

Proposal, Simulation and Performance Appraisal of an Optimal Medium Access Protocol for Wireless ATM Networks

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ABSTRACT

With the ever increasing demand for easy portability and mobility of devices supporting diverse mobile multimedia applications, the need for the adaptation of broadband infrastructure to wireless scenario has arisen. Mobile multimedia networks like the wireless ATM are faced with challenges relating to wireless channel access. In this paper we propose and simulate policies vis-a-vis medium access in wireless ATM networks. We also discuss the modelling of traffic generators, the evaluation of delay and buffer length bounds and the implementation of a WATM MAC simulator.

I. INTRODUCTION

The current cellular systems are not capable of efficiently supporting multimedia traffic service for mobile users. With ATM extending support to diverse traffic types and mobility being an inevitable need of the multimedia applications users, wireless ATM is a promise for mobile multimedia traffic support. Provisioning medium access for a number of mobile users over an error prone wireless channel is difficult. This is because of the stringent QoS requirements of some of the traffic classes, high bit rates involved and existence of an ATM backbone which was conventionally designed to operate over reliable media like optical fiber. We propose and simulate a medium access protocol employing policies conducive to provisioning of desired QoS to distinct traffic classes in multi-class multi-host scenario and efficient request access contention mechanism for mobile hosts supporting real time and non-real time traffic. The modeling of distinct traffic classes is also proposed. We analytically evaluate the bounds on buffer length and cell delays. Based on the observation of the results we propose a traffic shaping policy to optimize the QoS of traffic streams which have traffic rates close to and less than the inverse of the frame length duration

II. THE MAC PROTOCOL

In this section we describe salient strategies for optimal medium access in WATM networks. A frame structure comprising of 4 phases pertaining to downlink transmission of Protocol Data Units, channel access request contention, slot probing and transmission permission, and uplink data transmission respectively has been adopted

A. Channel Access Contention Resolution

For channel access contention we adopt Unsymmetric Randomized Slotted Aloha algorithm [2]. At the commencement of the

request access contention phase all the mobiles chose a random period less than a maximum wait interval after which they contend for requesting access. If during the slot in which a host requested access, there is a collision, the mobile host chooses another wait period that is larger than the previous maximum wait period by a factor (which has been found to be optimal at a value close to "e" [2]). In case there is a slot during which there is no contention for the channel, all mobiles reduce their remaining wait period to a random duration that is less than their present remaining wait period. This improves the delay on the request for channel access, especially the last few remaining requests.

B. Multiclass Scheduling

For the CBR and rt-VBR classes, Channel Condition Independent Fair Queuing [1] scheduling algorithm has been implemented. The CIF-Q adopts the Start-time Fair Queuing (SFQ) [3] and pertinent alleviation measures for the wireless scenario. The CIF-Q ensures delay bound and throughput guarantees for error free sessions. It provisions long term fairness implying that if a session becomes error free it should get back all the service lost when it was in error. The algorithm also ensures short term fairness which means that the difference between the normalized services received by any two error-free backlogged sessions during a time interval is bounded. In addition an error free backlogged session which has till now received more service than its fair share is guaranteed to receive at least a minimum fraction of its service in the error free system. This feature is the property of graceful degradation of service. For remaining traffic classes nrt-VBR, ABR and UBR round robin scheduling is implemented. In the nrt-VBR and ABR classes packet transmission is deferred if the channel is bad. However in UBR case, packets are dropped in case of bad channel state.

C. Modelling of Wireless Channel and Data Traffic Generators

The channel state is approximated by a two-state model. At a particular time, the channel state is classified as either *good* or *bad*. It is further assumed that if the state of the channel for any mobile host is *bad*, then the packet transmitted to or from that mobile will be completely lost. Accordingly, if the channel state is *good*, the packet will be correctly received. There is a fixed probability for transition of the channel from *good* to *bad* and corresponding probability for transition from *bad* to *good*.

Traffic generation for the CBR case is done by taking the inter-arrival time of packets as constant. The rt-VBR and nrt-VBR traffic is generated via a two state Markov model corresponding to

two states of packet generation at the reduced cell rate and the PCR averaging to the SCR of the source. The inter-arrival time of the ABR and UBR classes is taken to have a Pareto distribution

III. BUFFER LENGTH AND DELAY EVALUATION

The estimation of bounds on buffer lengths and cell delays on the basis of QoS parameters of the traffic classes is essential. At the time of call admission, the base station should assess the feasibility of accommodation of a connection on the basis of the bandwidth available and the bounds on buffer lengths, cell delays and jitter that will be applicable to a traffic source if it is accommodated. Hence the translation of the QoS parameters of the multi-class traffic to these bounds becomes important.

A. Bounds on Buffer Lengths

The CBR buffer bound can be evaluated as follows. Consider that a CBR cell arrives at the termination of the uplink transmission phase of a frame. All the cells arriving in a given frame are buffered and served in the next frame. Hence, the buffer servicing for the cells buffered in the current frame begins from next transmission phase which starts after a duration of frame length transmission time (FL) plus the pre-transmission phase duration (ptp) of the next frame. The pre-transmission phase duration is equal to the time of transmission of first three phases of a frame. Therefore,

$$B_{CBR} = 1 + (FL + ptp).PCR \quad (1)$$

For the rt-VBR traffic a burst can arrive while the flow is receiving service at SCR. This translates to a buffer requirement of $MBS - (MBS/PCR).SCR$. The arrival of a burst can span the duration of transmission of any number of frames. In the worst case consider that the burst arrival is complete when the transmission phase of a frame ends. Then before the servicing of rt-VBR buffer begins, ptp time and in addition the time corresponding to servicing CBR buffers will lapse. If C is the link speed in no of packets per second, the upper bound on rt-VBR buffer requirement is hence given by

$$B_{rt-VBR} = MBS - \frac{(MBS/PCR).SCR + ptp.SCR}{\sum B_{CBR}/C}.SCR \quad (2)$$

For the nrt-VBR case the connection should be able to store data while the higher priority CBR and rt-VBR traffic sources drain their buffers. Similar to the rt-VBR consideration above, a burst may span duration of transmission of any number of frames. On the completion of burst arrival at the termination of the transmission phase of a frame, time corresponding to servicing CBR and rt-VBR buffers will lapse before the nrt-VBR buffers are serviced. In the worst case consider that the CBR and VBR buffers are completely flushed in the frame under consideration.

$$B_{nrt-VBR} = MBS - \frac{(MBS/PCR).SCR + ptp.SCR}{\sum B_{CBR}/C}.SCR + \frac{\sum B_{rt-VBR}/C}.SCR \quad (3)$$

As regards the ABR traffic, packets should be buffered while the higher priority buffers are being serviced. Guarantee vis-a-vis Minimum Cell Rate (MCR) is given by the base station at the time of call admission. ABR connections should therefore be able to buffer the following

$$B_{ABR} = 1 + (FL + ptp).MCR + \frac{\sum B_{CBR}/C}.MCR + \frac{\sum B_{rt-VBR}/C}.MCR + \frac{\sum B_{nrt-VBR}/C}.MCR \quad (4)$$

UBR best effort traffic has no hard guarantees. Hence the buffer length requirement need not be estimated.

B. Bounds on Cell delays

We now evaluate the cell delay bounds corresponding to various traffic classes. The bounds are evaluated for the scenario when there are no wireless channel errors.

CBR traffic sources occupy the first priority level. A cell belonging to CBR traffic flow will be delayed waiting for its transmission in the transmission phase of the frame following the frame of its arrival. In the worst case, there will be additional delay corresponding to the time when other CBR sources send their share of channel bandwidth plus the time to transmit the packet for which delay is being evaluated. If $r_{CBR}(j)$ is the rate of jth CBR source, F_{CBR} is the set of flows of CBR class and L is the cell length, the HOL cell delay bound for the CBR traffic source is given by

$$D_{CBR}(i) = (FL + ptp)/C + [1 + \frac{\sum_{j \in F_{CBR}, j \neq i} r_{CBR}(j)}{r_{CBR}(i)}].L/C \quad (5)$$

For the rt-VBR traffic delay bounds, consider the number of aggregations (equal to PDU length) of bytes that will be transmitted in the duration of delay suffered by a cell of a rt-VBR source. This number will be equal to number of aggregations of bytes that will be transmitted in (FL + ptp) duration plus the number of PDUs of CBR traffic sources arriving in this delay time plus the buffered PDUs in all CBR queues plus the number of PDUs corresponding to the rates of all the other rt-VBR sources, i.e.,

$$D_{rt-VBR}(i).C = (FL + ptp) + D_{rt-VBR}(i)[\sum_{j \in F_{CBR}} PCR(j)] + \sum_{j \in F_{rt-VBR}} B_{CBR}(j) + [1 + \frac{\sum_{j \in F_{rt-VBR}, j \neq i} r_{rt-VBR}(j)}{r_{rt-VBR}(i)}].L \quad (6)$$

From the above equation the delay bound for rt-VBR traffic source is given by

$$D_{rt-VBR}(i) = \frac{(FL + ptp)/C + \sum_{j \in F_{CBR}} B_{CBR}(j)/C}{1 - \sum_{j \in F_{CBR}} PCR(j)/C} + \frac{(\frac{1 + \sum_{j \in F_{rt-VBR}, j \neq i} r_{rt-VBR}(j)}{r_{rt-VBR}(i)}) . L / C}{1 - (\sum_{j \in F_{CBR}} PCR(j)/C)} \quad (7)$$

The nrt-VBR, ABR and UBR need no delay guarantees. Hence delay bound evaluation is not necessary.

IV. MAC PROTOCOL SIMULATION

The simulations have been done via a C++ discrete event WATM MAC simulator. The simulator provisions a generic testbed for testing various features of WATM medium access control. A GUI interface has been developed in TCL and TK that helps the user to input the simulation parameters in a user-friendly fashion. A simulation run is over a scenario comprising of a given number of traffic sources belonging to various traffic classes. Each traffic source negotiates its pertinent class parameters at the time of call admission. Depending on analytic estimates of QoS provisioning, the call is either accepted or dropped. The accepted calls prevails for the rest of the simulation run. The average packet delay, buffer length, packet drops for each traffic class and overall throughput are the outputs of the simulator.

For the simulations the number of uplink PDU slots in every frame has been taken as 12. The number of access request slots for real time traffic and non-real time traffic have been taken as 8 and 16 respectively. The frame length is 1500 bytes. The probability

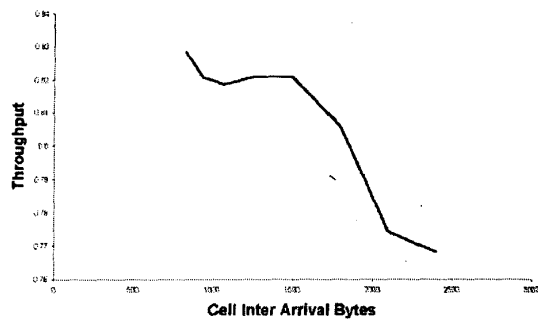


Fig. 1. Throughput Vs. Cell Interarrival Bytes

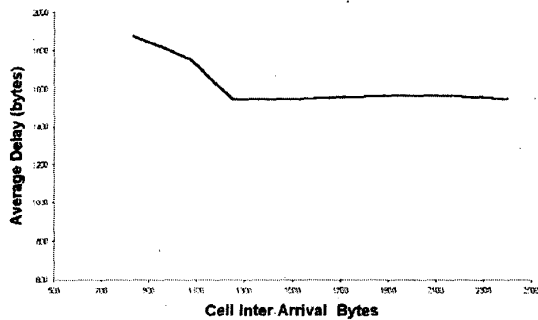


Fig. 2. Average Cell Delay Vs. Cell Interarrival Bytes

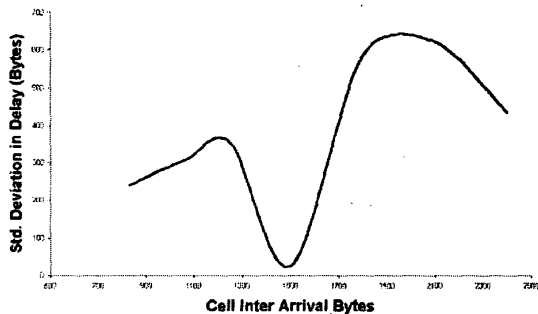


Fig. 3. Standard Deviation in Cell Delay Vs. Cell Interarrival Bytes

for retention of *bad* channel state has been taken as 0.001 and the probability of retention of *good* channel state as 0.999. the simulation run corresponds to the generation of total of 50000 packets. Representative results (for the CBR traffic) are presented in as network throughput (Fig. 1), average cell delays (Fig. 2), standard deviation of delay (Fig 3) and average buffer length (Fig 4). Each curve has been plotted against the cell inter-arrival time (corresponding to the sources of a specific class) expressed in bytes transmitted at link speed.

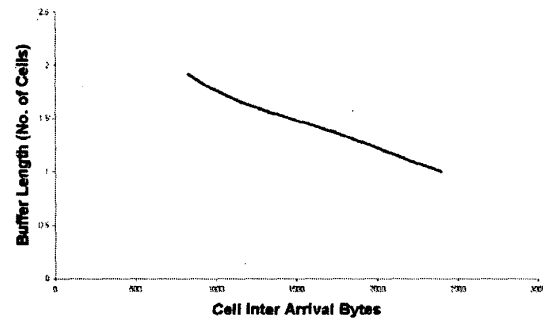


Fig. 4. Buffer Lengths Vs. Cell Interarrival Bytes

V. RESULTS

It can be seen from Fig. 1, that with increase in the cell interarrival bytes for the CBR traffic case, the overall network throughput degrades. The average CBR cell delay (Fig 2) decreases with decreasing traffic intensity (increasing cell interarrival bytes). The standard deviation in cell delays for the CBR traffic class is shown in Fig 3. The buffer length (Fig 4) increases with increasing traffic intensity (decreasing cell interarrival bytes).

The prevalence of local maximum in throughput and local minimum in delay and jitter at cell interarrival bytes equal to the frame length bytes is conspicuous for the CBR traffic. This occurs owing to a locking or resonance between cell generation and uplink PDU transmission. As an application of this observation, we propose traffic shaping mechanism to optimize performance vis-a-vis the throughput, delay and jitter for traffic sources which have cell interarrival times close to but less than the frame length duration. If for such sources traffic intensity were to be enhanced by adding dummy cells to the traffic stream to make the traffic intensity equal to one cell per frame, reduction in cell delay and jitter and increase in throughput could be achieved. Since the CBR sources are specifically sensitive to cell delays and jitter, reduction in these quantities means provisioning better QoS to these sources by artificially modifying the data stream.

VI. CONCLUSION

We have implemented optimal algorithms for request contention, scheduling and call admission in wireless ATM networks. Analytical derivation of the delay bounds and buffer length for our MAC protocol has been done. Also accomplished herewith has been the performance assessment of the proposed MAC protocol in terms of throughput, cell delays, jitter and buffer requirements for different traffic classes. Via simulations the possibility of enhancement in QoS of data streams by modifying the data rates has been explored. The bounds on buffer lengths and packet delays for various classes of traffic have been evaluated.

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