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

Lixin Wang, Mounir Hamdi, R. Manivasakan and Danny H. K. Tsang

Abstract— **The design of the** *Medium Access Control* **(MAC) protocol is the most crucial aspect for high-speed shared medium Local and Metropolitan Area Networks (LANs/MANs) since the decisions made at this level determine the major functional characteristics of these networks. Most of the previously proposed MAC protocols are** *not* **suitable for multimedia applications since they have been designed and "optimized" 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 (herein termed as** *Multimedia Medium Access Control* **protocol (***Multimedia***-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 QoS demands, namely, a constant-bit-rate (CBR) traffic, bursty traffic (say, variable-bit-rate (VBR) traffic) and emergency messages (say, control messages). We have developed a mathematical framework for the analysis and performance evaluation of our** *Multimedia*-**MAC protocol which involves a queueing system with** *vacation***. We have applied our** *Multimedia***-MAC design approach to a** *wavelength division multiplexing* **(WDM) network and evaluated its performance under various traffic conditions.**

I. Introduction

Future generation Local and Metropolitan Area Networks (LANs/MANs) will be required to provide a wide variety of services requiring different bandwidth and delay characteristics. The low-speed and non-quality-of-service-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 depends on how the hosts access the shared medium, the design of MAC protocols is the most challenging part. A plethora of MAC protocols have been proposed for wireline LAN/MANs, such as Metaring [6], FDDI [11], CRMA [9] to name a few [14]. In addition, a large number of MAC protocols have been proposed for wireless LAN/MANs as well [10], [12].

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 termed as *Multimedia Medium Access Control* (*Multimedia*-MAC) protocol. We also develop an analytical framework for the performance evaluation of our MAC protocol under different traffic and networking environments.

Corresponding author: Prof. Mounir Hamdi email: hamdi@cs.ust.hk. Prof. Hamdi is with the Department of Computer Science, Hong Kong University of Science and Technology, Kowloon, Hong Kong.

Dr. R. Manivasakan and Prof. Danny H. K. Tsang are with the Department of Electrical and Electronic Engineering, Hong Kong University of Science and Technology, Kowloon, Hong Kong email: eemani@ee.ust.hk, eetsang@ust.hk.

This paper is organized as follows. Section 2 gives an overview of MAC protocols. Section 3 introduces our *Multimedia*-MAC protocol. In section 4, we derive an analytical model for the performance evaluation of our protocol. We present an example application of our Multimedia-MAC protocol for wavelength division multiplexing networks in Section 5. Finally, Section 6 concludes the paper.

II. Overview of MAC Protocols

MAC protocols have been the subject of rigorous research over the past two decades. Conceptually, we can classify the MAC protocols into three categories: preallocation access protocols, reservation protocols and random access protocols [14].

- *Preallocation-based protocols*: The nodes access the shared medium in a predetermined way. A node having packets to send (backlogged node) is allocated one or more slots within a frame.
- *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.
- *Random access protocols*: The nodes access the shared medium with no coordination among themselves. Thus, when more than one packet are 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).

One can classify the multimedia traffic streams depending on their data burstiness and delay requirements as shown in Figure 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 signaling. These different

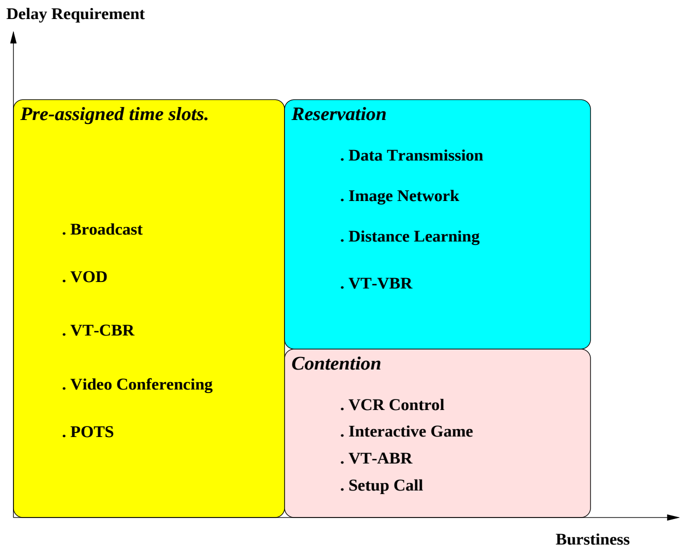

Fig. 1. Multiple traffic stream features and their appropriate MAC protocols.

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 cyclic and fixed available bandwidth. The best MAC protocols 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., 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 Figure 1. Although these applications do not generate a large amount of traffic data when compared to the other applications, they require a very low delay.

As we can see, none of these MAC protocols serves all types of traffic well although each one of them is ideal for a particular type of traffic stream whereas in a real world LAN/MAN network, traffic of all types would prevail [14]. In view of this, 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 as *multimedia medium access control* (*Multimedia*-MAC).

III. The *Multimedia*-MAC Protocol

The *Multimedia*-MAC protocol consists of three *sub-protocols* - namely, pre-allocation (wherein nodes access the shared medium in a predetermined way), reservation (wherein the nodes reserve one or more time slots within a frame before the actual packet transmission starts) and contention protocols (wherein the nodes access the shared medium with no coordination between them) - each of which serves a certain type of traffic. Note that the video/audio data (basically constant-bit-rate traffic, in our case) streams are best served by allocation based protocols (denoted by TDM). Similarly, the bursty (VBR) traffic is served best by reservation based MAC protocols (denoted by RSV) and finally the urgent messages are best handled by contention based (denoted by CNT) protocols. Obviously, the integration of these three protocols would serve a wide range of multimedia traffic streams quite well.

A time division multiplexing scheme controls the three different access strategies into a single protocol. Whenever a protocol uses the medium, the medium access is controlled according to the discipline of that protocol. A cycle in a *Multimedia*-MAC consists of a *fixed time frame* (of length L_{frame} slots) which consists of three segments namely TDM segment (of length L_{TDM} slots), RSV segment (of length L_{RSV} slots) and CNT segment (of length L_{CNT} slots) in that order.

IV. Analytical Modeling of the *Multimedia*-MAC Protocols

In our analysis of *Multimedia*-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 to be 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. Our model involves a queueing system with vacation (see Figure 2). The probability measure evaluated is the waiting time distribution of packets served by the different sub-protocols. 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 sub-protocols.

In the Multimedia-MAC protocol, the packets belonging to different traffic streams are served by different sub-protocols, which control the access to the shared medium during different segments in the fixed time frame. This is analogous to a queueing system with vacation (server unavailable for random time). This justifies our approach to model the queueing situation encountered by packets in the Multimedia-MAC protocol using queueing models with vacation. Here we successfully apply an approach which uses the decomposition properties of a general queueing system with vacation to compute the moments of the waiting time distribution from its Laplace-Stieltjes Transform (LST). Also an approximation for the tail probability from

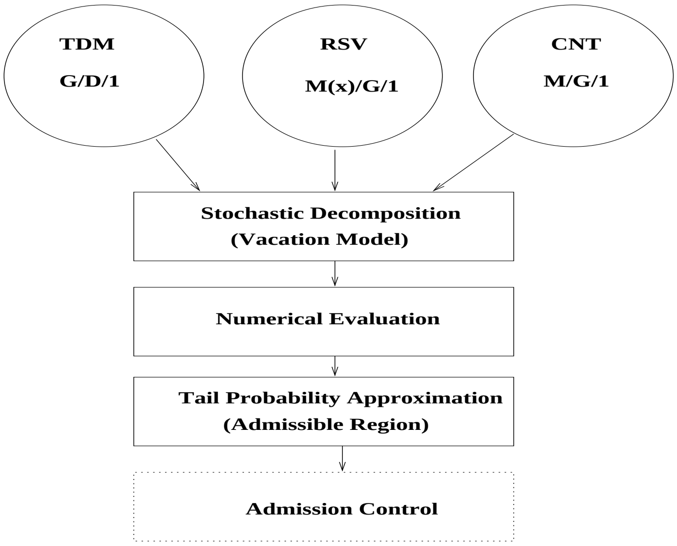

Fig. 2. Our approach to model and analyze the *Multimedia*-MAC protocol.

the moments is computed.

A. Queueing Models with Vacation for modeling Multimedia-MAC Protocol

In this subsection, we study different queueing models with vacation (The reader is referred to [20] for the corresponding queueing models *without* vacation).

The delay time of a packet from its arrival to its departure consists of three components: the queueing time, the service time and the vacation time. According to the decomposition property [7], [15], [17] in a general GI/G/1 queue under appropriate conditions, the waiting time is the sum of two independent random variables: the waiting time in the queueing system without vacation and *forward recurrence time* in the vacation. Let V be a random variable of the time from the arrival of a tagged customer during a vacation period up to the end of the frame and let $V^*(s)$ be the LST of the distribution of V. Use W_d to denote the total delay time (waiting time plus the service time) with vacation and $W^*(s)$ be the LST of the corresponding distribution. Also use $W_1^*(s)$ to denote the LST of the distibution 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 random variable $B(t)$. Then,

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

We use this as the key relationship to derive the LST of the distribution for the waiting time of packets served by different sub-protocols in our Multimedia-MAC protocol. As mentioned earlier, a cycle in a *Multimedia*-MAC protocol consists of a *fixed time frame*. Let T_{frame} denote the frame time and L_{frame} is the frame length in bits. Given the channel transmission rate is $\mathcal{R}$ and neglecting the gaps between frames, we have $\mathcal{R} = L_{frame}/T_{frame}$. The frame is further divided into several segments corresponding to the three sub-protocols. The segment lengths of the the TDM sub-protocol, the RSV sub-protocol and the CNT sub-protocols are denoted by L_{TDM}, L_{RSV} and L_{CNT}, respectively. The segments are further divided into fixed-length *slots* which are of length l_{slot} bits. For simplicity, we will measure the $L_{frame}, L_{TDM}, L_{RSV}$, and L_{CNT} in units of l_{slot}.

A.1 TDM sub-protocol model

The transmission using a TDM sub-protocol is controlled by a pre-allocated scheduling table. Each node, in turn, takes a TDM-segment to transmit its packets. Hence, from the point of view of the queue on each node, the transmission can be permitted only in its allocated TDM segment. The server is in vacation otherwise. Because we assume the transmission is slotted using fixed-length slots, which is of the same size as the packet size, the service time $B(t)$ can be considered as a *deterministic service*. Thus, the corresponding LST is given by,

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

where μ is the service rate in packets/sec. Because of the fixed transmission cycle of the TDM sub-protocol, the vacation period is also fixed in length. The traffic intensity is deterministic with mean λ_{TDM}. The arrival instant of an arbitrary (tagged) packet is uniformly distributed during the vacation period. Hence the forward recurrence time is given by,

$$V^*(s) = \frac{1 - e^{-\lambda L_v s}}{s\lambda L_v} \tag{3}$$

where $L_v = T_{frame} - T_{TDM}$ is the vacation period. Using the decomposition property and noting that the waiting time in a D/D/1 queue is 0, we have for the LST of the distribution for delay,

$$W^*_{TDM}(s) = \frac{(1 - e^{-\lambda L_v s})e^{-s/\mu}}{sL_v\lambda} \tag{4}$$

A.2 RSV sub-protocol model

In the RSV sub-protocol model, a node keeps on reserving the RSV segment in the next frame until its backlogged packets are served (until then, it holds the token). Thus, a node may get more than one RSV segment for transmitting *all* the packets. Hence, one service cycle consists of the sum of service times of packets backlogged in all nodes. In this case, the distribution of the vacation time is quite complicated to obtain [17]. 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 [17], where k is the number of transmitting 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 time of the service time obtained by each node is T_r, then the LST of the distribution for vacation time for each of the nodes is:

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

The RSV sub-protocol is designed for bursty traffic transmission. In particular, the packet arrival is modeled as a Poisson process with bulk arrivals. We know that for $M^{(x)}/G/1$ queue, the following result for the LST of the distribution of the waiting time W_1 holds [20],

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

where, $\rho = \frac{\lambda g}{\mu}$ and $G(.)$ is the arrival distribution of the bulk length random variable with geometric distribution with mean g.

From the decomposition property, we get the LST of the packet delay of the RSV sub-protocol as follows:

$$W_{RSV}^*(s) = \frac{(1-\rho)}{(s-\lambda_{RSV}+\lambda_{RSV}G[B^*(s)])} \times\frac{s(1-G[B^*(s)])}{(g[1-B^*(s)])}\left(\frac{N}{N+sT_r}\right)^N B^*(s) \quad (7)$$

A.3 *CNT sub-protocol model*

The CNT sub-protocol is a random access protocol. It best serves applications which produce messages of relatively small size but require low delay.

The system assumed in this work for the CNT sub-protocol 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 M-MAC protocol is divided into l_{slot} in bits, i.e., transmission time of each packet is one slot.
2. Defer first transmission principle (DFT) is employed. The transmission probabilities for the deferred packet is p.
3. The channel is noise free. The collision is the only reason for an unsuccessful transmission. No packet survives a collision.
4. All packets are of the 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 departed immediately. If the transmission fails, the packet is retransmitted in next slot with probability p.
5. The arrival process of packets follows the Bernoulli's process. That is, a packet arrives in a slot with probability λ and no packet with probability $1-\lambda$.
6. The serving of packets in a queue of each user is on 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 sub-protocol is modeled as a Geo/Geo/1/K queue with deterministic vacation. For analysis we use the tagged user approach (TUA) used in [16]. 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. p_s is the probability that a user transmits a packet successfully in a slot given that it makes a transmission in that slot. In [16], it has been shown that the probability generating function (PGF) $B(z)$ of the packet service time is given by,

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

where

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

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 $i = 1, 2, \ldots, L-1$ is given by

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

$$p_L = \frac{\lambda}{\mu}\left[\frac{\lambda(1-\mu)}{\mu(1-\lambda)}\right]^{L-1} p_0 \quad (11)$$

Using total probability law, we find

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

In order to solve for the two unknown variables μ and p_0, from (9) and (12), a numerical algorithm proposed in [16] is used here. Now the response time or delay (r) is defined as the time duration from when a packet enters the queue until its successful departure. The PGF of system response time is given by

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

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

$$E(r) = \frac{E(l)}{\lambda(1-p_B)} \quad (14)$$

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

where $E(l)$ is the mean queue length $E(l)$. By decomposition property (1), the Z transform of the delay distribution of packets in a CNT sub-protocol is given below.

$$D^*_{CNT}(s) = \frac{R(z)}{L_v}\left[\frac{1-z^{-L_v}}{1-z^{-1}}\right] \quad (15)$$

B. Numerical Evaluation of the Waiting Time Distribution

In the previous section, we have derived the LST of the waiting time of packets belonging to a traffic type served by different sub-protocols within the framework of the *Multimedia*-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 [3]. A Fourier-series method which can numerically invert the Laplace transforms and generating functions by a numerical method is discussed in [1], [2], [3]. Here we give the gist of the method [5] which computes the nth moment of a given continuous (discrete) random variable from the LST (PGF) of its distribution.

To begin with we give some definitions. For a non-negative discrete random variable X, let $p_k = Pr\{X = k\}, k = 0, 1, ...,$ then its probability generating function is defined by

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

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

$$F^*(s) = E[e^{-sX}] = \int_0^{\infty} e^{-sx} dF(x) \quad \text{For } Re(s) \geq 0 \quad (17)$$

The moment generating function (MGF) of a continuous random variable X is a function $M : R \rightarrow [0, \infty)$ given by $M(t) = E[e^{tX}]$. Let μ_n represent the nth moment of X, then, $\mu_n = M^{(n)}(0)$. Note that,

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

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

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

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

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

From the above equation, 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 [5]. The idea is to use the Cauchy integral formula for the sequence w_n and compute numerically this integral using m - point trapezoidal rule. To avoid large discretization errors due to either too fast or too slow an increase in $\frac{\mu_n}{n!}$, an adaptively modified moment generating function is inverted to get the nth moment of the given MGF (PGF). This involves the inclusion of the factor called *adaptive decay rate* 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, from

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

where

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

For α_n $n = 1, 2$, we use the following procedure: we arbitrarily set $\alpha_1 = 1$ 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:

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

where,

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

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 [5] where it has been shown that the method is reasonably accurate.

C. Deadline Missing Rate: The Tail of the Waiting Time Distribution

DMR is the probability $F_{DMR}(D)$ that the waiting time of an arbitrary packet served by a sub-protocol exceeds a given deadline (D), i.e., $F_{DMR}(D) = Pr\{W > D\}$. The DMR is useful in characterizing the QoS of a given traffic. (Note that $F_{DMR}(D)$ is the tail of the waiting time distribution). However, given an arbitrary deadline (D), the computation of waiting time distribution is quite difficult because of 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 sub-section. Therefore, we need to find an approximation to obtain the tail probability. In view of this, we draw the attention of the approximation [5] which is given below,

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

where $\eta = \lim_{n\to\infty} \eta_n$ and $\eta_n = n\mu_{n-1}/\mu_n$ and $A = \lim_{n\to\infty} A_n$ 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.

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, the 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. In particular, the DMR can be used for the call admission control as given below: Given the $F^t_{DMR}(D)$, 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 the $F^{t+}_{DMR}(D+)$, the DMR at the next time instant $t+$ (with required new deadline $D+$). If $F^{t+}_{DMR}(D+) < \Psi$, then the traffic is admitted or else it is rejected. In our case, the DMR of aggregate traffic at any time for the three sub-protocols (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 gives fairly an idea about the network status. This is plotted in Figure 3 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 *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 sub-protocols), traffic arrival, service and vacation distributions and deadline (D), the above mentioned surface can be computed. Then, using this surface and the call admission control, we can 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 of long-term statistics (which are done conventionally).

V. The Application of *Multimedia*-MAC on WDM Networks

The *Multimedia*-MAC design presented in the previous section can be applied to a wide variety of shared medium networks [18], [19]. Here, we apply the *Multimedia*-MAC to WDM networks. Based on the idea of our proposed

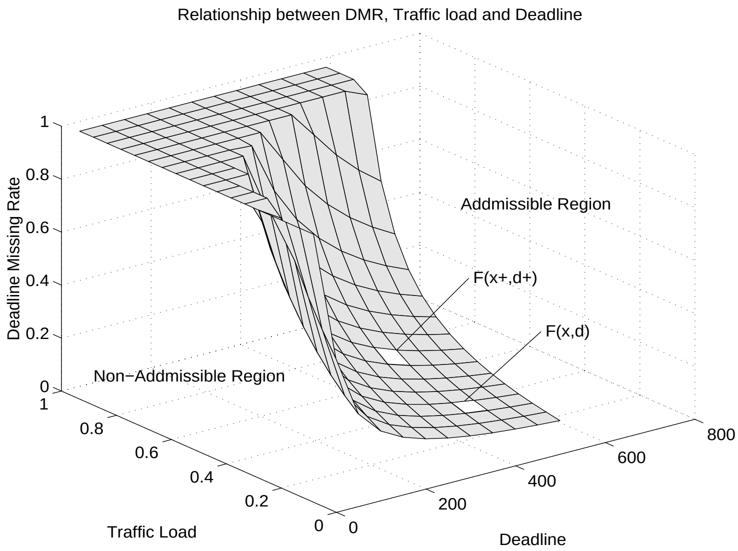

Fig. 3. Admissible region as a function of DMR, deadline, and traffic load.

Multimedia-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 serve well a wide range of traffic streams for multimedia applications. This MAC protocol is called as *Multimedia-WDMA* (M-WDMA).

A. The M-WDMA Architecture

We consider the single-hop topology where a WDM optical network is configured as a broadcast-and-select 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 destinations. The nodes in a WDM network can transmit and receive messages on any of the available channels by using and tuning one or more tunable transmitter(s) and/or tunable receiver(s). Consider a M-WDMA network with N nodes which are connected by a star-coupler and having C channels. Each node has a fixed channel (*home channel*) with wave-length λ_i $(i = 1, 2, .., C)$. The home channels are intended for the destination nodes to receive packets. In case the number of available data channels $C < N$, several nodes ($\lceil \frac{N}{C} \rceil$) 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 consists of three integrated *MAC sub-protocols*, namely, a TDMA protocol, a reservation-based protocol and a random-access protocol each of them active in a segment in the fixed time frame. We have data channels and control channels to transmit the data and control informations respectively.

Data channel: A time division multiplexing scheme controls the three different access strategies into a single protocol. Whenever a protocol uses the medium, the medium

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

access is controlled according to the discipline of that protocol. A cycle in a *Multimedia*-MAC consists of a *fixed time frame* (of length L_{frame} slots) which consists of three segments namely TDM segment (of length L_{TDM} slots), RSV segment (of length L_{RSV} slots) and CNT segment (of length L_{CNT} slots) in that order. Note that in the TDM and RSV sub-protocols, only a single node can access a channel *at a time* since they are all protocols based on pre-allocation and reservation in advance. But, in the CNT sub-protocol, multiple nodes can access the same channel *at the same time*.

Control Channel Configuration: The control channel operates in a TDMA manner independent of the data transmission (and is coincident). One cycle consists of N *mini-slots*, 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 sub-protocol; and 2) collision acknowledgments in the CNT sub-protocol. For the reservation of 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 sub-protocol. Note that the length of a mini-slot is $N+L_{cnt}$. Hence, the cycle time $T_{ctrl} = \frac{N(N+L_{cnt})}{R}$ 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, $\frac{N(N+L_{cnt})}{l_{slot}} < L_{frame}$ where l_{slot} is the slot length in bits. We find that from the above inequality,

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

This may lead to a network scale limitation. For example, if we choose, $L_{cnt} = 10$ slots, $L_{frame} = 30$ slots, $l_{slot} = 424$ bits, then the number of nodes should be $N < 107$. By adding an exclusive receiver for the collision detection at each node, the inequality becomes $N < \sqrt{l_{slot}L_{frame}}$ (which is 113 for the example above). Alternatively, we can increase the packet size to increase the network size.

Transmitter and Receiver Configuration: Each node has three tunable transmitters corresponding to the three sub-protocols. The three transmitters are used to serve the three different classes of traffic streams. These three transmitters operate in a *pipeline* fashion. That is, when one transmitter is transmitting a packet, the other transmitters tune to the next channel. Thus the simultaneous transmission of three types of traffic is avoided. An example of a configuration for transmission channels from all nodes in our M-WDMA MAC protocol is illustrated in Figure 4. In this figure, the X-axis denotes the time-

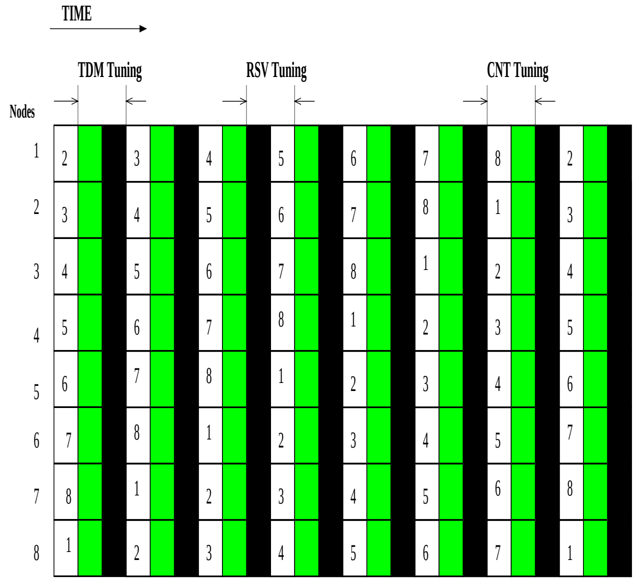

Fig. 4. A M-WDMA transmission schedule example.

slot and the Y-axis denotes spatial location of the backlogged nodes. The white segments 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 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, that means at that time, the current TDM segment is transmitted on channel 6. Next, the fifth frame, the frame will be transmitted on channel number 7. All these transmissions happens 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 are transmitting even though some of their transmitters may be in a non-transmitting mode (tuning to a certain channel). The tuning time (Γ) satisfies the following relationships: $L_{TDM} + L_{RSV} > \Gamma$, $L_{TDM} + L_{CNT} > \Gamma$ and $L_{CNT} + L_{RSV} > \Gamma$. It follows that $L_{frame} > 3\Gamma/2$. In particular, the length of the frame directly affects the bandwidth allocated to TDM segments. Hence, the total TDM bandwidth is given by: $B_{TDM} = \frac{L_{TDM}R}{L_{frame}(N-1)}$ where R is the channel bandwidth.

B. The M-WDMA Protocol

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

B.1 The TDM Sub-protocol

The operation of the TDM sub-protocol within our M-WDMA network is basically an *interleaved TDMA* MAC protocol [13]. The only difference between our TDM sub-protocol and ITDMA are 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 a RSV segment, the TDM transmitter starts to tune to the next channel. Note that the TDM sub-protocol does not need any control information to be transmitted over the control channel.

B.2 The RSV Sub-protocol

In a M-WDMA MAC protocol, the RSV packet transmission is controlled using a *multiple token* method [4]. An obvious advantage of using a token-based scheme in our RSV sub-protocol is that it can efficiently support 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 to a perfect scheduling scheme since in a frame only one single node is allowed to access the RSV segment at a time.

Each channel 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 *token rotation algorithm* to determine the token distribution in the next frame. The *token rotation algorithm* is depicted below. Here we use the round robin scheme for the pre-fixed order. This allows all the nodes to get equal chances for transmitting or receiving packets. It is worth to note 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 (destination) channel. The token rotation is executed according to the control channel timing.

[**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 dose not have
        any token;
Allocate the token to the node;
remove the node from the checking list;
remove the token from the token list;
}
```

B.3 The CNT Sub-protocol

The CNT sub-protocol of our M-WDMA MAC protocol is similar to the *interleaved slotted ALOHA* [13]. The active nodes compete for the slots in the current CNT segment. In case there is a collision, the retransmission is scheduled after a random number of slots (In particular, the geometric transmission attempts is assumed). In the context of the M-WDMA, two issues need to be addressed: handling of acknowledgement and the retransmission of collided packets. In M-WDMA, there are two receivers at each node, one is the home channel receiver and the other is the control channel receiver. Hence, the collision at the receiver side would be detected and the control channel keeps track of these collisions. Then, the retransmission of the packets by the same node would be attempted by the nodes. The second issue that needs to be addressed is the retransmission of collided packets. The collided packets have to be retransmitted *within* the current CNT segment, otherwise it will collide with the other types of transmissions. 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. Hence, 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 sub-protocols operate independently and one can think of these protocols operating in three different networks. These three virtual networks have bandwidth equal to the bandwidth allocated to that sub-protocol in a M-WDMA network. This is reasonable under the assumption that the three classes of traffic are independent of each other (called *protocol independent assumption*). With this assumption, the three sub-protocols can be studied independently. One can think of a (transmitting) node (logically) consists 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 sub-protocol 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 sub-protocols. Then, $\lambda_i = \frac{L_i}{L_{frame}}\lambda$ 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 FCFS discipline. Let $B(x)$ be the distribution function for the service time, with $1/\mu$ being its mean and $B^*(s)$ being its Laplace-Stieltjes Transform (LST). We denote the random variable 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.

C.1 The TDM Sub-protocol Model

According to a TDM sub-protocol, 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 $v_i = (N-1)L_{frame}T_{slot}$. By traffic independent assumption and statistical similarity of nodes, the queue is generic with vacation period given by,

$$V = (N-2)L_{frame}T_{slot} + (L_{frame} - L_{TDM})T_{slot} \quad (25)$$

Using L_v to denote the number of slots constituting the vacation period in the 'generic' queue, we have for L_v,

$$L_v = (N-2)L_{frame} + (L_{TDM} - L_{frame}) \quad (26)$$

Noting that $\mu = R$ (where R being the channel bandwidth), the delay (waiting time plus the service time) distribution is given by (4). Further the numerical evaluation method presented in sub-section IV-B allows one to compute the mean delay which is a performance measure studied here.

$$E[W_{TDM}] = W^{*(1)}_{TDM}(0) = \mu_1 \quad (27)$$

Computing the nth moment of the waiting time distribution from (23) and using the approximation (24) given in Section 4, we compute the deadline-missing rate (DMR). Here, we use first 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 $< 2\%$. Figure 5 show 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. [3]

[3]The analytical model for the mean delay is even more accurate than that for the DMR. It is not shown here graphically, because of the limitation imposed by the IEEE Transactions on Communications Editorial Board on the maximum number of figures that can be included in any paper which is set to 10. As a result, we are not including the figures that correspond to mean delay for all sub-protocols. However, these can be supplied at a moment notice.

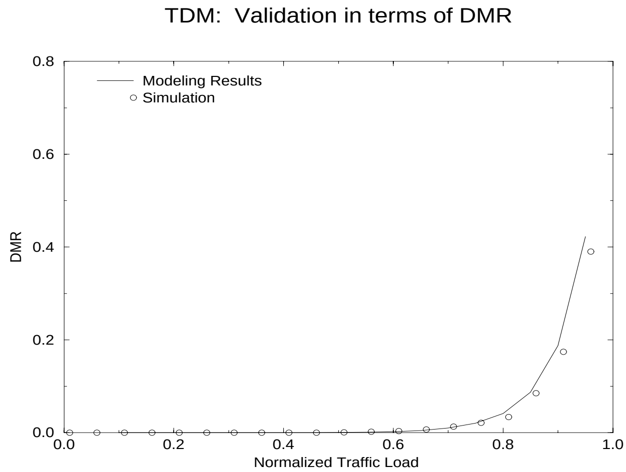

Fig. 5. TDM model validation in terms of DMR.

C.2 The RSV Sub-protocol Model

In a M-WDMA, the RSV sub-protocol 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 tokens 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). Hence the number of frames a node takes to transmit a message depends on the size of the bulk (G). The LST of the delay of a packet ($D^*_{RSV}(s)$) in the RSV protocol model can be obtained using (6). (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,

$$G(z) = \frac{1}{g(g+1-z)} \quad (28)$$

Now, the queueing situation is similar to the case analzsed in sub-section IV-A.2 and the mean waiting time and DMR can be obtained. The LST of the delay of a packet ($D^*_{RSV}(s)$) in the RSV protocol model can be obtained using (6). (Note that the parameter T_r can be obtained from $1/\mu$ and the distribution of G).

Computing the nth moment of the waiting time distribution from (23) and using the approximation (24) given in

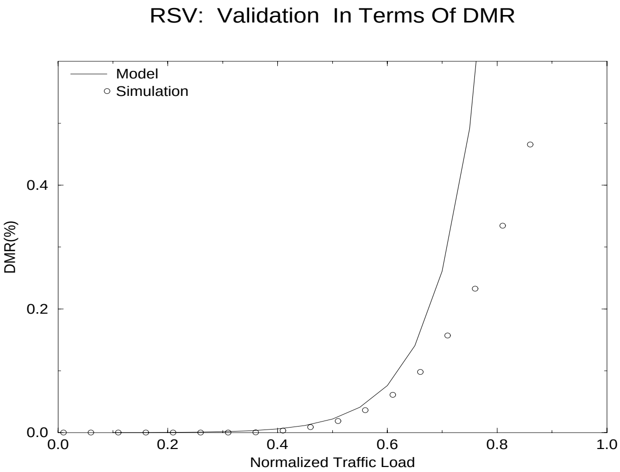

Fig. 6. RSV model validation in terms of DMR.

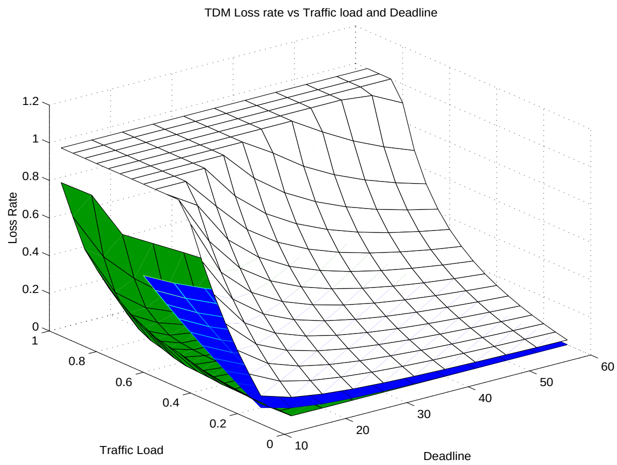

Fig. 8. Admissible Region for CBR traffic.

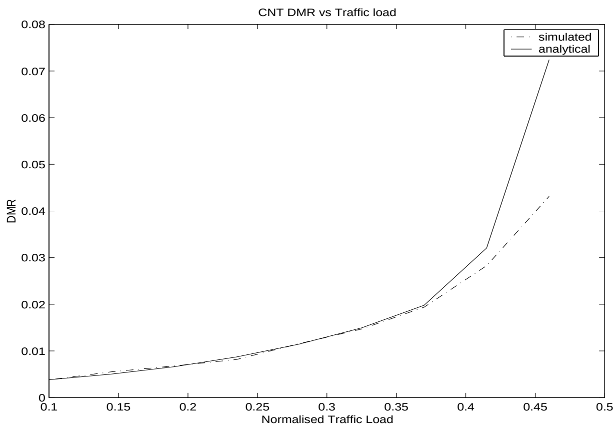

Fig. 7. CNT model validation in terms of DMR.

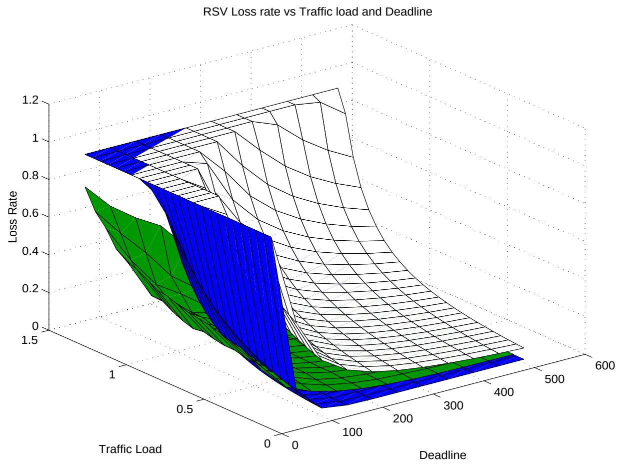

Fig. 9. Admissible Region for RSV traffic.

Section 4, we compute the deadline-missing rate (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 first n^* moments and first n^*+1 moments is $< 2\%$. Similar to the TDM model, we can calculate the DMR according to (24). Figure 6 compares the analytical results with the simulation results. The two results are close to each other especially when the traffic load is light.

C.3 CNT Sub-protocol 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., $V = (L_{TDM} + L_{RSV})T_{slot}$ (*unlike the queues in the RSV and TDM sub-protocol*). The expression for the delay in this case is obtained from (15) with $L_v = L_{TDM} + L_{RSV}$. Figure 7 illustrates the validation of our analytical model in terms of the DMR parameter. As we can see, again the results are reasonably close to each other.

C.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 TDM, RSV and CNT sub-protocols, for normalized traffic load and a range of required deadlines, we can obtain the corresponding *admissible region* as shown in Figures 8, 9 and 10.

In Figures 8, 9 and 10, we show three types of surfaces, the lowermost surface is generated from our analytical modeling results. The top layer surfaces are an upperbound computed according to *Chebyshev's inequality* [8], which are used as modeling references (in fact, a simple upper bound). The middle layer surface is obtained through intensive simulation.

By comparing these three figures, we can see the admissible region of a TDM sub-protocol is quite flat. This is because in a M-WDMA, the TDM sub-protocol has the highest priority for bandwidth allocation. This results in a lower DMR (analytical) bound even when the traffic load

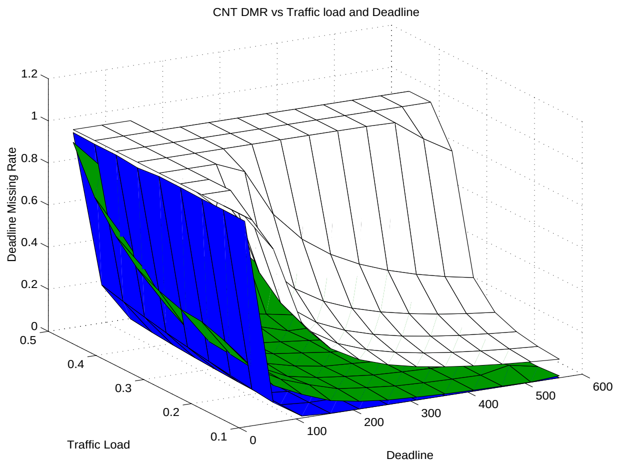

Fig. 10. Admissible Region for CNT traffic.

is high. For high traffic loads, the (analytical) DMR is not very high in the case of RSV sub-protocol. This is the advantage of the RSV sub-protocol because the RSV sub-protocol is primarily used for bursty traffic with relatively loose delay requirements. However, when the deadline becomes strict, the QoS of a RSV sub-protocol is eventually worse than that in TDM - as is expected.

The CNT admissible region is realistic 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, for example) 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 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, deadline missing rate). 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 wavelength division multiplexing network. We have shown that our general framework can readily be used for such networks. If addition, we have shown that our analytical results are reasonably accurate when compared to 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.

References

[1] J. Abate, G. Choudhury, and W. Whitt. Exponential approximations for tail probabilities in queues: I: Waiting times. *Operations Research*, 43(5):885–901, September-October 1995.

[2] J. Abate, G. Choudhury, and W. Whitt. Exponential approximations for tail probabilities in queues II: Sojourn time and workload. *Operations Research*, 44(5):758–763, September-October 1995.

[3] J. Abate and W. Whitt. The fourier-series method for inverting transforms of probability distributions. *Queueing System*, 10:5–88, 1992.

[4] C. M. Krishna Anlu Yan, Aura Ganz. A distributed adaptive portocol providing real-time services on WDM-based LAN's. *Jounal of lightwave technology*, 14(6), June 1996.

[5] G. Choudhury and D. M. Lucantoni. Numerical comutation of the moments of a probability distribution from is transform. *Operations Research*, 44(2):368–381, March-April 1996.

[6] I. Cidon and Y. Ofek. "Metaring - A Full-duplex Ring with Fairness and Spatial Reuse". *Proc. INFOCOM'90*, 3:969 – 981, 1990.

[7] M. Doshi. G/G/1 queue with general vocation—a survey. *Queueing System*, November 1989.

[8] G. R. Grimmett and D. R. Stirzaker. *Probability and Random Processes.* Oxford Science Publications, 1992.

[9] H. R. Muller, M. Mehdi Nassehi and J. W. Wong. "DQMA and CRMA: New Access Schemes for Gbit/s LANs and MANs". *Proc. INFOCOM'90*, 1:185–191, 1990.

[10] I. F. Akyildiz, J. McNair, L. C. Martorell, R. Puigjaner, and Y. Yesha. "Medium Access Control Protocols for Multimedia Traffic in Wireless Networks". *IEEE Network*, pages 39–47, July/August 1999.

[11] V. Iyer and S. Joshi. "FDDI's 100 Mbps Protocol Improves on 802.5 spec's 4 Mbps Limit". *EDN*, pages 151–160, May 2 1985.

[12] J. Sanchez, R. Martinez and M. Marcellin. "A survey of MAC protocols proposed for wireless ATM". *IEEE Network*, pages 52–62, Nov/Dec 1997.

[13] P. W. Dowd K. Borgineni, K. M. Sivalingam. Low-complexity multiple access protocols for wavelength-division multiplexed photonic networks. *IEEE Journal on Selected Areas In Communications*, 11(4), May 1993.

[14] L. Lenzini, J.O. Limb, W. Lu, I. Rubin, and M. Zukerman. Analysis and synthesis of mac protocols: Special issue. *IEEE Journal on Selected Areas In Communications*, 18(9), Sept. 2000.

[15] J. Medhi. *Stochastic Models in Queueing Theory.* ACADEMIC PRESS, INC., 1991.

[16] T. Wan and A. U. Sheik. "Performance and Stability Analysis of Buffered Slotted ALOHA Protocols Using Tagged User Approach ". *IEEE Trans. Vehicular Technology*, 49(2):582–593, Mar. 2000.

[17] H. Takagi. *Queueing Analysis: a Foundation of Performance Evaluation*, volume I. North Holland, 1991.

[18] L. Wang and M. Hamdi. M-wmac: An adaptive channel access protocol for multimedia wireless networks. *Seventh International Conference on Computer Communications and Networks (IC3N'98)*, Oct 1998.

[19] L. Wang and M. Hamdi. M-wmac: An adaptive channel access protocol for personal communication system. *Wireless Personal Communication Systems*, 13:79–96, 2000.

[20] Lixin Wang. *"Multimedia Access Protocols for Shared Medium Networks"*. PhD thesis, Hong Kong University of Science and Technology, Hong Kong, 2000.