Multimedia-MAC Protocol: Its Performance Analysis and Applications for WDM Networks

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***Abstract*—The design of the *medium-access control* (MAC) protocol is the most crucial aspect for high-speed and high-performance local and metropolitan area networks, since the decisions made at this level determine the major functional characteristics of these networks. Most of the MAC protocols proposed in the literature are *not* suitable for multimedia applications, since they have been designed with one *generic* traffic type in mind. As a result, they perform quite well for the traffic types they have been designed for, but poorly for other traffic streams with different characteristics. In this paper, we propose an integrated MAC protocol called the *Multimedia-MAC* (M-MAC), which integrates different MAC protocols into a hybrid protocol in a shared-medium network to efficiently accommodate various types of multimedia traffic streams with different characteristics and quality-of-service demands, namely, a constant-bit-rate traffic, bursty traffic (say, variable-bit-rate traffic), and emergency messages (say, control messages). We have developed a mathematical framework for the analysis and performance evaluation of our M-MAC protocol, which involves a queueing system with *vacation*. We have applied our M-MAC design approach to a wavelength-division multiplexing network, and evaluated its performance under various traffic conditions.**

***Index Terms*—Performance evaluation, quality-of-service (QoS) guarantees, medium-access control (MAC) protocols, wavelength-division multiplexing (WDM) networks.**

I. Introduction

FUTURE-generation local and metropolitan area networks (LANs/MANs) will be required to provide a wide variety of services requiring different bandwidth (BW) and delay characteristics. The low-speed and non-quality-of-service (QoS)-oriented services could be handled by evolutionary versions of the conventional networks. However, the high-speed and QoS-oriented services require a new generation of LANs and MANs. Since the performance of LANs/MANs greatly depends on how the hosts access the shared medium, the design of medium-access control (MAC) protocols is the most challenging part. A plethora of MAC protocols have been proposed for wireline LANs/MANs: metaring [1]; fiber-distributed data interface (FDDI [2]; and cyclic-reservation multiple access (CRMA) [3], to name a few [4]. In addition, a large number of MAC protocols have been proposed for wireless LANs/MANs, as well [5], [6].

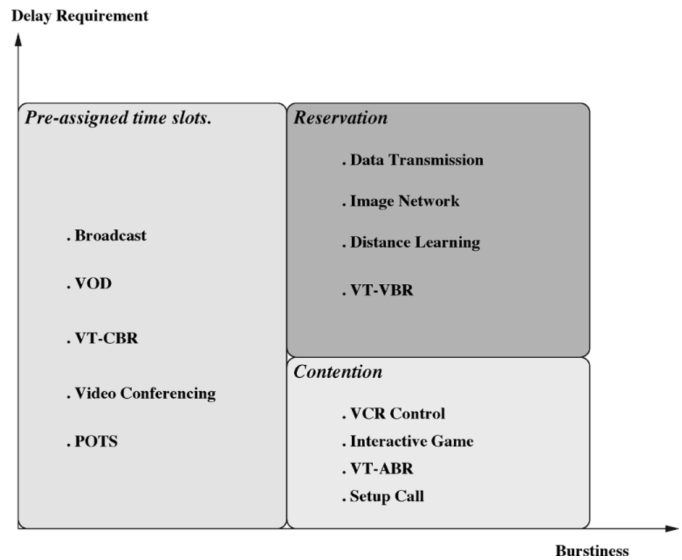

Fig. 1. Multiple traffic-stream features and their appropriate MAC protocols.

The objective of this paper is to propose a new hybrid protocol which best serves the various types of traffic with widely varying characteristics and evaluate its performance. The proposed MAC protocol is called the Multimedia-MAC (M-MAC) protocol. We also develop an analytical framework for the performance evaluation of our MAC protocol under different traffic and networking environments.

This paper is organized as follows. Section II gives an overview of MAC protocols. Section III introduces our M-MAC protocol. In Section IV, we derive an analytical model for the performance evaluation of our protocol. We present an example application of our M-MAC protocol for wavelength-division multiplexing (WDM) networks in Section V. Finally, Section VI concludes the paper.

II. Overview of MAC Protocols

One can classify the multimedia traffic streams depending on their data burstiness and delay requirements, as shown in Fig. 1. Video/audio streams and plain old telephone service (POTS) have small data burstiness, but require almost constant transmission delay and almost fixed BW in order to guarantee their QoS. On the other hand, applications such as image networking

Paper approved by T. T. Lee, the Editor for Wireless Communication Theory of the IEEE Communications Society. Manuscript received August 30, 2002; revised July 15, 2004. This work was supported in part by the Hong Kong Research Grant Council under Grant HKUST6160/03E.

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Digital Object Identifier 10.1109/TCOMM.2006.869802

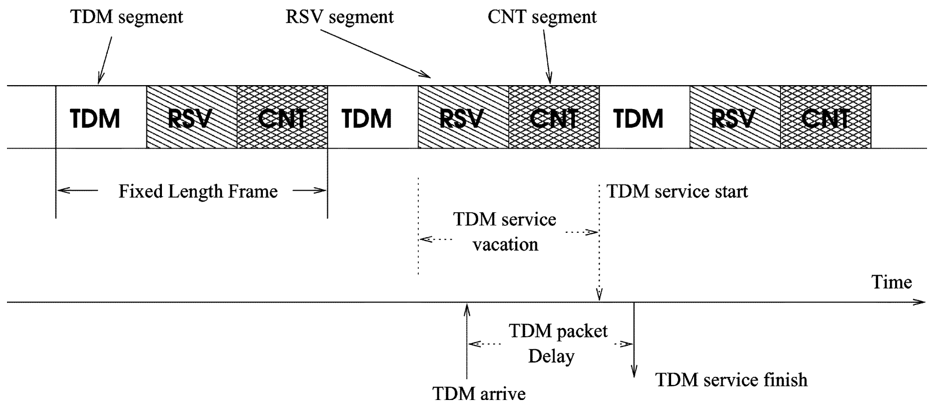

Fig. 2. M-MAC protocol construction.

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 application includes control messages for video-on-demand systems or interactive games, and network control and management signaling. 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 allocation-based MAC protocols, since they can guarantee that each node has a fixed available BW. The best MAC protocol for this purpose would be a simple round-robin time-division multiplexing-access (TDMA) scheme. On the other hand, reservation-based MAC protocols are very well suited for applications where the traffic streams are bursty (i.e., variable-bit-rate (VBR) traffic) or the traffic load of the nodes is unbalanced, since reservation-based MAC protocols schedule the transmission according to a particular transmission request. Finally, random-access (contention) MAC protocols have the potential of meeting the delay requirements of very urgent messages since their access delay is relatively small (when the load is reasonably low). Some examples of these urgent messages (e.g., call setup) are listed in Fig. 1. Although these applications do not generate a large amount of traffic data when compared with the other applications, they require very low delay.

MAC protocols have been the subject of rigorous research over the past two decades. Conceptually, we can classify the MAC protocols proposed in the literature into three categories: preallocation-access protocols, reservation protocols, and random-access protocols [4].

- *Preallocation-based protocols*: Nodes access the shared medium in a predetermined way. A backlogged node (a node having packets to send) is allocated one or more slots within a frame [7]–[9].
- *Reservation-based protocols*: A backlogged node has to reserve one or more time slots within a frame before the actual packet transmission can take place. The reservation is typically done using dedicated control slots within a frame or a separate reservation channel [10]–[16].
- *Random-access protocols*: The nodes access the shared medium with no coordination among themselves. Thus, when more than one packet is transmitted at the same time slot, collision occurs, and all the transmitted packets are lost. The collision-resolution mechanism dictates the performance of these protocols (and thus, is an integral part of these protocols) [17], [18].

In [19], a collision-free MAC protocol for an all-optical slotted packet network based on WDM multichannel ring topologies is reported, where nodes are equipped with one fixed wavelength receiver and one wavelength tunable transmitter. A novel reservation-based MAC protocol for passive WDM is reported in [20]. In all of the above proposed MAC protocols, it has been implicitly assumed that the traffic is of one *generic* type. On the contrary, the real-world multimedia traffic exhibits different characteristics and requires different QoS demands, namely, CBR traffic, bursty traffic (say, VBR traffic) and emergency messages (say, control messages). Hence, the above protocols are *not* suitable for multimedia applications.

In light of the above, we propose an efficient access scheme for shared-medium networks that:

1) integrates different types of MAC protocols into a single MAC protocol;
2) efficiently supports different types of traffic;
3) can be widely applied to various kinds of shared-medium networks.

We call this protocol the M-MAC.

III. The M-MAC Protocol

The M-MAC protocol consists of three subprotocols: namely, preallocation, wherein nodes access the shared medium in a fixed predetermined way, and the scheme is denoted by TDM; reservation, wherein the nodes reserve one or more time slots within a frame before the actual packet transmission starts, denoted by RSV; and contention, wherein the nodes access the shared medium with no coordination between them, denoted by CNT, each of which serves a certain type of traffic (see Fig. 2). These three subprotocols are active, one at a time, in a particular time slot (within a frame) which is dictated by a time-division multiplexing (TDM) scheme. Whenever a subprotocol uses the medium, the medium access is controlled according to the discipline of that subprotocol. We assume that the packets are of fixed size (L bits), and it takes $T_{\rm slot}$ s for the communication channel (of BW B b/s) to transmit. We call this time $T_{\rm slot}$ the *slot time*, and measure other quantities in terms of this quantity. A cycle in a M-MAC consists of a fixed time frame (of

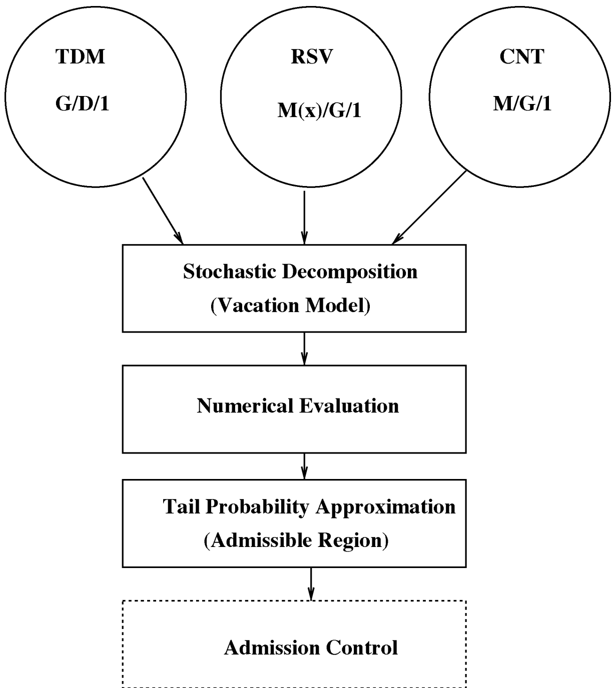

Fig. 3. Our approach to model and analyze the M-MAC protocol.

length L_{frame} slots), which consists of three segments, namely, a TDM segment (of length L_{TDM} slots), an RSV segment (of length L_{RSV} slots), and a CNT segment (of length L_{CNT} slots), in that order (see Fig. 2). In this paper, we assume that the various subprotocol segment lengths (L_{TDM}, R_{RSV}, and R_{CNT}) are fixed. Using the above notation, the frame length is given by $T_{\text{frame}} = L_{\text{frame}} \times T_{\text{slot}}$ s.

IV. Analytical Modeling of the M-MAC Protocols

As noted above, in the M-MAC protocol, the packets belonging to different traffic streams are served by different subprotocols, which control the access of the shared medium during different segments in the fixed time frame (L_{frame} slots). This means that the packets belonging to a particular traffic stream (to be served by the corresponding subprotocol), has to wait for its next turn when the current slot is exhausted. The medium is accessed by some other subprotocols at this interval, and is analogous to a queueing system with vacation (server unavailable for certain time). This justifies our approach to model the queueing situation encountered by packets in the M-MAC protocol using three independent queues with vacation, each corresponding to the TDM, RSV, and CNT subprotocols (see Fig. 3). Here, we successfully apply an approach which uses the decomposition properties of a general queueing system with vacation (explained below) to compute the moments of the waiting-time distribution from its Laplace–Stieltjes transform (LST). Also, an approximation for the tail probability from the moments is computed.

In our analysis of M-MAC, we are particularly interested in its performance with respect to the QoS metrics, namely, the deadline-missing rate (DMR) and the mean delay of packets. DMR is defined as the probability that the waiting time of an arbitrary packet exceeds a given deadline. The DMR is useful in finding whether a given packet has missed its transmission deadline or not. We first consider the waiting-time distribution of packets served by the different subprotocols. Then, we employ a numerical method to evaluate the distribution and obtain its moments. From the computed moments, we approximately determine the DMR of packets served by the subprotocols.

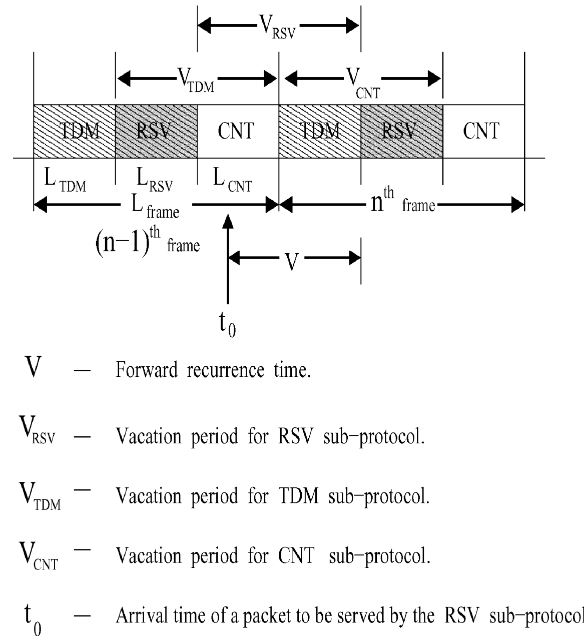

Fig. 4. Vacation period in a M-MAC protocol.

A. *Queueing Models With Vacation for Modeling the M-MAC Protocol*

In this subsection, we study different queueing models with vacation [21]–[23]. The reader is referred to [24] for the corresponding queueing models *without* vacation.

The *total* delay time of a packet (in a queue with vacation) from its arrival to its departure consists of three components: the queueing time; the vacation time of the server; and the service time. Intuitively, one can see that the waiting time (queueing time) of a packet in a queue with vacation is the sum of two random variables (RVs): the waiting time in the queueing system without vacation and *forward recurrence time* in the vacation (see Fig. 4). The forward recurrence time is a random time interval between the packet arrival instant (t_0) during the vacation and the instant of the next consecutive return of the server to the service (or the instant of the termination of current vacation period). The forward recurrence time is denoted by V in Fig. 4. Since these two RVs, viz., the waiting time in the queueing system without vacation and forward recurrence time in the vacation, are independent, the LST of the waiting-time density function is given by the product of the LST of these RVs. The above result is more formally called the *decomposition property* [21]–[23] of the waiting time in a general GI/G/1 queue with vacation. This result is given below.

Let $V^*(s)$ be the LST of the distribution of forward recurrence time V. Use W_d to denote the total delay time (waiting time plus the service time) with vacation, and $W^*(s)$ to denote the LST of the corresponding distribution. Also use $W_1^*(s)$ to denote the LST of the distribution of W_1 (which denotes the waiting time in an analogous queue *without* vacation). We use $B^*(s)$ to denote the LST of the distribution of the service time RV $B(t)$. Then

$$\mathbf{W^*(s) = W_1^*(s)B^*(s)V^*(s)}. \tag{1}$$

We use this as the key relationship to derive the LST of the distribution for the waiting time of packets served by different subprotocols in our M-MAC protocol.

1) TDM Subprotocol Model: In the TDM subprotocol, the transmission of message is controlled by a preallocated scheduling table. Note that in a TDM, the BW allocated to each user is fixed, and thus, *complete partitioning* of the link is accomplished. Now, corresponding to each connection/service or destination, packets from the local node are stuffed into the TDM slots. But since the TDM subprotocol is active only during part of the whole frame, these packets are buffered when they arrive. As and when the turn for the TDM subprotocol occurs (during the $L_{\rm TDM}$ segment), the packets buffered are transmitted. We model this situation by the queue with vacation, with the service times being *deterministic*, since the packet size is constant. Since we assume that the packet is of fixed length (of L b), transmission is also fixed and equals $T_{\rm slot} = L/B$ s. Hence, the service time $B(t)$ is better modeled as *deterministic*. Thus, the corresponding LST is given by

$$B^*(s) = e^{-\frac{s}{\mu}} \tag{2}$$

where μ is the service rate in packets/s. Because of the fixed transmission cycle of the TDM subprotocol, the vacation period is also fixed. We assume that the arrival process is deterministic (CBR nature) with mean $\lambda_{\rm TDM}$ arrivals/s. Given that an arrival occurs during vacation, the arrival instant of that (tagged) packet is uniformly distributed during the vacation period $V_{\rm TDM}$. Hence, the forward recurrence time is given by

$$V^*(s) = \frac{1 - e^{-\lambda_{\rm TDM} V_{\rm TDM} s}}{s\lambda_{\rm TDM} V_{\rm TDM}} \tag{3}$$

where $V_{\rm TDM} = T_{\rm FRAME} - T_{\rm TDM}$ is the vacation period. Using the decomposition property and noting that the waiting time in a D/D/1 queue is zero, we have for the LST of the distribution for total delay

$$W_{\rm TDM}^*(s) = \frac{(1 - e^{-\lambda_{\rm TDM} V_{\rm TDM} s}) e^{-\frac{s}{\mu}}}{s V_{\rm TDM} \lambda_{\rm TDM}}. \tag{4}$$

2) RSV Subprotocol Model: In the RSV subprotocol model, a node keeps on reserving slots in the RSV segment $(L_{\rm RSV})$ in the next frame until its backlogged packets are served. Until then, it holds the token. (See the algorithm that follows.) Thus, a node may get more than one RSV segment for transmitting *all* the backlogged packets. Hence, one RSV service cycle consists of the sum of service times of packets belonging to all backlogged nodes. In this case, the distribution of the vacation time is quite complicated to obtain [23]. We adopt the following approach. For the purpose of analytical simplicity, we assume that the service times (at a node) are independent and exponentially distributed. Then, the summation of the service time for all nodes is Erlang-k distributed [23], where k is the number of active (backlogged) nodes. Assume a symmetric system where the traffic load is identical on each node, and a fair system where each node gets identical service. Then, the probability that a node gets a transmission chance in a network with N nodes is $p = 1/N$. Suppose the mean service time a node receives is T_r, then the LST of the distribution for vacation time for the queue at each node is given by

$$V^*(s) = \left(\frac{N}{N + sT_r}\right)^N. \tag{5}$$

The RSV subprotocol is designed for bursty traffic transmission. Hence, it is justified that the packet-arrival process is modeled as a Poisson bulk-arrival process. Given the following for the $M^{(x)}/G/1$ queue, arrival follows a Poisson process with arrival rate λ, the bulk size is geometrically distributed $G(.)$ with mean g, and $B^*(s)$ being the LST of the service distribution, we know that for the above $M^{(x)}/G/1$ queue *without vacation*, the LST of the distribution of the waiting time W_1 holds [24]

$$W_1^*(s) = \frac{(1-\rho)}{s - \lambda + \lambda G[B^*(s)]} \frac{(1 - G[B^*(s)])}{g[1 - B^*(s)]} \tag{6}$$

where $\rho = \lambda g/\mu$.

In order to get the total delay of packets in a similar queue but *with vacation*, we use the decomposition property. Thus, we get the LST of the total packet delay with arrival rate $\lambda_{\rm RSV}$ of the RSV subprotocol as follows:

$$\begin{aligned} W_{\rm RSV}^*(s) = &\frac{(1-\rho)}{(s - \lambda_{\rm RSV} + \lambda_{\rm RSV} G[B^*(s)])} \\ &\times \frac{s(1 - G[B^*(s)])}{(g[1 - B^*(s)])} \left(\frac{N}{N + sT_r}\right)^N B^*(s). \end{aligned} \tag{7}$$

3) CNT Subprotocol Model: The CNT subprotocol is a random-access protocol. It best serves applications which produce messages of relatively small size but which require lower delay.

The system assumed in this paper for the CNT subprotocol is a slotted ALOHA system with finite nodes (finite user population) N, and finite buffer size of L packets. For $L \to \infty$, the case of infinite buffer capacity is obtained. Following are the assumptions.

1) The CNT segment within the fixed time frame of the M-MAC protocol is divided into slots of length $T_{\rm slot}$ s. (Note, this is the transmission time of a packet).
2) The defer first transmission (DFT) principle is employed [25]. With DFT, all packets are transmitted with a given permission probability in each slot (packets already waiting in the queue are *not* discriminated against by a fresh arriving packet). The retransmission probability for the deferred packet is p.
3) The channel is noise-free. A collision is the only reason for an unsuccessful transmission. No packet survives a collision.
4) All packets are of same length with one slot of transmission time. A station would know the transmission status (success or failure) of a packet immediately after it had finished the transmission. If the transmission is successful, the packet is removed immediately. If the

transmission fails, the packet is retransmitted in the next slot with retransmission probability p.

5) The arrival process of packets follows Bernoulli's process. That is, a packet arrives in a slot with probability λ and no packet with probability $1-\lambda$.
6) The service of packets in a queue of every user is on a first-come, first-served (FCFS) basis. The arriving packets finding the queue full, get dropped and do not return.
7) We also assume that all nodes are statistically identical in terms of arrival rate and service time.

With the above assumptions, it is clear that the queue of packets in the buffer in the CNT subprotocol is modeled as a Geo/Geo/1/K queue with deterministic vacation. For analysis, we use the tagged-user approach (TUA) [25]. Note that in the contention protocols, a packet service time in a user queue depends on the behavior of all other users (user queues) in the system. The influence of a user on the channel depends on its busy probability p_b,[1] and its retransmission probability p. ps is the probability that a user transmits a packet successfully in a slot, given that it makes a transmission in that slot. In [25], it has been shown that the probability generating function (PGF) $B(z)$ of the packet service time is given by

$$B(z) = \frac{pp_s z}{1-(1-pp_s)z} \tag{8}$$

where

$$pp_s = p(1-p_b p)^{N-1}. \tag{9}$$

Equation (8) is, in fact, the PGF of geometrical distribution with parameter pp_s. Note also that the mean service rate (μ) is given by $\mu = pp_s$. Also, the state probabilities $p_i, i = 1, 2, \ldots, K-1$ are given by

$$p_i = \left[\frac{\lambda(1-\mu)}{\mu(1-\lambda)}\right]^{i-1} \frac{\lambda}{\mu(1-\lambda)} p_0, \quad 1 \le i \le K-1 \tag{10}$$

$$p_K = \frac{\lambda}{\mu}\left[\frac{\lambda(1-\mu)}{\mu(1-\lambda)}\right]^{K-1} p_0. \tag{11}$$

Using the total probability law, we find

$$p_0 = \left(\frac{\mu}{\mu-\lambda} - \frac{\lambda^2}{\mu(\mu-\lambda)}\left[\frac{\lambda(1-\mu)}{\mu(1-\lambda)}\right]^{K-1}\right)^{-1}. \tag{12}$$

In order to solve for the two unknown variables μ and p_0, from (9) and (12), a numerical algorithm proposed in [25] is used here. Now the response time or total delay $(\mathbf{D})$, defined as the delay (in terms of discrete time slots) experienced by packets from the arrival instant till the time they depart. The PGF of $\mathbf{D}$ is given by

$$R(z) = \frac{p_0}{1-p_L}\left[1 + \left(\frac{\lambda}{1-\lambda}\right)\left(\frac{1}{1-(1-\mu)z}\right) \times \left(\frac{1-[xB(z)]^{K-1}}{1-xB(z)}\right)\right] B(z) \tag{13}$$

[1]A user is said to be idle if its queue is empty, otherwise, it is busy.

where $x = \lambda(1-\mu)/\mu(1-\lambda)$. Using Little's law, the average response time is, therefore, given by

$$E(\mathbf{D}) = \frac{E(\mathbf{Q_1})}{\lambda(1-p_b)} \tag{14}$$

where $E(\mathbf{Q_1})$ is the mean queue length. By decomposition property (1), the Z-transform of the delay distribution of packets in a CNT subprotocol is

$$D^*_{\mathrm{CNT}}(z) = \frac{R(z)V_{\mathrm{CNT}}}{1-z^{-1}} \tag{15}$$

where $V_{\mathrm{CNT}} = L_{\mathrm{frame}} - (L_{\mathrm{TDM}} + L_{\mathrm{RSV}})$ is the vacation time for the CNT queue.

B. Numerical Evaluation of the Waiting-Time Distribution

In the previous section, we derived the LST of the waiting time of packets belonging to a traffic type served by different subprotocols within the framework of the M-MAC. However, it is often very difficult, or even impossible, to analytically invert the LST of a continuous probability distribution or the Z-transform of a discrete probability distribution [26]. A Fourier-series method which can numerically invert the Laplace transforms and generating functions is discussed in [26]–[28]. Here we give the gist of the method [29] which computes the nth moment of a given continuous (discrete) RV from the LST (PGF) of its distribution.

To begin with we give some definitions. For a nonnegative discrete RV X, let $p_k = PrX = k, k = 0, 1, \ldots$, then its PGF is defined by

$$G(z) = E[z^X] = \sum_{k=0}^{\infty} p_k z^k |z| < 1. \tag{16}$$

When X is a continuous variable, the cumulative distribution function of X is denoted by $F(x)$, then the LST of $F(x)$ is given by

$$F^*(s) = E[e^{-sX}] = \int_0^\infty e^{-sx} dF(x), \quad \text{for } Re(s) \ge 0. \tag{17}$$

The moment generating function (MGF) of a continuous RV X is a function $M : R \to [0, \infty)$ given by $M(t) = E[e^{tX}]$. Let μ_n represent the nth moment of X, then, $\mu_n = M^{(n)}(0)$, using the notation that $M^{(n)}(a)$ to denote nth differentiation of the function $M(x)$ with respect to x and evaluating its value at a. Note that

$$M(z) = \sum_{n-0}^{\infty} \frac{\mu_n}{n!} z^n. \tag{18}$$

Recall the definition of Z-transform of a sequence w_n,

$$W(z) = \sum_{n=0}^{\infty} \omega_n z^n. \tag{19}$$

Note that $M(z) = W(z)$ with $w_n = \mu_n/n!$ from (18). Also, note that

$$W(z) = M(z) = \begin{cases} F^*(-z), & \text{continuous case} \\ G(z), & \text{discrete case.} \end{cases} \tag{20}$$

From the above equation, it is clear that it is enough if we develop an algorithm to invert the Z-transform. For the inversion of the Z-transform, we use the lattice-Poisson algorithm [29]. The idea is to use the Cauchy integral formula for the sequence w_n and compute this integral numerically using the m-point trapezoidal rule. To avoid large discretization errors due to either too fast or too slow an increase in $\mu_n/n!$, an adaptively modified MGF is inverted to get the nth moment of the given MGF (PGF). This involves the inclusion of the *adaptive decay rate* factor α_n in the argument of MGF to define the modified MGF. If we are given the MGF $M(z)$, in order to compute μ_n, form

$$W_n(z) = M(\alpha_n z) = \sum_{k=0}^{\infty} \alpha_n^k \frac{\mu_k}{k!} z^k \quad (21)$$

where

$$\alpha_n = (n-1)\frac{\mu_{n-2}}{\mu_{n-1}} n \geq 3. \quad (22)$$

For α_n $n = 1, 2$, we use the following procedure. We arbitrarily set $\alpha_i = 1$ and compute μ_1. Next, using μ_0 (which is 1 by definition) and μ_1, and using (22), we compute α_2. We give only the final expression for the moments and the discretization error as

$$\begin{aligned} \mu_n &= \frac{n!\omega_{nn}}{\alpha_n^n} \\ &= \frac{n!}{2nlr_n^n\alpha_n^n}\Bigg\{W_n(r_n) + (-1)^n W_n(-r_n) \\ &\qquad + 2\sum_{j=1}^{nl-1} Re\left(W_n\left(r_n e^{\frac{\pi i j}{nl}}\right) e^{-\frac{\pi i j}{l}}\right)\Bigg\} - \bar{e} \end{aligned} \quad (23)$$

where

$\bar{e}$ $\sum_{j=1}^{\infty} \alpha_n^{2ljn}(n!/(n+2ljn)!)\mu_{n+2ljn}10^{-\gamma j}$;

r_n radius of the contour;

l is some integer to control the round-off error.

For more details of its derivation the reader is referred to [29], where it has been shown that the method is reasonably accurate.

C. DMR: The Tail of the Waiting-Time Distribution

DMR is the probability $F_{\mathrm{DMR}}(D)$ that the waiting time of an arbitrary packet served by a subprotocol exceeds a given deadline (D), i.e., $F_{\mathrm{DMR}}(D) = Pr\{W > D\}$. The DMR is useful in characterizing the QoS of a given traffic. (Note that $F_{\mathrm{DMR}}(D)$ is the tail of the waiting-time distribution). However, given an arbitrary deadline (D), the computation of the waiting-time distribution is quite difficult, because the number of moments required to produce an adequately accurate result is unknown, although we can get quite a number of moments with reasonable accuracy by the method we discussed in the previous subsection. Therefore, we need to find an approximation to obtain the tail probability. In view of this, we draw attention to the approximation [29] which is given below

$$F_{\mathrm{DMR}}(x) \approx Ae^{-\eta x} \quad (24)$$

where $\eta = \lim_{n\to\infty} \eta_n$, $\eta_n = n\mu_n - 1/\mu_n$, $A = \lim_{n\to\infty} A_n$, and $A_n = \eta_n^n \mu_n/n!$. With the above equations, we can obtain the DMR of an arbitrarily given deadline for waiting time.

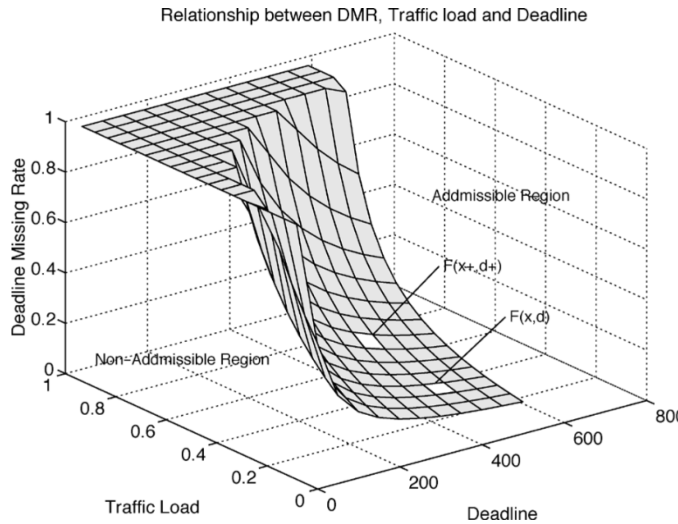

Fig. 5. Admissible region as a function of DMR, deadline, and traffic load.

D. Admissible Region

The DMR (for a particular type of traffic) depends on the given deadline and the waiting-time distribution. The waiting-time distribution of packets, in turn, depends on the parameters of the queueing system modeled, namely, the arrival, service, and vacation time distributions. Note here that the performance of a queue in the "generic node" is dependent on the queues in the other nodes, although we made an assumption that this queue is studied *independently*.[2] Hence, the quantity DMR gives an idea about the QoS offered by the network to the incoming calls. Thus, the DMR can be used for the call-admission control. Given the $F_{\mathrm{DMR}}^{t}(D)$, the DMR at time t (with deadline D) when a new call arrives (with DMR demand Ψ and other parameters like its arrival rate, etc.), we can estimate $F_{\mathrm{DMR}}^{t+}(D+)$, the DMR at the next time instant $t+$ (with required new deadline $D+$). If $F_{\mathrm{DMR}}^{t+}(D+) < \Psi$, then the traffic is admitted, or else it is rejected. In our case, the DMR of aggregate traffic at any time for the three subprotocols (the resulting queues modeled by individual sets of arrivals, service, and vacation disciplines) is computed. Therefore, the relation between the DMR, the deadline, and the traffic load fairly gives an idea about the network status. This is plotted in Fig. 5 using the previously described analytical methods. The space shown in the figure is divided by a surface. At any point on the surface, given the deadline and traffic load, if the required DMR is above the (computed) DMR corresponding to this point (for the same deadline), the traffic can be admitted. Hence, the region (called the *admissible region*) which is above the surface corresponds to those points where the traffic can be admitted.

To summarize, given the traffic type (whether it is served by any of the three subprotocols), arrival, service, and vacation distributions, and deadline (D), the above-mentioned surface can be computed using this surface, and the call-admission control is used to decide whether a call can be admitted or not. The advantage of this method is that the decision is instantaneous, rather than using time-consuming methods based on estimation

[2]This is clear, particularly in view of the fact that the performance of contention and reservation protocols are very much dependent on the queues in the other nodes.

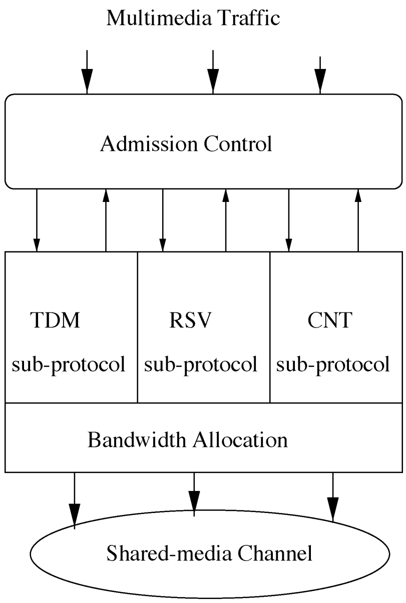

Fig. 6. Admission control and M-MAC.

of long-term statistics (which are done conventionally). This idea is illustrated in Fig. 6.

V. Application of M-MAC on WDM Networks

The M-MAC design presented in the previous section can be applied to a wide variety of shared-medium networks [30], [31]. Here, we apply the M-MAC to WDM networks. Based on the idea of our proposed M-MAC protocol, together with the consideration of the physical transmission characteristics of WDM networks, we propose a MAC protocol for single-hop WDM networks, which combines the advantages of three types of MAC protocols within a single framework to well serve a wide range of traffic streams for multimedia applications. This MAC protocol is called Multimedia-WDMA (M-WDMA). M-WDMA is very similar to the Multimedia-MAC protocol described in previous section, except that the concept is applied to optical networks, with a slight change in the protocol to suit various physical characteristics and limitations of the optical networks, say, the nonzero tuning time of the transmitting laser.

A. The M-WDMA Architecture

We consider the broadcast and select a multiwavelength optical network in star topology with N nodes connected by a star coupler (see Fig. 7). It is a single-hop network in which all the inputs from the various nodes are combined in a passive star coupler, and the mixed optical information is broadcast to all destination nodes. As in the M-MAC protocol, in M-WDMA also, we have data and control channels. Each node has three tunable transmitters to transmit data, one fixed tuned transmitter for transmitting control information, one fixed tuned receiver for receiving control information, and one fixed tuned receiver to receive data in the *home* channel. The data transmitter in each node can tune to any wavelength in the set $\{\lambda_i | \, i = 1, 2, \ldots, C\}$. Nodes transmit and receive the control information in the control channel with wavelength λ_c. Each node receives the data in the home channel through a fixed tuned receiver with wavelength λ_i, for $i = 1, 2, \ldots, C$. If $N = C$, then the ith node is assigned with the ith wavelength, and the collision is avoided. In case the number of available data 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 the packets carried in this channel.

An example of the above-described architecture for the case $N = C$ is illustrated in Fig. 8. The nodes in a WDM network can transmit data through the appropriate transmitter (depending on the data traffic type) at any time, and can transmit control information over the control channel. In Fig. 8, the X-axis denotes the timeslot, and the Y-axis denotes the transmitting node number. The white segments (columns) denote the TDM segments, the light-shaded segments are the RSV segments, and the dark segments are the CNT segments. The numbers in the TDM segments denote the wavelength in which the data is transmitted. Note that this transmitting wavelength (*home channel*) decides the destination node. For instance, in the considered example, in the second row and the fourth TDM slot, the number 6 denotes that the node 2 is transmitting a packet to node 6 in the channel number 6 (with wavelength λ_6). Similarly, in the same row, the fifth TDM slot, the data will be transmitted on channel number 7 (with wavelength λ_7). This means that the transmitter was transmitting at λ_6 in the fourth TDM slot and was in the process of tuning to λ_7 during the fourth RSV and CNT slots. Again, in the fifth TDM slot, the data will be transmitted on channel number 7 (with wavelength λ_7). The three data transmitters are used to serve the three different classes of traffic streams and operate in a *pipeline* fashion. That is, when one transmitter is transmitting a packet, the other transmitters are in the process of tuning to the wavelength to be used in their next turn. The tuning time (Γ) satisfies the following relationships: $L_{\rm TDM} + L_{\rm RSV} > \Gamma$, $L_{\rm TDM} + L_{\rm CNT} > \Gamma$, and $L_{\rm CNT} + L_{\rm RSV} > \Gamma$. It follows that $L_{\rm frame} > 3\Gamma/2$. In particular, the length of the frame directly affects the BW allocated to TDM segments. Hence, the total TDM BW is given by $B_{\rm TDM} = L_{\rm TDM} B / L_{\rm frame}(N-1)$, where B is the channel BW. By and large, all the channels would be in use in the TDM and CNT subprotocols if there are sufficient BW demands. (In the case of the CNT subprotocol, there are as many collision domains as there are channels C). But, in the case of RSV, the number of channels used in any frame depends on the result of the bipartite graph matching problem which is computed by the multiple token-rotation algorithm (discussed later) at the end of the previous frame.

Data Channel: The data transmission format in the M-MAC WDM network is illustrated in Fig. 9. Note that a data frame in M-WDMA has a similar structure as in M-MAC, except that within an interval of $T_{\rm frame}$ s, at most C number of channels transmit packets to a particular destination node. To see this, consider the TDM subprotocol. Here, multiplexing over time is *not* done within an interval of $L_{\rm TDM}$ slots. However, if you consider a cycle of $N-1$, such frames (of length $L_{\rm frame}$ slots), then one can see that the multiplexing over time is accomplished. A frame in a M-WDMA consists of a fixed time frame (of length $L_{\rm frame}$ slots) which consists of three segments, namely, a TDM segment (of length $L_{\rm TDM}$ slots), an RSV segment (of length

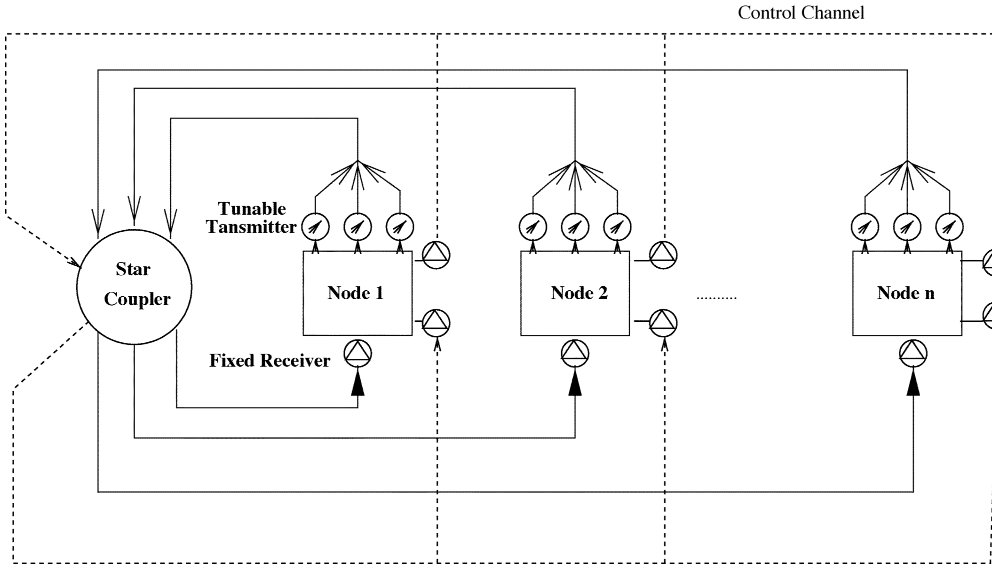

Fig. 7. M-WDMA network architecture.

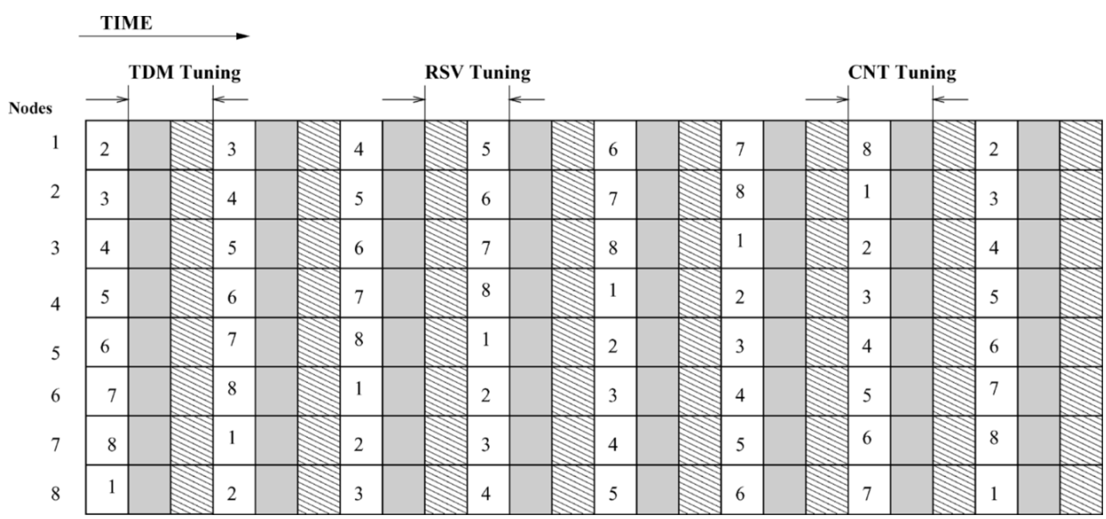

Fig. 8. M-WDMA transmission schedule example.

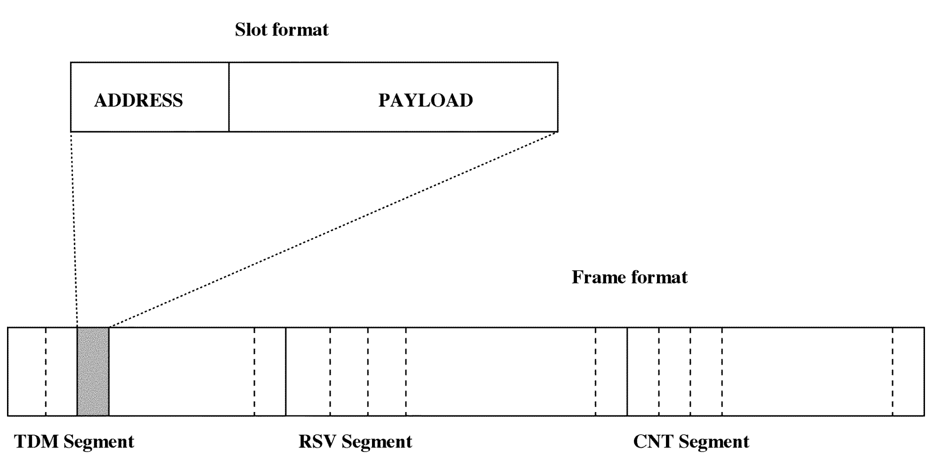

Fig. 9. M-WDMA frame and slot formats.

$L_{\rm RSV}$ slots), and a CNT segment (of length $L_{\rm CNT}$ slots), in that order. This feature is the main difference between M-MAC and M-WDMA. This feature of the multiwavelength nature increases the capacity, but puts a restriction on the performance of the devices; for example, the laser in the transponder requires nonzero tuning time (to tune from the wavelength which it is transmitting in the current frame to another wavelength in the corresponding slot in the next frame). This imposes the cyclic structure for the M-WDMA MAC frame. In the example given above, the cyclic structure has $7 \times L_{\text{frame}}$ slots. The total BW available for the TDM subprotocol is given by: $B_{\text{TDM}} = L_{\text{TDM}}B/L_{\text{frame}}$ b/s, where B b/s is the bit rate supported by the channel (a single wavelength). Similarly, the net BW available for the RSV and CNT subprotocols is given by $B_{\text{RSV}} = L_{\text{RSV}}B/L_{\text{frame}}$ b/s and $B_{\text{CNT}} = L_{\text{CNT}}B/L_{\text{frame}}$ b/s, respectively.

Control Channel Configuration: The control channel operates in a TDMA manner independent of the data transmission (and is coincident). One cycle consists of N *minislots*, each of which is designated to a node. Based on the broadcast information over the control channel (of cycle length T_{cntrl}), all the control procedures can be done by the nodes locally and network-wide synchronously. The control messages include: 1) reservation requests which are used by the reservation subprotocol; and 2) collision acknowledgments in the CNT subprotocol. For the reservation of the RSV protocol, we use a bit map to represent the reservation request, each node taking a bit (1 to denote that the node wants to transmit, and 0 to denote that there are no transmission requests). Similarly, we use the bit map (of length L_{cnt} slots corresponding to the number of slots in the data channel) to indicate the success (denoted by 1) or failure (denoted by 0) during the contention in a CNT subprotocol. Note that the length of a minislot is $N + L_{\text{cnt}}$. Hence, the cycle time $T_{\text{ctrl}} = N(N + L_{\text{cnt}})/R$, where R 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, $(N(N + L_{\text{cnt}})/L_{\text{slot}}) < L_{\text{frame}}$, where L_{slot} is the slot length in bits. We find that from the above inequality

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

This leads to a network scale limitation. For example, if we choose $L_{\text{cnt}} = 10$ slots, $L_{\text{frame}} = 30$ slots, $L_{\text{slot}} = 424$ b, then the number of nodes should be $N < 107$. By adding an exclusive receiver for the collision detection at each node, the inequality becomes $N < \sqrt{L_{\text{slot}}L_{\text{frame}}}$, which is 113 for the example above.

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. These three subprotocols in the M-WDMA MAC protocol operate independently.

1) The TDM Subprotocol: The operation of the TDM subprotocol within our M-WDMA network is basically an *interleaved TDMA* MAC protocol [32]. The only difference between our TDM subprotocol and TDMA is that in a M-WDMA network, we take tuning time into consideration. 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. Note that the TDM subprotocol does not need any control information to be transmitted over the control channel.

2) The RSV Subprotocol: In the RSV subprotocol, the time slots in all the C channels are reserved whenever there is a demand for the transmission of packets. But contention arises when more than one node wants to transmit packets to the same destination node (which is identified by a particular home channel at fixed wavelength). Hence, the problem is that in a given interval of $T_{\text{RSV}}(= L_{\text{RSV}} \times T_{\text{slot}})$ s, *which* node has to use which *wavelength*. This problem is similar to the bipartite graph matching problem. Here, we chose to use the *multiple-token rotation* algorithm [33] to resolve the contention. An advantage of using a token-based scheme in our RSV subprotocol is that it can efficiently support bursty traffic streams, and its implementation can be simple.

```
[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;
           }
}
```

Fig. 10. Token rotation algorithm.

In the multiple-token rotation algorithm, each channel (wavelength) is associated with a "token." A node can send its packets onto the destination channel only if it holds the corresponding token. 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 multiple-token rotation algorithm to determine the token distribution in the next frame. The multiple-token rotation algorithm is depicted in Fig. 10. The multiple-token rotation algorithm implicitly uses the round-robin scheme for scheduling a packet to be transmitted at a particular wavelength (channel). This ensures fairness among nodes. Also, this algorithm is executed at the beginning of the previous control slot, so that the computed mapping can be used to schedule the packets in the next data frame (by encoding the bits accordingly in the RSV segment of control frame).

The CNT Subprotocol: The CNT subprotocol of our M-WDMA MAC protocol is similar to the *interleaved slotted ALOHA* [32]. The active nodes compete for the slots in the current CNT segment in all the channels (wavelengths). In case there is a collision, the retransmission is scheduled after a random number of slots (in particular, the geometric retransmission attempts are assumed). We already noted that in an M-WDMA, each node has one fixed tuned receiver (called "home channel" for receiving data) and another fixed tuned receiver for receiving control information. Now, when more than one transmitter wants to transmit to the same destination node, *receiver collision* occurs. This collision is detected in the home channel, and the corresponding control slot (which corresponds to the particular channel in which the collision has occurred) in the next control frame carries this information. In the subsequent slots, the data channel again tries to transmit the same packet. Care has to be taken so that the retransmissions are carried out *within* the current CNT segment. Hence, 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 can be carried out across frames.

C. Modeling of the M-WDMA Network

This section investigates the performance of the M-WDMA MAC protocol analytically and through simulations. In a M-WDMA MAC protocol, three subprotocols operate independently, and one can think of these protocols operating in three different networks. These three virtual networks have BW equal to the BW allocated to that subprotocol in an M-WDMA network. This is reasonable under the assumption that the three classes of traffic are independent of each other (called the *protocol independent assumption*). With this assumption, the three subprotocols can be studied independently. One can think of a (transmitting) node (logically) consisting of $N-1$ queues, each corresponding to $N-1$ receivers (apart from itself). The transmitter polls one queue at a time to serve (transmit) a packet. Also, since a particular subprotocol is active in its own segment, these logical queues can be modeled using a queue with vacation.

We assume that all the nodes in the M-WDMA network are statistically identical, i.e., arrival and service process of packets have identical distributions (and thus, the system is *symmetric*). Let λ be the network normalized traffic load, then λ_{TDM}, λ_{RSV}, and λ_{CNT} are the mean traffic loads for the individual segments of the respective subprotocols. Then, $\lambda_i = (L_i/L_{\mathrm{frame}})\lambda$, where i can be TDM, RSV, or CNT.

We consider a system of N nodes and C channels. Logically, each node has $3C$ queues corresponding to C channels and the three types of traffic. We assume each queue has infinite capacity and uses the FCFS discipline. Let $B(x)$ be the distribution function for the service time, with $1/\mu$ being its mean and $B^*(s)$ being its LST. We denote the RV of vacation length (in terms of slot time) as V, its LST as $V^*(s)$ and its mean as $E[V]$. By applying the analytical results of suitable queueing models with vacation (as given in Section IV) for the queue encountered by the packets belonging to different traffic streams, we obtain the performance measure, namely, the mean delay of packets and the DMR.

1) The TDM Subprotocol Model: According to a TDM subprotocol, node i gets a chance to transmit L_{TDM} packets to node j in every $N-1$ frames. Hence, the vacation length in the (logical) queue (at node i) of packets destined to node j is given by

$$V_{\mathrm{TDM}} = (N-1)L_{\mathrm{frame}}T_{\mathrm{slot}} - L_{\mathrm{TDM}}T_{\mathrm{slot}} \qquad (25)$$

using the traffic independent assumption and assuming the statistical similarity of nodes.

As mentioned in a previous section, the packets waiting for transmission by the TDM subprotocol can be modeled as a queue with vacation. Having computed the value for V_{TDM}

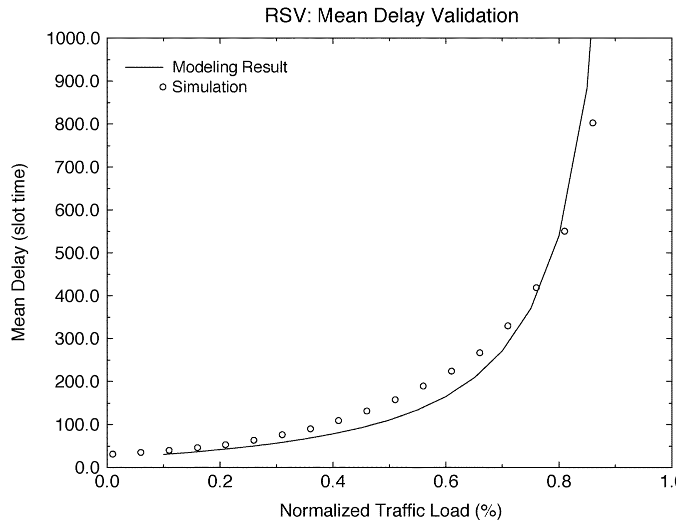

Fig. 11. TDM model validation in terms of mean delay.

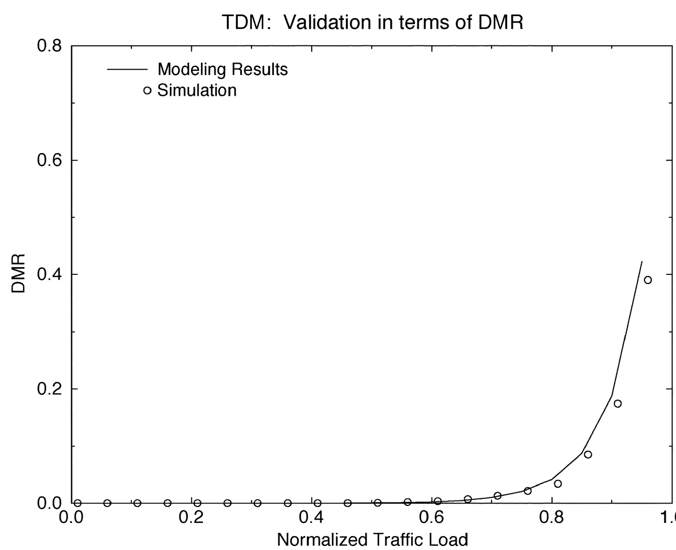

Fig. 12. TDM model validation in terms of DMR.

and noting that $\mu = B$ (where B is the channel bit rate in b/s), the distribution for the total delay experienced by the packets is given by (4). Further, the numerical evaluation method presented in subsection IV-B allows one to compute the mean delay, which is the performance measure studied here

$$E[W_{\mathrm{TDM}}] = W_{\mathrm{TDM}}^{*(1)}(0) = \mu_1. \qquad (26)$$

Fig. 11 presents the results corresponding to the mean delay of packets (served by the TDM subprotocol) computed by the analytical model and discrete-event simulation. As one can see from the figure, the results obtained by simulations agree very well with the analytical results, confirming the accuracy of our analytical model.

Computing the nth moment of the waiting-time distribution from (23) and using the approximation (24) given in Section IV, we compute the DMR. Here, we first use n^* moments for the estimation of DMR, where n^* is selected such that the error introduced in the estimation of DMR using first n^* moments and first n^*+1 moments is $<2\%$. Fig. 12 shows our analytical results for the TDM model and discrete-event simulation results corresponding to DMR. Although going through a quite complicated computation process, the results are quite close to each other, which implies good accuracy of our TDM model.

2) The RSV Subprotocol Model: In a M-WDMA, the RSV subprotocol is a token-based protocol. Once a node i gets a token corresponding to the destination (or channel) j, the RSV segment is reserved for it until all of its packets are transmitted. Hence, it might take more than one time frame. Whenever the queue corresponding to destination node j is in service, new arrivals are automatically added to the queue (backlogged packets) or immediately served when the queue is empty. For simplicity, we assume all the token rotations to be mutually independent and the rotation is of round-robin fashion. The vacation time for an output queue at a node i destined to node j is the sum of the frame times that this token (corresponding to destination node j) is withheld by the other nodes. The service time for a packet is deterministic (we ignore the packet length variations), and hence, the LST of the service time distribution $(B^*(s))$ is given by $B^*(s) = e^{-s/\mu}$. Note that the service time for a given message (composed of packets) depends on the bulk size (number of packets) of the message. Hence, the number of frames a node takes to transmit a message depends on the bulk size $(G(.))$ of the message. The message arrivals are Poisson-distributed with geometrically distributed $(G(.))$ message size. Hence, the buffer with packets waiting to be transmitted in the queue of the RSV subprotocol can be best modeled by the $M^{(x)}/G/1$ queue. Accordingly, using (6), one can obtain the LST of the total delay of a packet $(D^*_{\mathrm{RSV}}(s))$ in the RSV subprotocol model. (Note that the parameter T_r can be obtained from $1/\mu$ and the distribution of G.) Note that here, we assume bulk arrivals with a mean λ_{RSV}, in which the bulk arrivals are *geometrically* distributed with mean g. That is, the PGF of the bulk size $G(.)$ is given by

$$G(z) = \frac{1}{g(g+1-z)}. \tag{27}$$

Now, the queueing situation is similar to the case analyzed in subsection IV-A.2, and the mean waiting time and DMR can be obtained. The LST of the delay of a packet $(D^*_{\mathrm{RSV}}(s))$ in the RSV protocol model can be obtained using (6).

Computing nth moment of the waiting-time distribution from (23), and using the approximation (24) given in Section IV, we compute the DMR. As usual, we use the first n^* moments for the estimation of DMR, where n^* is selected such that the error introduced in the estimation of DMR using the first n^* moments and the first n^*+1 moments is $<2\%$.

Fig. 13 shows both the results from our simulations of CNT protocol and analytical results of our model corresponding to the performance measure of mean delay. The result shows that the proposed model is reasonably accurate.

Similar to the TDM model, we can calculate the DMR according to (24). Fig. 14 compares the analytical results with the our simulation results. The two results are close to each other, especially when the traffic load is light.

3) CNT Subprotocol Model: Since all the slots in a given CNT segment can be used by any node, the vacation time for a queue (corresponding to any node j) at node i is the sum of the TDM and RSV segments, i.e., $V = (L_{\mathrm{TDM}} + L_{\mathrm{RSV}})T_{\mathrm{slot}}$

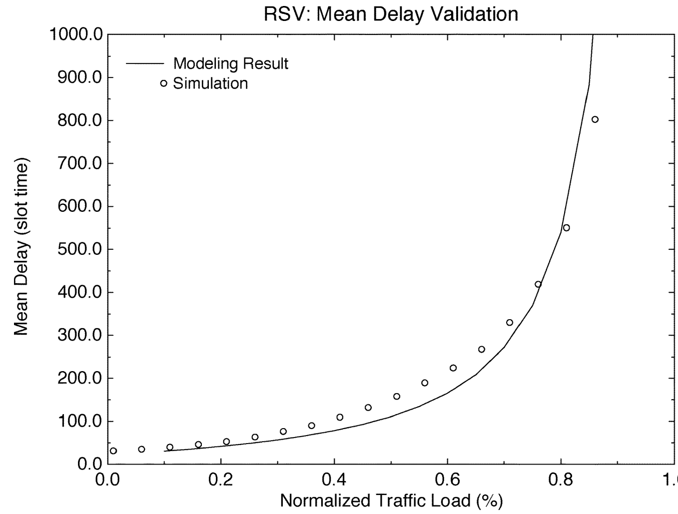

Fig. 13. RSV model validation in terms of mean delay.

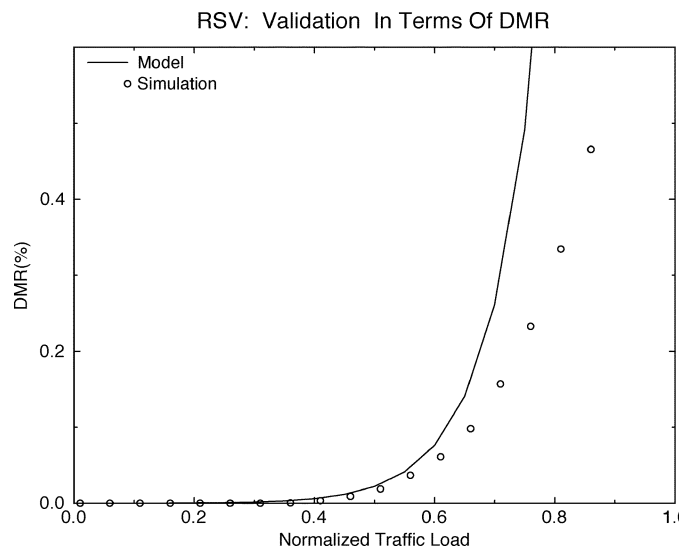

Fig. 14. RSV model validation in terms of DMR.

(unlike the queues in the RSV and TDM subprotocols). The expression for the delay in this case is obtained from (15), with $L_v = L_{\mathrm{TDM}} + L_{\mathrm{RSV}}$. By choosing the exact parameters for both modeling and simulation, we validate our analytical CNT model in Fig. 15 in terms of mean delay and Fig. 16 for DMR. As we can see, again the results are reasonably close to each other.

4) Admissible Region Comparison: As illustrated previously, the admissible region can be effectively used by an admission-control policy to decide whether to admit a traffic stream or not. By calculating the DMR of the TDM, RSV, and CNT subprotocols, for normalized traffic load and a range of required deadlines, we can obtain the corresponding admissible region, as shown in Figs. 17–19.

In Figs. 17 and 18, we show three types of surfaces, and the lowermost surface is generated from our analytical modeling results. The top-layer surfaces are an upper bound computed according to Chebyshev's inequality [34], which are used as modeling references (in fact, a simple upper bound). The

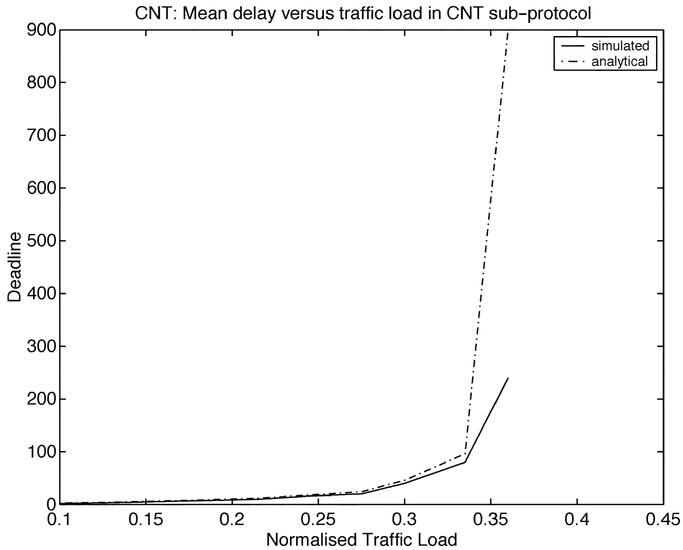

Fig. 15. CNT model validation in terms of mean delay.

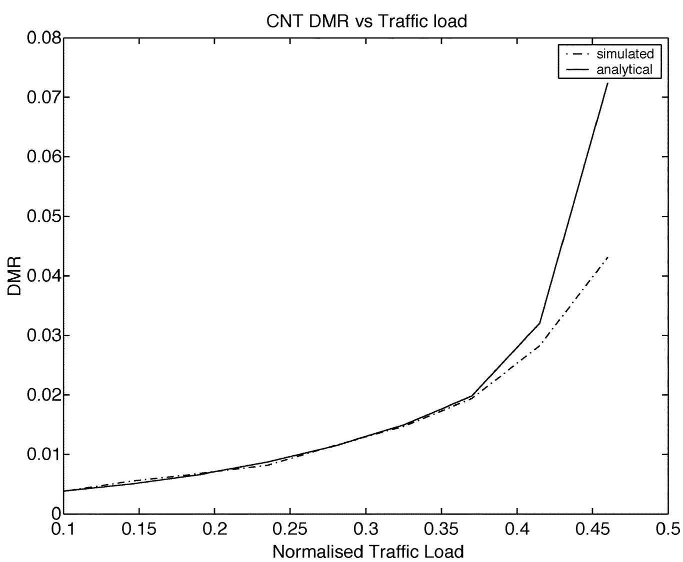

Fig. 16. CNT model validation in terms of DMR.

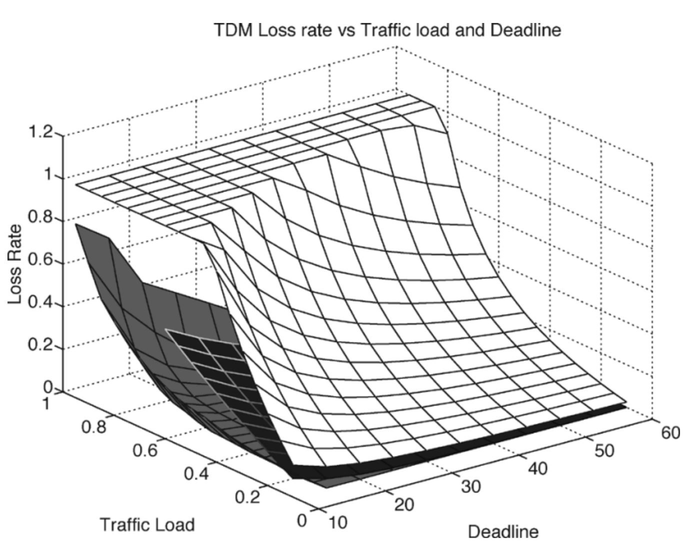

Fig. 17. Admissible region for CBR traffic.

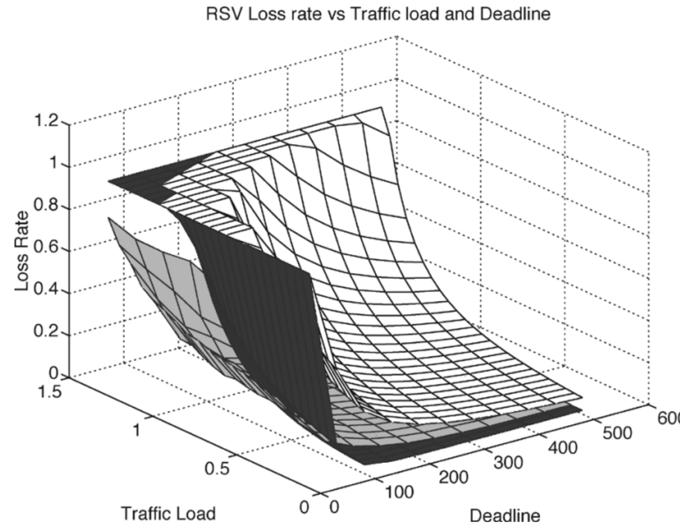

Fig. 18. Admissible region for RSV traffic.

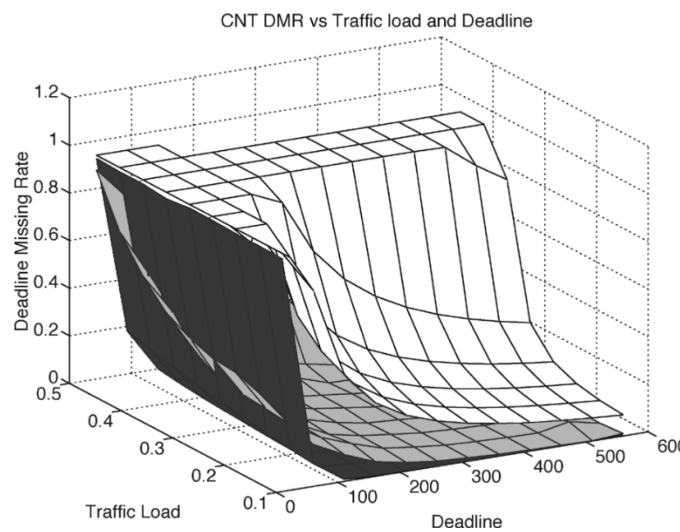

Fig. 19. Admissible region for CNT traffic.

middle-layer surface is obtained through intensive simulation. A similar comparison on CNT traffic is shown in Fig. 19.

By comparing these three figures, we can see the admissible region of a TDM subprotocol is quite flat. This is because in an M-WDMA, the TDM subprotocol has the highest priority for BW allocation. This results in the lower DMR (analytical) bound, even when the traffic load is high. For high-traffic loads, the (analytical) DMR is not very high in the case of the RSV subprotocol. This is the advantage of the RSV subprotocol, because the RSV subprotocol is primarily used for bursty traffic with relatively loose delay requirements. However, when the deadline becomes strict, the QoS of an RSV subprotocol is eventually worse than that in TDM, as is expected.

The CNT admissible region is valid only in very low-traffic areas. That is, under light traffic conditions, which is again expected from contention-based protocols. However, a very low DMR can be achieved even when the deadline is very strict. Thus, some urgent packets (e.g., for network control) can exploit this feature when the traffic load is low. As a general observation, one can also see that the simulation results are much closer to the analytical modeling results than that of the upper bound.

VI. Conclusion

This paper introduces a new methodology that combines different types of MAC protocols into a single shared-medium 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 shared-medium network; and 3) to be applied to a wide variety of shared-medium networks. We have also derived a detailed analytical model that can be used to determine the admissible region given various practical parameters (e.g., traffic load, type of traffic, DMR). This admissible region can readily be used by an admission-control policy to decide whether to admit an arriving stream or not while satisfying its QoS, and, at the same time, not altering the QoS of the already admitted streams.

We have illustrated the usage of our MAC protocol design and analytical model through a WDM network. We have shown that our general framework can readily be used for such networks. In addition, we have shown that our analytical results are reasonably accurate when compared with simulation. In particular, our analytical model can effectively be used with admission-control algorithms for providing QoS guarantees to multimedia applications.

We believe our framework can be applied to many other types of shared-medium networks, especially in the next generation of networks, where the integrated or multimedia services are provided in a single physical network.

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