ATM SWITCH PERFORMANCE WITH DYNAMIC BANDWIDTH ALLOCATION

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ABSTRACT

A model is proposed for ATM switch to study its performance with dynamic bandwidth allocation. The proposed scheme allocates bandwidth dynamically, proportional to expected queue length and the threshold value associated with it. The scheme is then compared with the results of Lu and Hansson's research, where static bandwidth allocation and bandwidth allocated proportional to expected queue length are analyzed. The proposed scheme performs better under certain operating conditions when the threshold value is optimum.

Keywords: ***ATM, Bandwidth allocation, Modeling, Simulation, Dynamic time slice, Delay; Queue length***

1.0 INTRODUCTION

In today's telecommunication network, traffic from many applications is serviced by separate, dedicated networks, e.g., the traditional telephony for voice, and cable TV for motion pictures, etc. Broadband Integrated Services Digital Network (B-ISDN) is a communication network that integrates diverse services such as data, voice, and video [1, 2]. Asynchronous Transfer Mode (ATM) is the promising information transfer method for the B-ISDN [3, 4, 5]. It is a high speed, fast packet switching, connection-oriented, method using a fix-sized packet called cell. ATM protocol reference model consists of three layers, which are physical layer, ATM layer and ATM adaptation layer. Physical layer is used to transfer the ATM cells from one node to another. ATM layer provides the switching and routing of the ATM packets according to their virtual channel identifier (VCI) and virtual path identifier (VPI) labels. The ATM layer is also responsible for generating the headers for the ATM cells and extracting this header from incoming cells. Error detection/correction for the ATM header is another function of the ATM layer. ATM adaptation layer maps various types of traffic into and out of the ATM cells. ATM networks are connection-oriented networks and therefore it is possible for each connection to have a route set-up at the start of the connection. This route will remain the same for the duration of the connection to ensure cell sequence at the receiver. The cell must contain the connection identifier within itself that uniquely identifies the connection throughout the network.

As a result of the focused research on the ATM during the past few years, many challenging problems have arisen. ATM integrates different traffic classes (data, voice and video) together [11]. These traffic classes are type 1A, 1B, 2 and 3 and vary on their bandwidth requirements and tolerance to message delay and loss. For data source, it is sensitive to cell loss, but not delay. For voice and video traffic, they are sensitive to delay, but not cell loss. These different traffic classes thus demand for different quality of services (QoS) [6]. Therefore, in order to efficiently handle future unknown demands such as increased flexibility, capacity and reliability, the implementation problems in ATM network, such as traffic control, routing and bandwidth allocation must be solved [11, 12, 13].

In this paper, based upon the dynamic time slice (DTS) scheme [7], the DTS server provides different buffering for different traffic classes. The server cyclically visits each queue and devotes a period of its time to it. The scheme is used to facilitate dynamic bandwidth allocation and mutual overload protection among the queues. Three different bandwidth allocation strategies are compared. The three bandwidth allocation strategies are static bandwidth allocation, bandwidth allocated proportional to the expected queue length, and the bandwidth allocated proportional to expected queue length with threshold value. Results for static bandwidth allocation and bandwidth allocated proportional to expected queue length are presented by Lu and Hannson [7]. It is seen that bandwidth allocated proportional to expected queue length has better performance than static bandwidth allocation. The purpose of this work is to study the alternative strategy where bandwidth is allocated proportional to the expected queue length with a threshold value, and compare it with other strategies. It is shown that the proposed strategy has better performance than previous two strategies.

2.0 SYSTEM DESCRIPTIONS AND MODEL DEVELOPMENT

To investigate the proposed algorithm and its efficiency, a model of a single link, which represents a single virtual path is considered. Typically, there could be diverse services using this path and so a number of models of sources are needed. This study implements a basic source model, representing some of the services that will be demanded in an ATM network, that is voice, video and data.

The simulation model is made up of three submodules, each of which performs a well define functions. Source submodule is where the ATM cells were generated. The cell stream from a number of sources is then input to the ATM switch buffers submodule. This submodule smoothes the arrival of cells to the ATM network and so takes care of cell scale congestion. This buffer is the limited resource critical to the operation of the model. DTS submodule is where several bandwidth allocation strategies operate and where bandwidth is calculated and allocated to the buffers. The diagram is shown in Fig. 1.

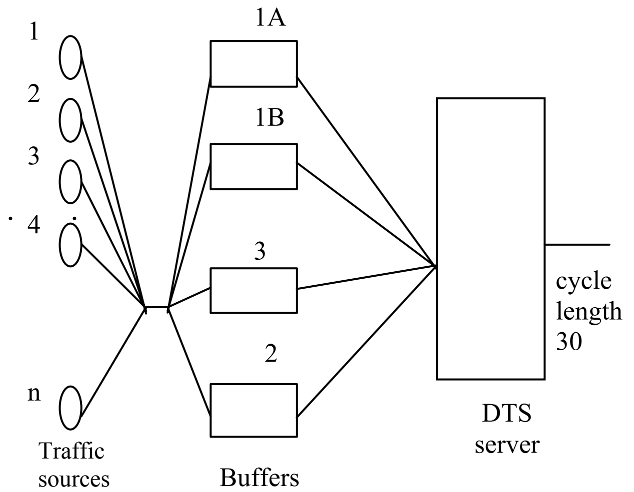

Fig. 1: Simulation model

2.1 Dynamic Time Slice Server Submodule

DTS scheme is based on the principle of having different buffering for different traffic classes [7]. The server cyclically visits each buffer and devotes a period of its time to it. The scheme is used to facilitate dynamic bandwidth allocation and mutual overload protection among the queues. The bandwidth allocation can be chosen to guarantee a bandwidth to a certain traffic class. The sum of the allocated bandwidths on class 1A, 1B and 3 equals the maximum cycle length.

The transmission of cell starts immediately after every buffer has received and stored cells generated by all the sources. The buffer with highest priority will send cells first then only will be followed by lower priority buffer. The maximum DTS cycle length is defined in the server and gives the maximum number of cells that can be transmitted during one cycle. This scheme is dynamic in the sense that any unused bandwidth allocated for buffer 1A, 1B and 3 will be allocated to buffer 2. A new bandwidth allocation is calculated after the transmission.

In this example, the maximum cycle length is set to 30, bandwidth allocated is 10 slots for class 1A, 1B and 3. But 1A-buffer has 9 cells, 1B-buffer 7 cells and 3-buffer 8 cells, thus the total of spare slots would be 6, yielding 6 spare slots available for transmission of type 2 cells (assuming buffer 2 is not emptied). Consequently, the cycle length equals the maximum cycle length (9+7+8+6=30), as shown Fig. 2.

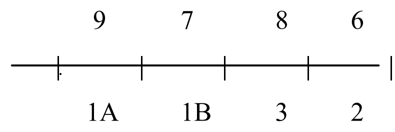

Fig. 2: Number of cells transmitted

2.2 Buffer Management and Requirements

The function of the DTS buffer is to receive and store cells until transmitted. Each of the four traffic classes has its owns buffer. The queues use the first-in-first-out (FIFO) principle. The buffer size could be chosen to achieve an appropriate cell loss in a certain traffic class. However, the type 1A-1B- and 3-buffer sizes are assumed to be the same. In the study, buffer sizes of 25, 30, 35 and 40 will be inspected to determine which buffer size will give the better performance with the lowest cost of implementation. The class 2-buffer is assumed to be infinite.

The highest priority buffer will be class 1A, followed by 1B, 3 and the lowest is class 2. This mean that class 1A buffer will transmit cells first and then will be followed by other buffers according to their priority. Cells assignment to which buffer are fairly distributed. Each buffer has percentage of 0.25 of cells arrival if there are cells generated.

2.3 Source Submodule

This part deals with the selection of traffic source models for ATM networks. It should be noted that by the term model for a traffic source we shall be referring to an algorithm giving the generation time X_i of the ith cell, for i = 1, 2,…., the X_i's will be taken as random variables. There are three types of traffic source models, viz., data, voice and video. However, this study assumes that all the sources are the same and the only model is the basic model as discussed below.

Source characterisation [8, 10] is necessary for the precise identification of the behaviour of each particular source, it does provide network management with the ability to manipulate flexibly various services in terms of connection acceptance, negotiation of the quality of service (QoS), congestion control, traffic enforcement and resource allocation. In ATM networks, there is a general trend to visualise cells generation as a succession of active and silent (also called idle) periods [9]. Cell generation occurs only during active periods; a group of successive cells that are not interrupted by an idle period is called a burst. In the model, the node will generate 1 to 4 cells at a time. A total number of 20 sources are modeled.

A well-known exponential interarrival time distribution, Poisson process, where each time an arrival occurs, the time until the next arrival is obtained by randomly sampling from an exponential distribution
where

$$\mathrm{IA} = \frac{-1\ln(RAND)}{\lambda} \qquad (3.1)$$

IA = interarrival times between successive arrivals,

λ = load of traffic

2.4 Bandwidth Allocated Proportional to Expected Queue Length with Threshold Value

This strategy is based on the measurements of the queue lengths. The point of measuring is after the server has visited it as shown in Fig. 3. This strategy is slightly different from bandwidth allocated proportional to expected queue length strategy. Additional control mechanism is added, if buffer 1A, 1B or 3 exceed certain threshold value (that means congestion has occurred or going to occur) then amount of bandwidth allocated is the threshold value. The remaining bandwidth will be allocated to the other two higher priority buffers proportional to its queue length. Unused bandwidth will become spare slots for the lowest priority buffer that is buffer type-2. If buffer 1A, 1B and 3 do not exceed the threshold value, then the function of the strategy is the same as bandwidth allocated proportional to expected queue length without control mechanism.

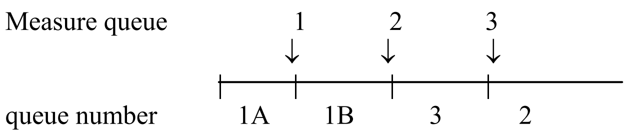

Fig. 3: Measurement of the queue lengths

After the server has calculated the queue length, it will predict the number of cell arrivals for the next cycle in the queue. A proportioned slice of bandwidth is allocated to each queue. The formula for 1A bandwidth allocation is

Bandwidth allocation 1A =

$$\frac{\text{ABSOLUTE ((queue length 1A + expected 1A))}}{\sum \text{queue length(1A, 1B, 3) + expected(1A, 1B, 3)}} \qquad (3.2)$$

where
expected(n) = expected number of cells arriving at buffer n during one cycle
= average cell arrivals in previous 20 cycles

3.0 PERFORMANCE METRICS

There are 2 performance parameters: cell loss ratio and average cell delay. These performance metrics data are the average value of four sets of data. The first 500 cycle performance metric values would be discarded to stabilise the generation, queuing, measurements and transmissions of cells.

Cell loss happens when the buffer is full and a new cell arrives at the particular queue. A queue is full when the number of cells in queue equals the maximum queue length. A cell loss is most likely to happen in a buffer before the server is serving it. Buffer type-2 has no cell loss due to its infinite buffer size. Thus, the total number of cell losses are accumulated from buffer type 1A, 1B and 3.

$$\text{Cell lost ratio} = \frac{\sum \text{Cell loss}}{\sum \text{Arrived cells}} \qquad (3.3)$$

Cell delay is the waiting time of an ATM cell in the buffer to switch through the link or the waiting time of a cell after residents in buffer until transmission. The waiting time is measured in cycle. Buffer type-2 cell delay requirements are not so strict, it may be transmitted during slack period. So, cells from buffer type-2 are excluded from the calculation of cell delay. Total cell delays are accumulated from buffer 1A, 1B and 3.

$$\text{Average cell delay} = \frac{\sum \text{Cell Delay}}{\sum \text{Departed cells}} \qquad (3.4)$$

4.0 RESULTS AND DISCUSSIONS

The model proposed here is evaluated and compared with the two earlier strategies [7]. It may be noted that the appropriate use of buffer size is important. Inappropriate selection of buffer size can seriously affect the performance of the system. Large buffer size increases cell delay and too small buffer size causes serious cell loss.

The input parameters considered are maximum cycle length - 30 cells per cycle; total cycles - 30,000; total sources - 20, 1 to 4 cells generated at a time for each source; traffic load - 0.82, 0.84, 0.86, 0.88, 0.90 and 0.92. Analysis of buffer sizes - 25, 30, 35 and 40 for traffic type-1A, 1B and 3, but infinite buffer size for traffic type-2. Analysis of threshold values - 10, 14, 18, 22 and 26. Output parameter are the performance metrics discussed earlier in Section 3, they are cell loss ratio and average cell delay. The simulation is carried out on a Pentium 200 MHz system using the Turbo C++ compiler. The results are shown in Figs. 4-10.

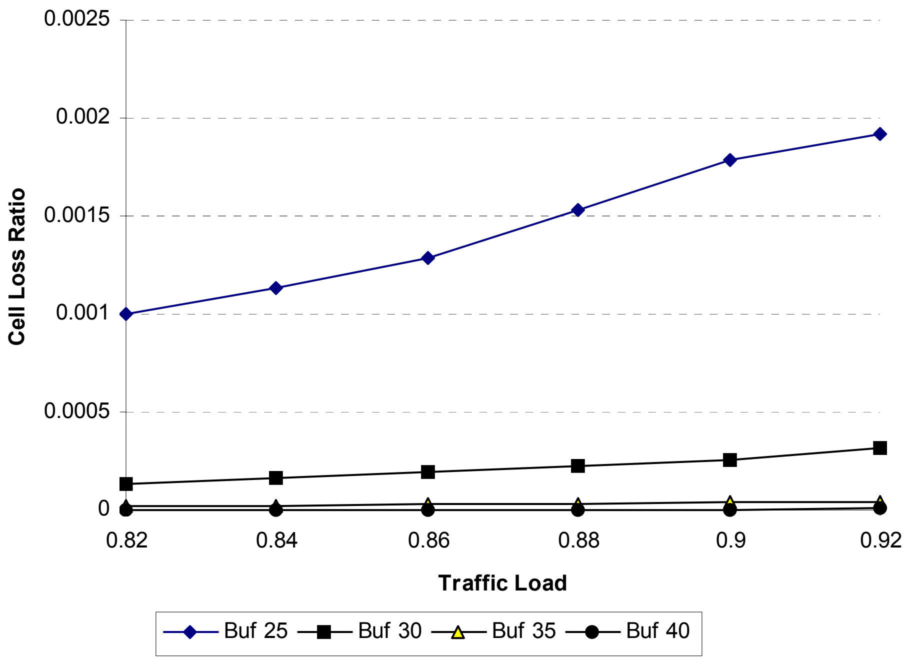

Fig. 4: Cell loss ratio vs. traffic load with different buffer sizes in bandwidth allocated proportional to expected queue length with threshold value 22

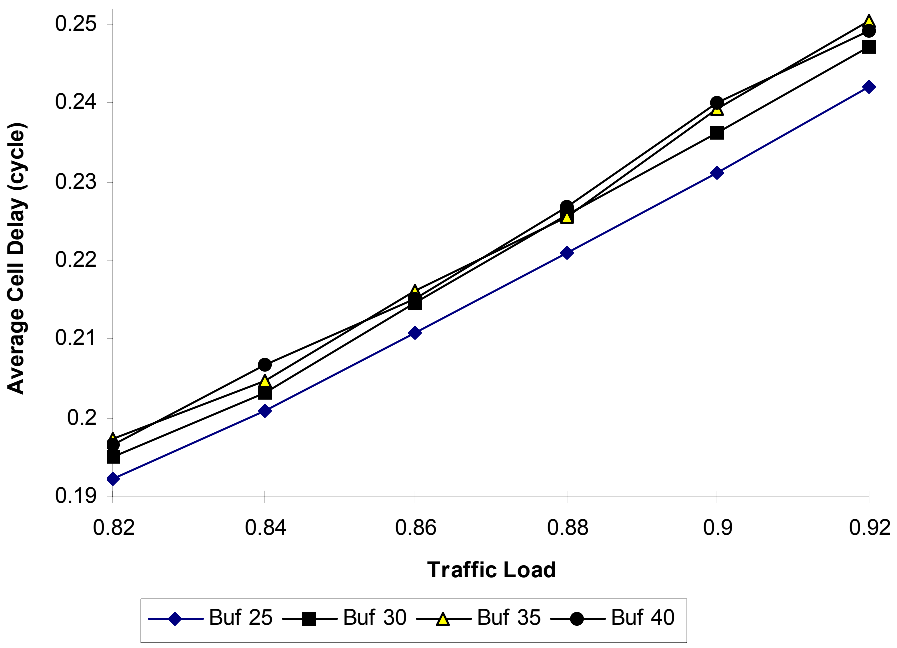

Fig. 5: Average cell delay vs. traffic load with different buffer sizes in bandwidth allocated proportional to expected queue length with threshold value 22

Figs. 4-5 show the cell loss ratio and average cell delay with different buffer sizes (with threshold value 22) in the proposed model bandwidth allocated proportional to expected queue length with threshold value. From Fig. 4, it may be seen that the cell loss ratio is reduced with the increment of buffer size. Buffer size 40 gives the best cell loss performance and buffer size 25 gives the worst . The cell loss performance increases significantly from buffer size 25 to 30, whereas only small performance increment takes place from buffer size 30 to 35, and a very small performance increment from buffer 35 to 40. From Fig. 5, it may be seen that average cell delay is longer when buffer size is large. The increment of cell delay time is significant from buffer size 25 to 30. But there is little difference in cell delay performance from buffer 30 to 40.

It can be seen that the buffer size of 30 is suitable for the proposed model with threshold value 22, as it gives the effective use of resources with the lowest cost of implementation. The effect of different threshold values is seen in Figs. 6-7, which helps us in finding the optimum threshold value.

Fig. 6 shows the cell loss ratio for varying traffic load, and with different threshold values. It can be seen that the threshold value of 22 has the lowest cell loss ratio with steady increment in comparison to the other threshold values. Fig. 7 shows the delay performance for threshold values 10, 14, 18, 22 and 26. The result suggests that the threshold value should be greater than 22. Therefore, considering both cell loss and delay performance, threshold value of 22 is the most suitable value for the proposed model with buffer size of 30, to have best performance in cell loss and average cell delay.

Further, using threshold value of 22 and buffer size of 30, the performances comparison among the proposed model and Lu and Hannson's [7] models are presented. The proposed strategy is compared with the two strategies, the static bandwidth allocation and bandwidth allocated proportional to expected queue length [7].

Fig. 8 compares the cell loss ratio performance between the proposed strategy and the static bandwidth allocation [7]. From the graph it can be seen that as the traffic load increases, cell loss ratio also increases for both the models, but for the proposed strategy the cell loss ratio remains significantly lower. It can be clearly seen that the proposed strategy works better than the static allocation strategy. This is expected since the bandwidth is allocated according to the expected queue lengths of the next cycle. If the buffer is congested or going to occur in next cycle, then the bandwidth allocated will be greater.

Fig. 9 compares the bandwidth allocated to the expected queue length with the proposed strategy, having threshold value of 22. Here also the proposed strategy has slightly better performance because when the expected queue length exceeds or equals certain threshold value, then the bandwidth assumes that threshold value for the particular buffer. The threshold value of 22 is properly selected to give efficiency and effective use of buffer. If the queue length is predicted to arrive at the threshold value, then the particular buffer is going to be congested and needed instant attention. This certainly will solve congestion problem for that buffer and reduce cell loss ratio. Similar observation is made for average cell delay shown for the three cases in Fig. 10.

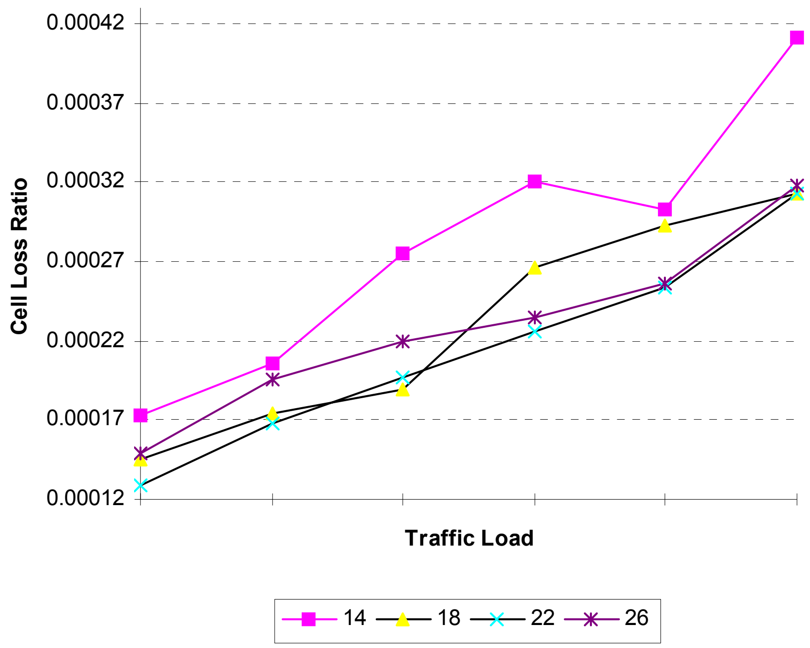

Fig. 6: Cell loss ratio vs. traffic load in bandwidth allocated proportional to expected queue length with different threshold values (Buffer size 30)

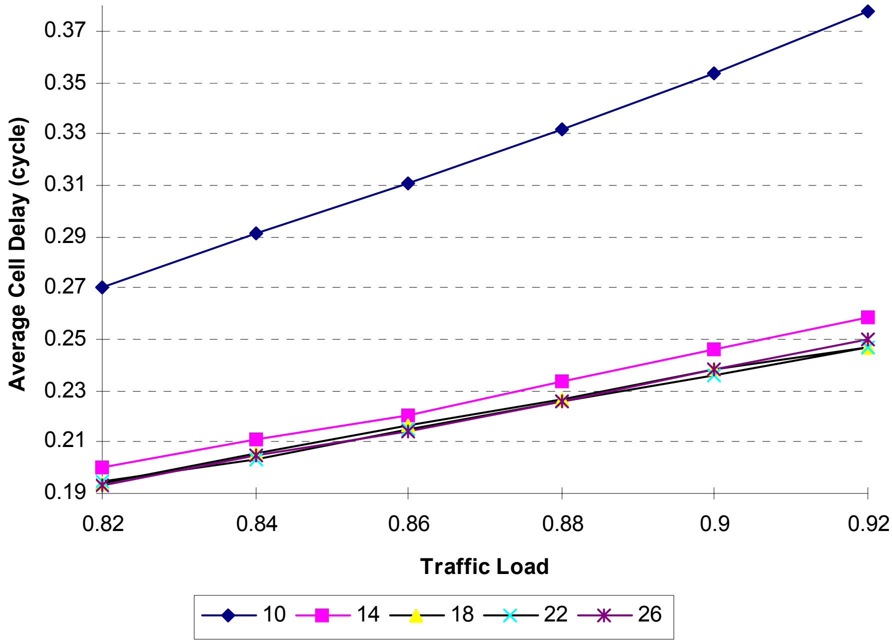

Fig. 7: Average cell delay vs. traffic load in bandwidth allocated proportional to expected queue length with different threshold values (Buffer size 30)

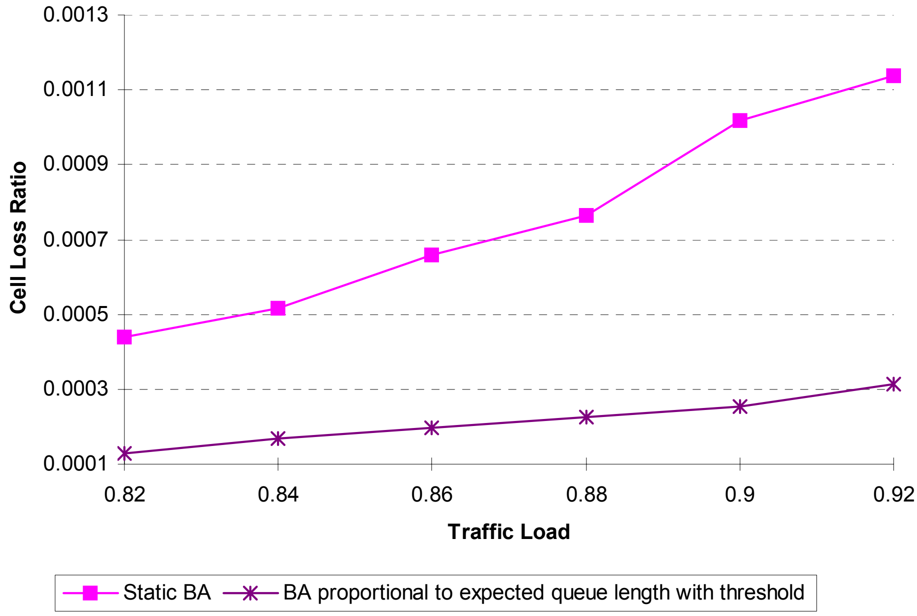

Fig. 8: Cell loss ratio vs. traffic load for static BA and BA proportional to expected queue length with threshold value of 22 (Buffer size 30)

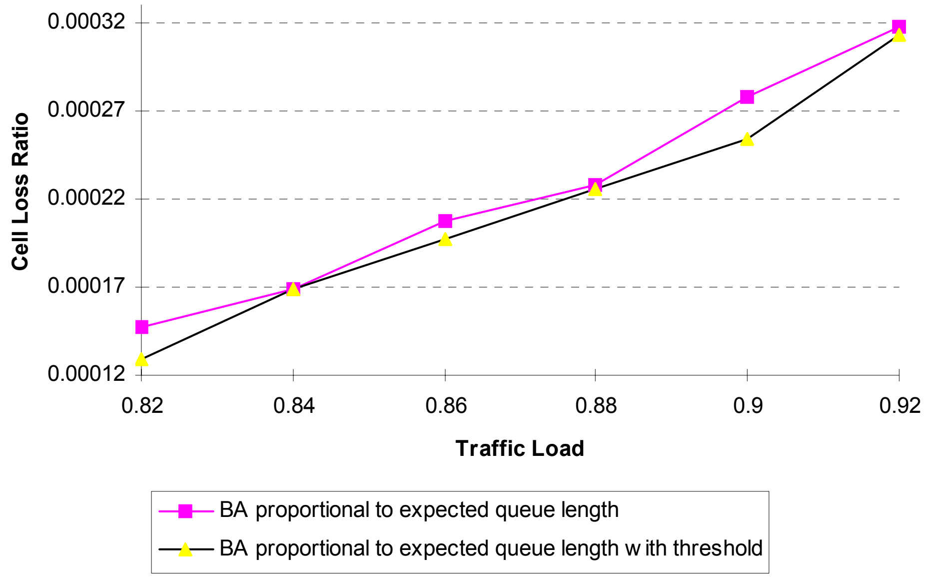

Fig. 9: Cell loss ratio vs. traffic load for static BA and BA proportional to expected queue length with threshold value of 22 (Buffer size 30)

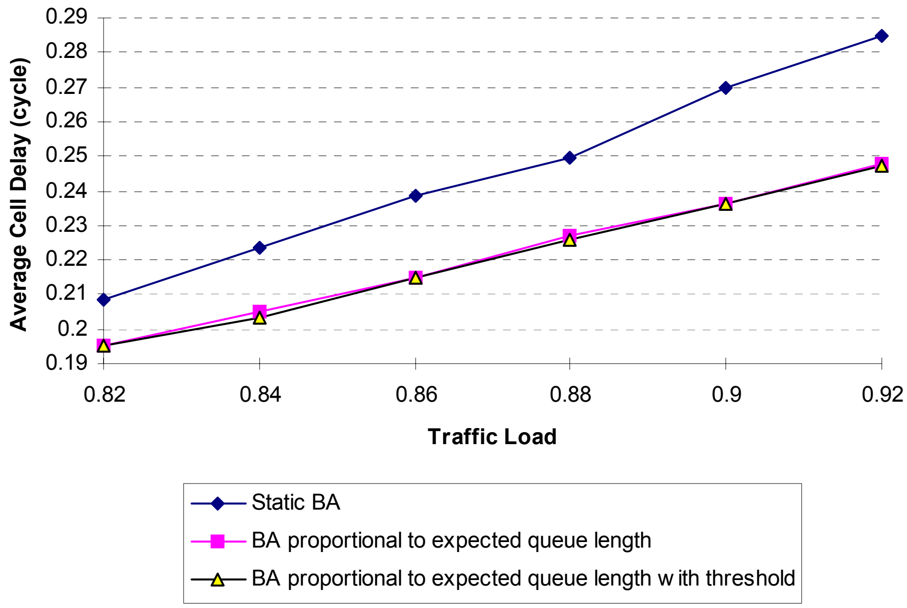

Fig. 10: Average cell delay vs. traffic load for static bandwidth allocation, bandwidth allocated proportional to expected queue length and bandwidth allocated proportional to expected queue length with threshold value 22

From the graph it can be seen that as traffic load increases, average cell delay also increases in all three cases. This happens because when traffic load increases, more cells arrive and queue in the buffer. When queue length is increasing, cell residents in the buffer will wait longer in the buffer before transmission. If the buffer is heavily congested, then cells in that congested queue suffer the maximum waiting for transmission and increase in the average cell delay. The strategy-2 of Lu and Hannson [7] and the proposed strategy show better result due to the fact that they are effective to handle bandwidth allocated for high priority buffers. The strategy-2 and the proposed strategy give a comparatively small differences in average cell delay performance, and the proposed strategy works slightly better.

From the results discussed, it can be concluded that the proposed strategy, which allocates the bandwidth proportional to expected queue length with a threshold value, for the next DTS cycle, provides a better solution for congestion control. As in other strategies also, this strategy will reduce the performance in the low priority queue, where buffer is unlimited in size but the traffic is independent of delay.

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BIOGRAPHY

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