The transceiver tuning time is 16 time units. CBR traffic, VBR1 traffic, and VBR2 traffic are generated for both protocols. We consider CBR traffic as real-time traffic, which occupies 30% of total traffic. The VBR1 traffic is 17% in total. The remainder is the VBR2 traffic. For the CBR traffic stream, it is assumed that a message, which consists of several packets, is generated periodically. The message is always considered with ten packets. VBR1 and VBR2 traffics are generated as Poisson processes.

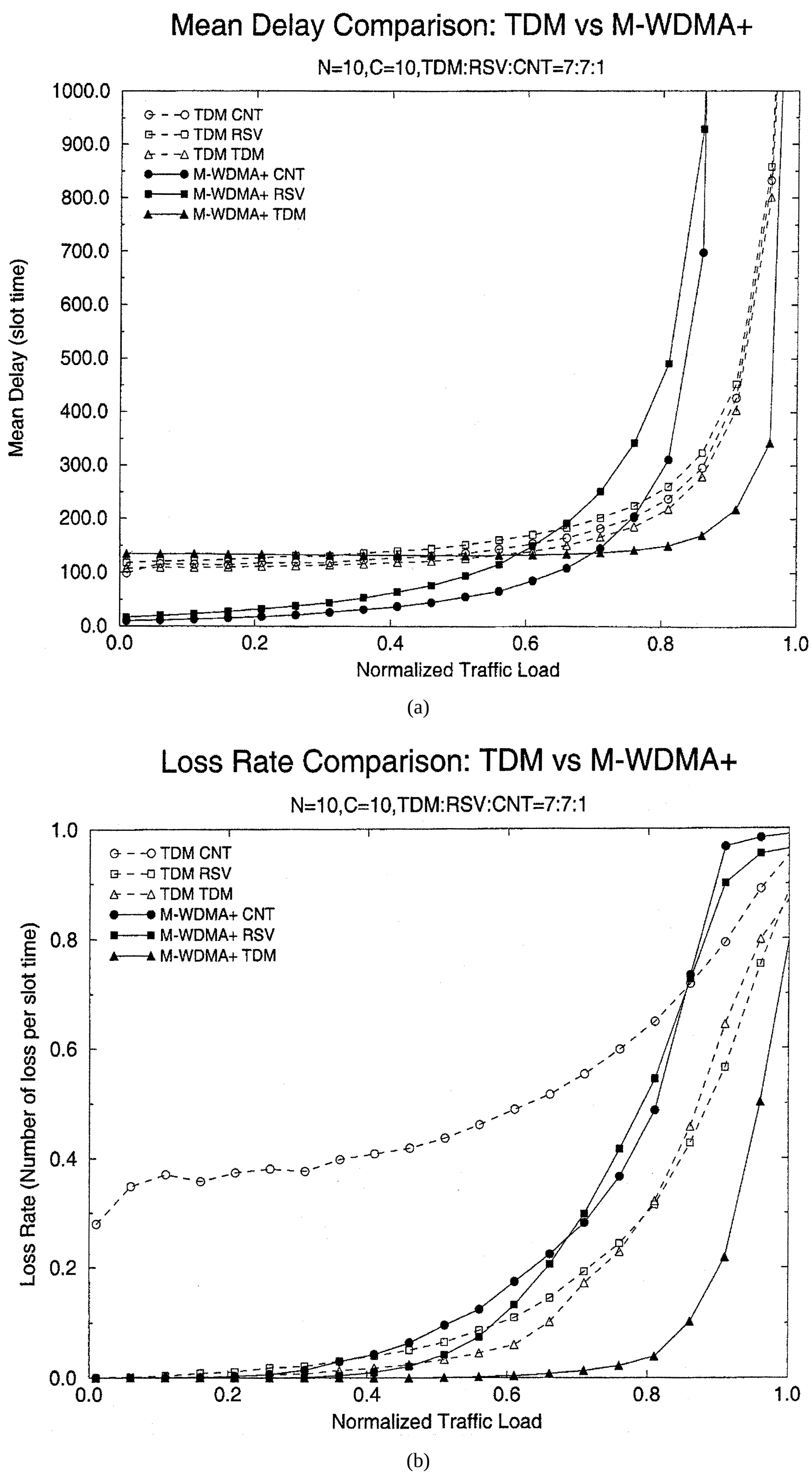
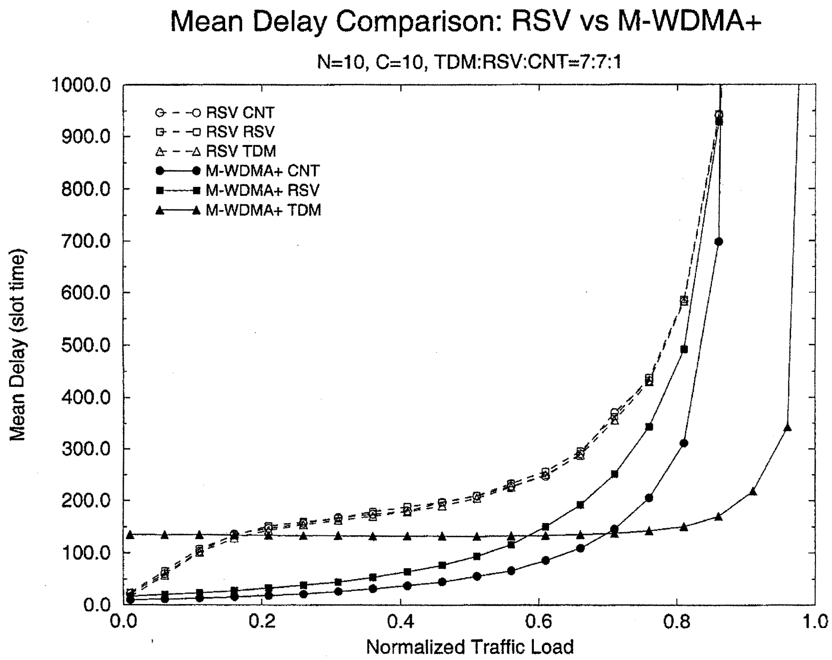


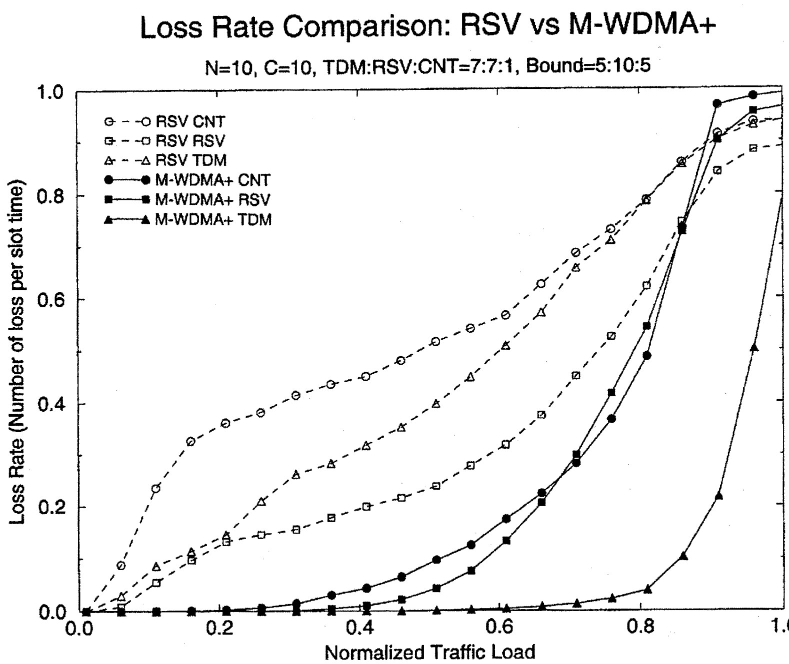

Fig. 11. Performance comparison between M-WDMA+ and ITDMA. (a) Mean delay. (b) Loss rate.

The length of a message from VBR1 traffic follows an exponential distribution with mean as five packets. The length of a VBR2 message also follows an exponential distribution with mean as 15 packets. We assume that the destination nodes are randomly chosen among ten nodes following a uniform distribution. For a fair comparison, exactly the same traffic pattern and proportions are designed for both protocols. We present the simulation results in terms of network throughput and average packet delay of two protocols in Fig. 14.

Fig. 14(a) shows the average packet delay versus normalized offered load for both protocols. An interesting observation from the figure is that by using the CONRAD protocol, the delays of VBR1 and VBR2 are almost convergent together. This is because the packets from VBR1 and VBR2 streams enter into



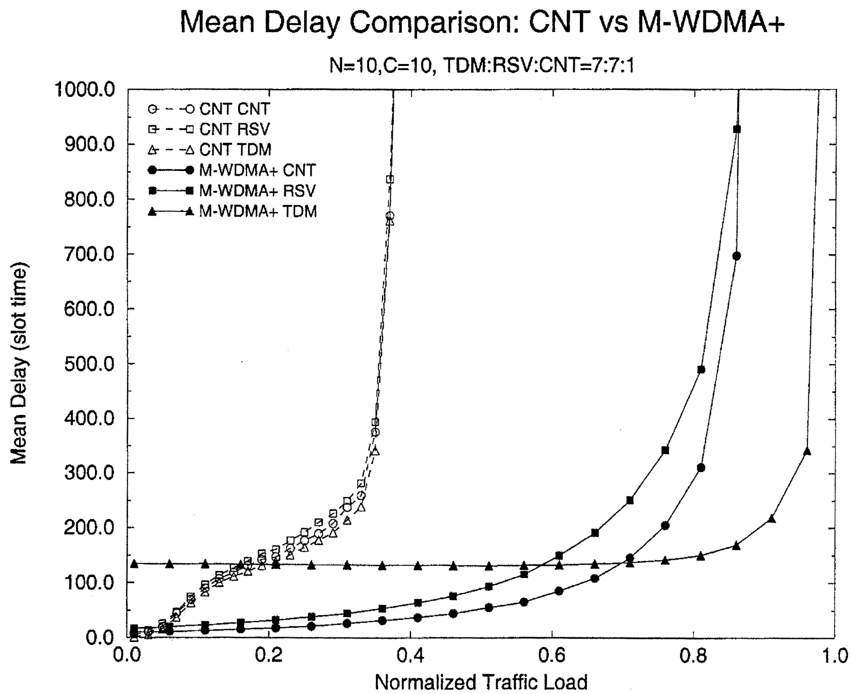

(a)



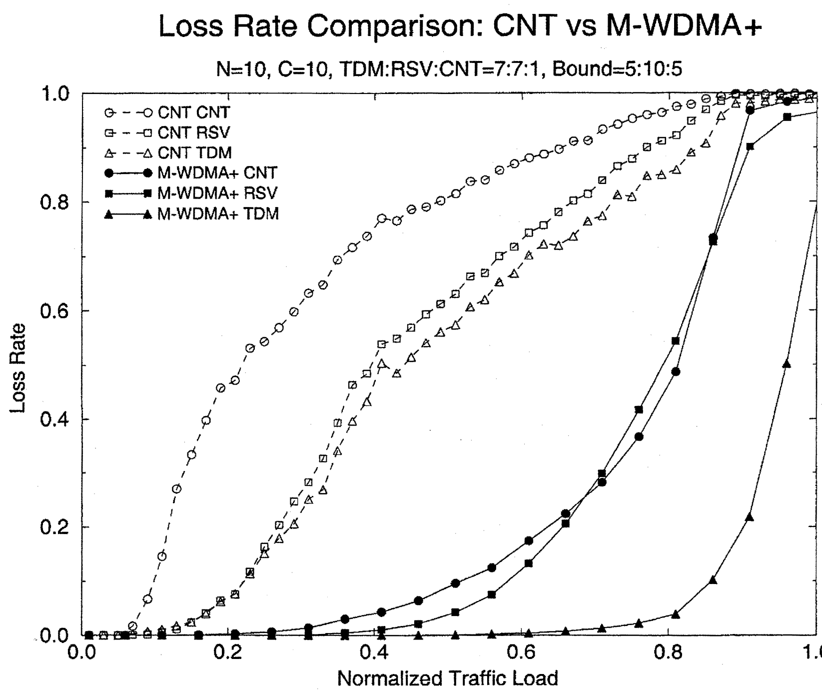

(b)

Fig. 12. Performance comparison between M-WDMA+ and MT-WDMA. (a) Mean delay. (b) Loss rate.

one queue whose discipline is first-in–first-out. We also observe that using the CONRAD protocol, only with very light traffic, the average packet delays for VBR1 and VBR2 traffics are small. The average packet delay for CBR traffic is smaller than those of VBR1 and VBR2 traffics. We can also find that the average packet delays for the three types of traffics using the M-WDMA+ protocol are always smaller than those using the CONRAD protocol. Using the M-WDMA+ protocol when the traffic load is moderate (the normalized offered load is less than 0.6), the average packet delays are not seriously affected when there is a small increment of offered traffic. The delay of CBR traffic is larger than the delays of VBR1 and VBR2. The delay of VBR1 traffic is the smallest among three types of traffic. However, when the traffic load is heavy, the delay of VBR1 traffic is increased significantly and it becomes larger than those of other two traffics. The reason is that the VBR1 traffic uses the CNT subprotocol, which is designed based on a slotted ALOHA scheme. It is well known that under light traffic conditions, the delay in slotted ALOHA could be small. Under heavy traffic load, many more collisions would occur, making the delay large. In summary, the overall performance of the M-WDMA+ protocol is always better than that of CONRAD protocol in terms




(a)

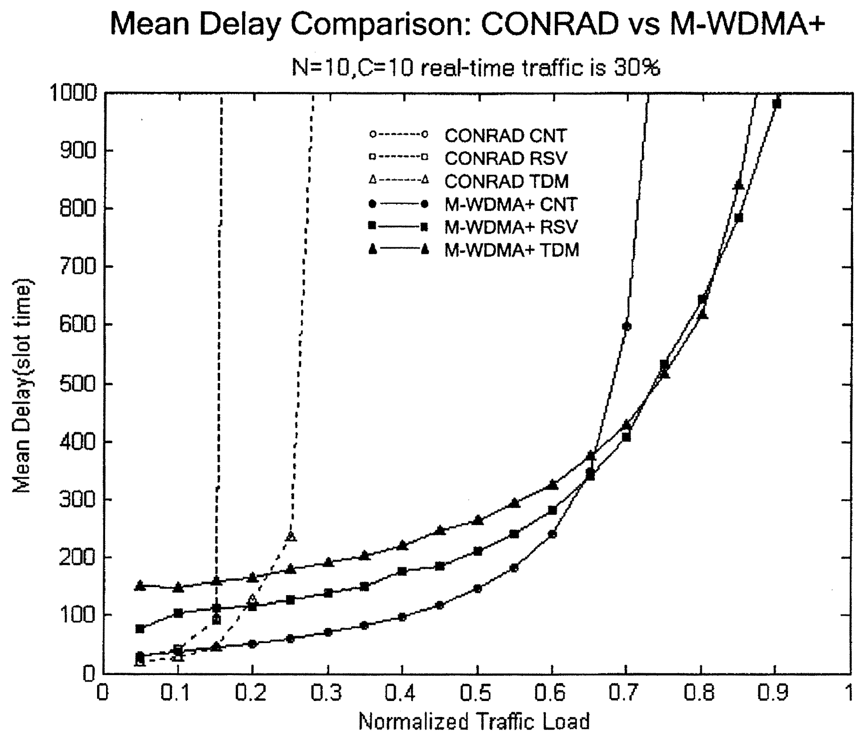



(b)

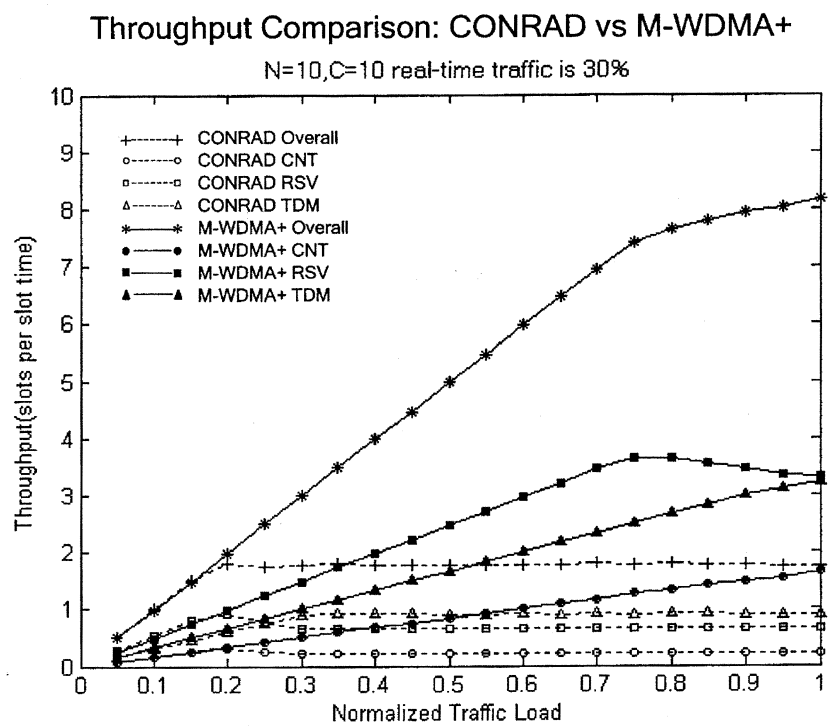
Fig. 13. Performance comparison between M-WDMA+ and ISA. (a) Mean delay. (b) Loss rate.

of average packet delay for all three types of traffic in the network. The primary reason for it is that the M-WDMA+ protocol has employed three subprotocols to adapt to different types of traffic and three transceivers to cover the overhead of the tuning time of them.

From Fig. 14(b), we can see that the M-WDMA+ protocol always has higher throughput than that of the CONRAD protocol. Using the CONRAD protocol, the overall throughput reaches the maximum value when the normalized offered load is about 0.2. Using the M-WDMA+ protocol, the overall throughput keeps increasing and reaches the maximum throughput as the traffic load approaches 1.0. The reason of these facts is that by using the CONRAD protocol, the transceiver has to take time to tune to the specified channel for data transmission. This action makes transmission channels idle for some time slots. It results in low throughput, even under heavy traffic. It is obvious that when the normalized offered load is more than 0.2, the throughput is not increasing any more. However,




(a)

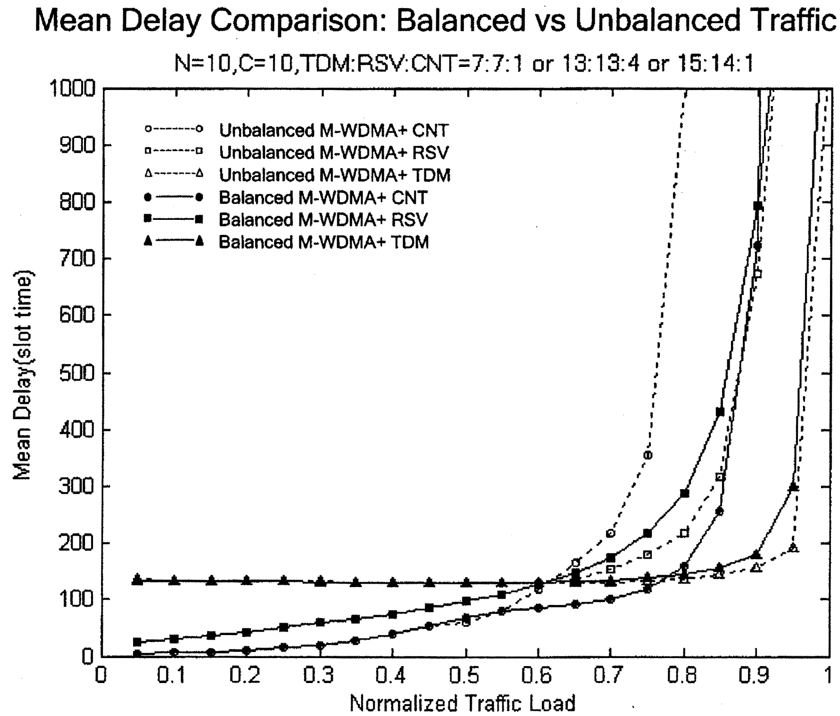



(b)

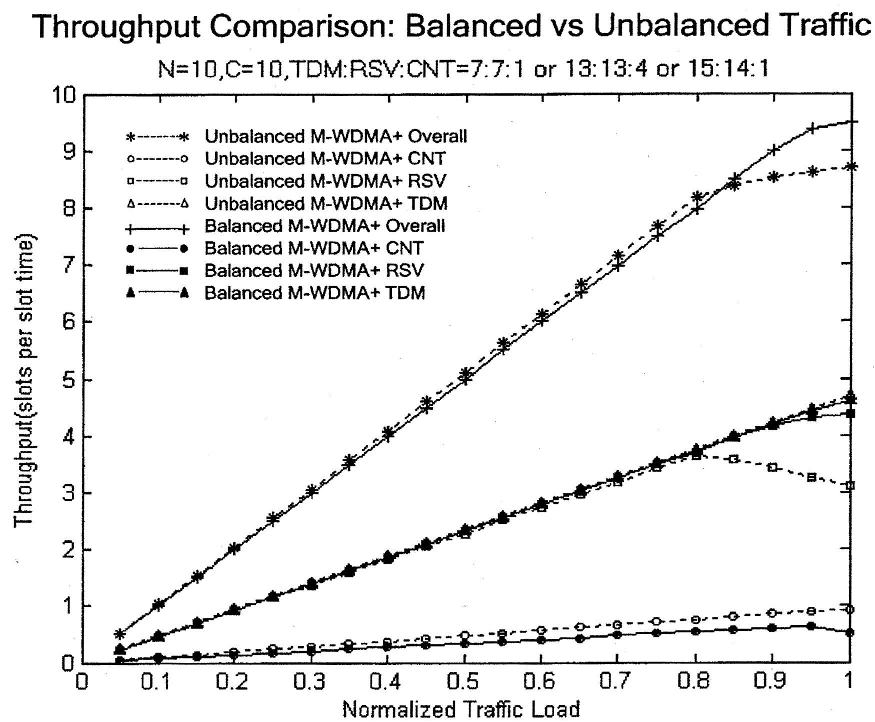
Fig. 14. Performance comparison between M-WDMA+ and CONRAD. (a) Mean delay. (b) Throughput.

there is no such thing when using the M-WDMA+ protocol, simply because of the pipeline working style of the group of transceivers.

*6) Performance of M-WDMA+ Under Unbalanced Traffic:* In the previously described simulation experiments, we evaluated the performance of the M-WDMA+ protocol under the assumption that the network is symmetrically loaded. The traffic ratio of three types of traffic is always the same as $\mathrm{TDM} : \mathrm{RSV} : \mathrm{CNT} = 7 : 7 : 1$ to every node. In this subsection, we present one group of results from the simulation experiment, in which unbalanced traffic is applied to the network. We consider that the traffic ratio of the three types of traffic to be different to different nodes. We compare the performance of our proposed protocol under the balanced




(a)




(b)

Fig. 15. Performance comparison between balanced and unbalanced traffic for M-WDMA+. (a) Mean delay. (b) Throughput.

and unbalanced traffic to show that the performance of the M-WDMA+ protocol can still be good enough to support multimedia traffic even under the unbalanced traffic load.

In this simulation experiment, we assume that in the network, the traffic ratio for three nodes is $\mathrm{TDM} : \mathrm{RSV} : \mathrm{CNT} = 14 : 14 : 2$, the traffic ratio for another three nodes is $\mathrm{TDM} : \mathrm{RSV} : \mathrm{CNT} = 13 : 13 : 4$, and the traffic ratio for thr other four nodes is $\mathrm{TDM} : \mathrm{RSV} : \mathrm{CNT} = 15 : 14 : 1$. The simulation results have been shown in Fig. 15 as follows.

From Fig. 15(a), we can find that under unbalanced traffic condition, for CBR and VBR2 traffic, the average packet delays are quite lower than those under balanced traffic. However, the average packet delay for VBR1 traffic is quite higher than that under balanced traffic. The reason for these facts is obvious.

Under the unbalanced traffic condition with the ratio of three traffic loads as $\mathrm{TDM} : \mathrm{RSV} : \mathrm{CNT} = 14 : 14 : 2$ for three nodes, $\mathrm{TDM} : \mathrm{RSV} : \mathrm{CNT} = 13 : 13 : 4$ for three nodes, and $\mathrm{TDM} : \mathrm{RSV} : \mathrm{CNT} = 15 : 14 : 1$ for four nodes, the amount of the VBR1 traffic is more than that under the balanced traffic condition with the traffic ratio as $\mathrm{TDM} : \mathrm{RSV} : \mathrm{CNT} = 7 : 7 : 1$ for tne nodes. The M-WDMA+ protocol has allocated more bandwidth to the VBR1 traffic. However, due to the feature of CNT subprotocol, the VBR1 traffic can only experience lower delay when the traffic load is lower than 0.6. When the traffic load gets higher than 0.6, the delay of the VBR1 traffic experiences higher delay. On the contrary, the network has allocated less bandwidth to VBR2 traffic and CBR traffic because the amounts of the VBR2 traffic and CBR traffic under the unbalanced traffic condition are less than those under the balanced traffic condition. It is shown that when the traffic load is light, the delays for the VBR2 traffic and CBR traffic under unbalanced traffic condition are almost the same as those under the balanced traffic. When the load gets higher, the delays for VBR2 traffic and CBR traffic are a bit lower than those under the balanced traffic condition. From these facts, it has clearly shown that the M-WDMA+ protocol can dynamically allocate the network bandwidth according to the overall traffic load of the different types of traffic so that the performance of the network in terms of average delay could almost not be affected under the unbalanced traffic loads.

Fig. 15(b) depicts that the network under balanced and unbalanced traffic loads has almost the same overall throughputs when the traffic is not very high (normalized traffic load $< 0.8$). However, when the traffic load becomes heavier, the throughput of VBR2 traffic under unbalanced traffic load is lower than that of the VBR2 traffic under the balanced traffic load. It causes the overall throughput of the network under the unbalanced traffic load to be lower than that under the balanced traffic load. The reason of the lower throughput of the VBR2 traffic is clear. The amount of the VBR2 traffic under the unbalanced traffic condition with specified traffic ratio is quite smaller than that under the balanced traffic condition. The lower VBR2 traffic load results in the lower throughput of the VBR2 traffic. On the other hand, the overall throughput under the unbalanced traffic load has not decreased as much as that of VBR2 traffic because the network has allocated a portion of its bandwidth to serve the VBR1 traffic as it has increased its amount under the unbalanced traffic load. Due to the nature of the CNT subprotocol, the overall throughput under the unbalanced traffic load cannot be kept as high as that under the balanced traffic condition. This fact shows that under the unbalanced traffic load, the performance of the proposed protocol will only suffer a bit in terms of throughput. However, this degradation on the overall throughput is not serious, especially, when the total traffic load is not heavy.

## VI. Conclusion

This paper introduced a new approach that combines different types of MAC protocols into a single WDM network to better serve a wide variety of multimedia applications. Some of the goals of this approach are: 1) to keep the advantages of the individual MAC protocols with respect to specific types of traffic streams; 2) to efficiently support a large range of traffic streams with different characteristics and QoS requirements in a single WDM network and for a transmitter to access the channel; and 3) to dynamically allocate the network bandwidth to the different classes of traffic in order to boost the network performance. We have investigated the performance of our M-WDMA network, and it was clearly shown that it outperforms state-of-art MAC protocols for the WDM networks in serving multimedia applications. As a result, we can reasonably expect the integration of various subprotocols into a single protocol to have a good potential for meeting the QoS requirements of future integrated services WDM networks.

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