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Evaluation of Performance of an ATM Multiplexer Carrying Multimedia Traffic

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Abstract. This paper examines the performance of an ATM multiplexer loaded with multimedia traffic. A simulation model has been developed to evaluate the multiplexer performance. Traffic generated from multimedia traffic sources having different burstiness factors is fed to the multiplexer input, to evaluate the cell loss and cell delay performance of the multiplexer and also to evaluate the output ATM trunk utilization. The simulation results are compared with approximate analytical results. Moreover an outline has been given for the acceptance of new calls depending on the source burstiness factor and peak source rate to avoid cell loss and excessive delays at the multiplexer.

1. Introduction

In the near future broadband networks will support a wide range of services ranging from low speed data to full-motion video. A transport technology based on asynchronous transfer mode (ATM) is widely recognized as the target technology for future Broadband Integrated Services Digital Networks (B-ISDN) [1,2]. The B-ISDN is based on a fully optical transmission network operating at a speed of 155 Mb/s at the user network interface (UNI) and at 622 Mb/s in the network. It offers very fast and efficient connection-oriented service i.e. setup and clearing of the virtual connections, and packet (cell) transfer between ATM access ports (nodes). ATM is a fast packet switching technique based on virtual connections (VC) using small fixed size packets (53 bytes) called cells. An ATM cell carries a five octet header carrying information, such as generic flow control (GFC), a virtual channel and a virtual path identifier (VCI, VPI), payload type and cell loss priority indications (PT, CLP) and one octet header error control field (HEC) [1]. The 4 bit GFC field provides flow control at the user-network interface (UNI) for the traffic originated at user equipment and directed to the network. As the GFC field has no use within the network, this field has been removed from the NNI cell format. Fig. 1(a, b) shows the

ATM cell format for user-network interface (UNI) and network-network interface (NNI). Among other things, the major advantages of ATM include (i) flexibility in accommodating existing and future applications, (ii) simple multiplexing and switching, (iii) potential for statistical multiplexing to save bandwidth. Statistical multiplexing exploits the variance of source statistics to achieve an efficiency gain by serving a number of sources where combined peak rate is greater than the output link capacity. Since the link capacity is highly utilized, there is a potential for serious congestion problems and packet loss at high loads. Most of the ATM traffic is bursty in nature, which generates cells at or near its peak rate for a short period followed by longer periods of low or no generation at all. These characteristics of the packetized bursty traffic complicates the congestion problem and if necessary actions are not taken at the entry point, serious congestion may arise inside the network, causing excessive delays and loss of cells. Once congestion occurs, reactive flow control mechanisms such as window flow control cannot be employed due to very high speed transmission of cells, as a very large number of cells for the same VC are already in transit. The only solution might be a preventive control at the network edges *i.e.* at the ATM node where a new call should be accepted if the required grade of service (GOS) can be assured.

<table>
<tr><th>BITS →
BYTES
↓</th><th>1 ... 4</th><th colspan="2">5 ... 8</th></tr>
<tr><td>1</td><td>GFC</td><td colspan="2">VPI</td></tr>
<tr><td>2</td><td>VPI</td><td colspan="2">VCI</td></tr>
<tr><td>3</td><td colspan="3">VCI</td></tr>
<tr><td>4</td><td>VCI</td><td>PT</td><td>CLP</td></tr>
<tr><td>5</td><td colspan="3">HEC</td></tr>
<tr><td>6 - 53</td><td colspan="3">INFORMATION FIELD</td></tr>
</table>

Fig. 1 a. ATM cell format for user-network interface (UNI).

<table>
<tr><th>BITS →
BYTES
↓</th><th>1 ... 4</th><th colspan="2">5 ... 8</th></tr>
<tr><td>1</td><td colspan="3">VPI</td></tr>
<tr><td>2</td><td>VPI</td><td colspan="2">VCI</td></tr>
<tr><td>3</td><td colspan="3">VCI</td></tr>
<tr><td>4</td><td>VCI</td><td>PT</td><td>CLP</td></tr>
<tr><td>5</td><td colspan="3">HEC</td></tr>
<tr><td>6 - 53</td><td colspan="3">INFORMATION FIELD</td></tr>
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Fig. 1 b. ATM cell format for network-network interface (NNI).

A bursty source can be characterized by three main parameters; peak bit rate, average bit rate and duration of the burst. The bit rate process, resulting from the superposition of a number of voice or video sources, exhibits very high fluctuations over short time periods. This leads to major congestion problems unless an appropriate flow control scheme is maintained.

The aggregate packet arrival process, which is nonrenewal in nature, can be approximated by suitable mathematical model such as Markov modulated Poisson process (MMPP). The analysis of this queuing structure involves the solution of GI/D/1 queues with overload control. An exact mathematical analysis of such types of queues is intractable and approximations are necessary [3-5]. The approximations often limit the scope of the results, thus rendering simulation tools more powerful and accurate to use. Sometimes mathematical analysis requires the solution of a large number of simultaneous equations satisfying a set of boundary conditions [5-7]. It may require the solution of large stochastic matrices using numerical techniques and consuming large CPU time [8,9].

In this paper we shall use mainly the event driven simulation technique to study the performance of an ATM multiplexer having input traffic from different traffic sources having different burstiness factors. Section II describes the input data generation schemes and the parameters for different types of data sources. In section III we provide some analytical means based on GI/GI/1 queuing models those can be used for the situation simulated. In section IV, we present the simulation model where the multiplexer and the output trunk is modeled to accept cells from the input sources and transmitting them at the ATM link. The cell loss statistics and buffer occupancy figures are collected. The input traffic streams are varied to see the effect of different traffic streams on the ATM node performance. In section V the simulation and analytical results are presented. In section VI we propose a traffic admission control policy to avoid cell loss and excessive delays in the ATM multiplexer. We explain, with examples, the procedure for acceptance of new calls depending on the admission control policy so that congestion can be avoided. Section VII concludes this paper.

2. ATM Traffic Model

To achieve a good bandwidth utilization, statistical multiplexing is preferable at ATM nodes, where the sum of the connection peak rates is allowed to exceed the link rate. However, a simple traffic control mechanism is preferable with guaranteed grade of service (GOS) requirements. ATM traffic such as voice, video and image data are bursty in nature. An ATM traffic source can be characterized at call, burst and cell levels. The traffic model capture the statistical behavior at all levels of traffic characteristics.

A two state Morkovian representation of an ATM source has been suggested for many applications [3,10,11]. Three parameters are required to fully describe the traffic

$$\lambda = 1/(T+\alpha T/\beta) \tag{1}$$

where T is the packet (cell) interarrival time. Thus the mean number of arrivals, in time t is

$$M_1(t) = \lambda t = t/(T+\alpha T/\beta) \tag{2}$$

The squared coefficient of variation C_a^2 for the arrivals is given in [3] as

$$C_a^2 = (1-(1-\alpha T)^2\,)\,/\,(\,\alpha T+\beta T)^2 \tag{3}$$

If n identical independent ON-OFF sources are superimposed and $N_i(0, t)$ denotes the number of cell arrivals in (0, t) from the ith stream, then the aggregate cell arrivals from the superposition is given by

$$N^S(0,t) = \sum_{i=1}^{n} N_i(0,t)$$

where S represents superimposed sources.

So $$M_1^S(t) = E[N^S(0,t)] = nM_1(t) \tag{4}$$

and also according to [3]

$$C_a^2 = \lim_{t \to \infty} \operatorname{Var} N^S(0,t)/(E[N^S(0,t)] = \lim_{t \to \infty} \operatorname{Var}[N(0,t)]/E[N(0,t)]$$

$$= \operatorname{Var}[X]/E^2[X] = (1-(1-\alpha T)^2\,)\,/\,(\alpha T+\beta T)^2 \tag{5}$$

3. Approximate Queuing Analysis

The superposition of ON-OFF sources in a statistical multiplexer produces a nonrenewal arrival process that leads to the analysis of GI/GI/1 queuing models. As there is no known exact solution to the above queuing model, we need to take the help of available approximate solutions for the above problem. As ATM cells are of fixed size, the queuing model becomes GI/D/1 type. Kobayashi[12] derived the queuing delay W for GI/GI/1 cases using the equation

$$W = \rho_T/\mu(1-\rho_T) \tag{6}$$

where

$$\rho_T = \exp(\,-2(1-\rho)/\rho(C_a^2+C_S^2/\rho))\,; \quad \{\,\rho = \lambda/\mu\}$$

For fixed service time ρ_T becomes

ρ_T =exp(-2(1-ρ)/ρC_a^2) as the squared coefficient of variation , C_S^2 =0 for fixed service time.

According to [13-14] the waiting time W for the GI/D/1 case reduces to

$$W= \rho C_a^2 /(2\mu(1-\rho) \tag{7}$$

We shall use both equations 6 and 7 to compare our simulation results later.
The total queuing delay of a cell in an ATM multiplexer with several input queues and a single output link, comprises two components of the delay W_1 and W_2 . Where W_1 is the delay experienced by a cell while the server serves cells from other input sources and W_2 is the delay experienced by a cell while cells (ahead) in the same queue are served. According to [15] the delay W_1 is given by

$$W_1 = Mw(1-\rho/M)/2(1-\rho) \tag{8}$$

where M is the number of input sources and w is a fixed walk time and ρ is the throughput of output the channel. The delay component W_2 can be modeled with the GI/D/1 queuing model and equation (6) or equation (7) above can be used. Thus the total queuing delay can be written as

$$W= W_1 +W_2 = Mw(1-\rho/M) / 2(1-\rho) + W_{GI/D/1} \tag{9}$$

With the assumption of zero walk time (w=0), the queuing delay W becomes equal to $W_{GI/D/1}$ (W_1 =0) i.e. W= W2 = $W_{GI/D/1}$.

4. Simulation Model

Simulation models remain the most flexible means for performance evaluation under a variety of conditions and are also necessary for validation of analytical approximations. The discrete event simulation procedure has been applied to study the performance of our multimedia system having traffic input from different data sources and feeding to a central switch connected to a high speed output link. The link capacity is chosen as 155 Mb/sec and data packets of size 53 bytes, called as cells are transmitted to the output link if available from the input traffic sources. The service policy is round robin and individual queues are provided for storing cells arriving from each data source. We have studied the ON-OFF type of sources, where each source has an ON period when it generates cells at the peak rate and then goes to idle for the OFF period to become ON again later. This cycle continues.

The ON and OFF periods are exponentially distributed and the durations of ON and OFF periods depend on the nature of the source. A simplified flow chart of the simulation model is shown in Fig. 3.

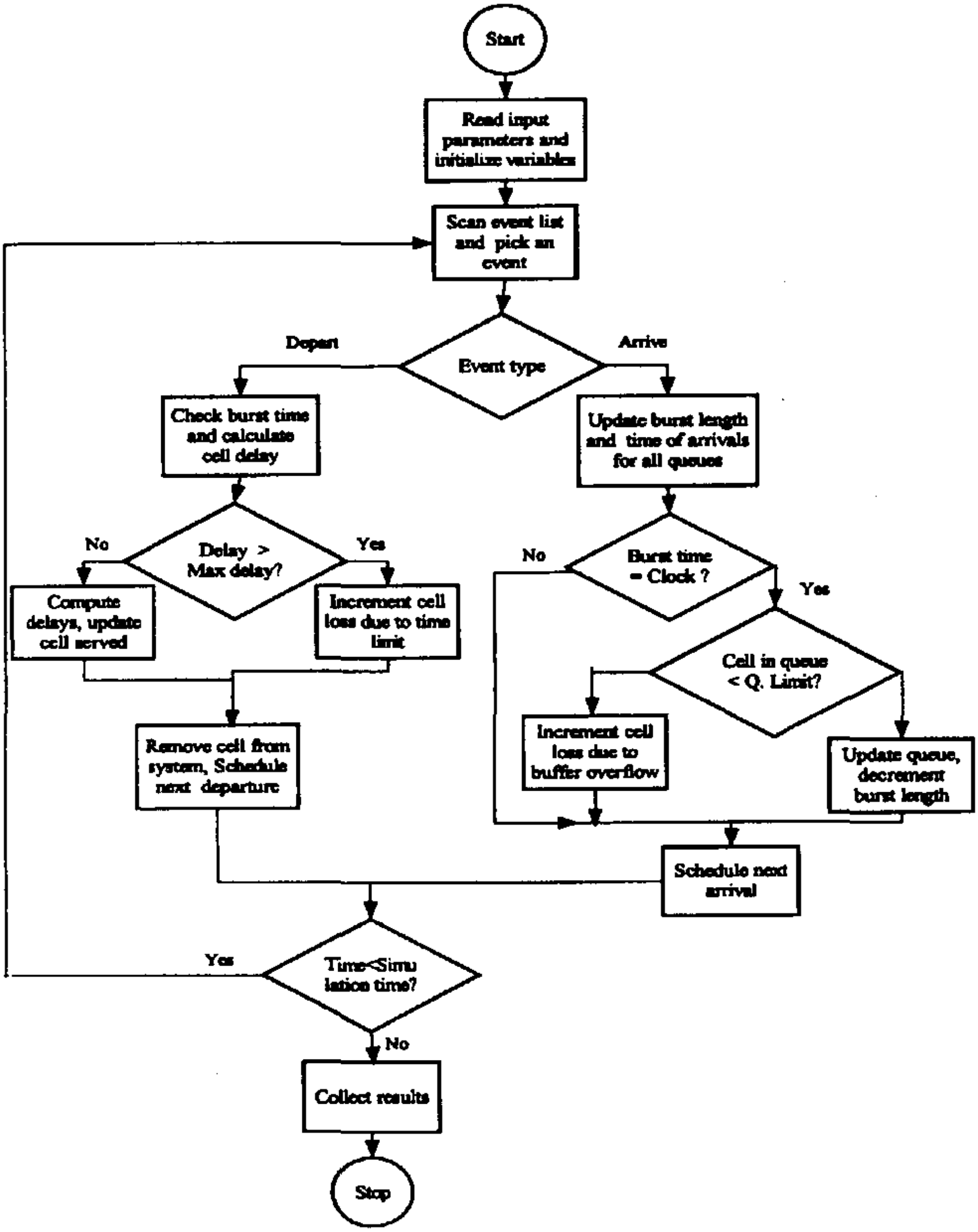

Fig. 3 . A simplified flowchart of the simulation model.

Input to the simulation model

The following parameters are given as input to the simulation model, where these parameters take appropriate values depending on the source type.

B = Mean burst length in cells/burst,
P = Peak bit rate at ON period,
P/L = Peak to output link rate for the source,
P/A = Source burstiness factor.

Thus we can compute the mean duration of ON and OFF periods as below.

h = 53*8*(B/P) = burst duration in seconds,
Let k = Mean duration of the OFF period.

Then k can be calculated from h, P and A as below.

$$k = (hP/A) - h \tag{10}$$

The mean duration of an ON-OFF cycle is (h+k).

The squared coefficient of variation C_a^2 in terms of h , k and the burst length B can be written as

$$C_a^2 = (2B-1)/(1+h/k)^2 \tag{11}$$

Assumptions

The ATM node receives cells from N input sources where traffic generated from the sources are independent of one another. A buffer size of 400 cells is provided at each incoming line. Cells delayed for over 400 cell transmission times (nearly 1.1 ms) are marked as lost cells due to excessive delay and removed from the buffer. Each simulation run can be extended to over 5 seconds in case of heavy traffic and for much longer periods in cases of low traffic. Low traffic cases are those where source burstiness is very high (10 or more).

5. Results and Discussion

The simulation results of interest are output link utilization versus normalized mean delay, cell loss versus output link utilization and output link utilization versus traffic sources multiplexed. Results were collected for homogenous multiplex of traffic and heterogeneous multiplex of traffic sources. Depending on source peak bit rates (P_i), average to peak ratios (A_i/P_i), and number of sources multiplexed (N), the above results are presented. The number of input sources multiplexed depend on the source type and traffic parameters. Unless otherwise stated a moderate mean burst length of 100 cells/burst is considered for the results presented in the following subsection.

We run the simulation for three different traffic sources having different burstiness factors. The source burstiness factor (P/A) has been chosen from a very low value to a very high value. We have chosen sources with low, medium and high burstiness factors. As A./P, the source utilization is the inverse of burstiness (P/A), we categorized sources as low, medium or highly bursty according to the following convention to run the simulation program and to produce the output results.

Low burst sources: A/P from 0.7 -1.0
Medium burst sources: A/P from 0.2 - 0.5
High burst sources: A/P from 0.01 - 0.05.

Sources with low burstiness factor

As low burst sources have higher A/P ratios, a higher output link utilization is obtained with moderate delay and low cell loss rate. Fig. 4 shows that we can reach an output link utilization of over 70% with almost negligible delay. The cell loss is also negligible for an output link utilization of up to 90%. Fig. 5 shows the packet loss pattern.

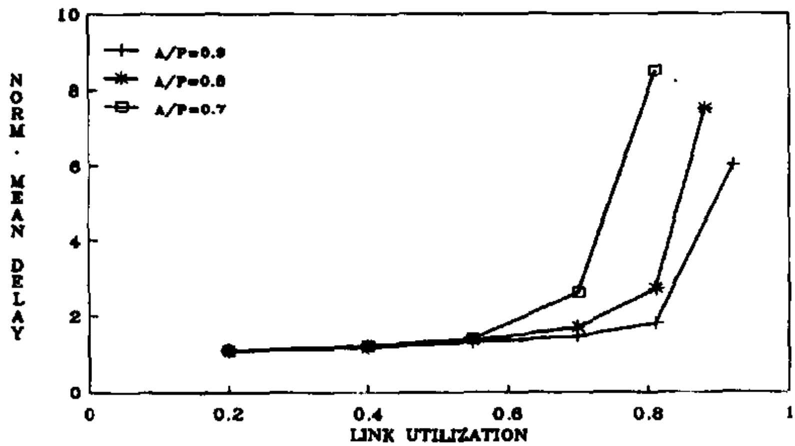

Fig. 4. Output link utilization versus delay for low-burst sources.

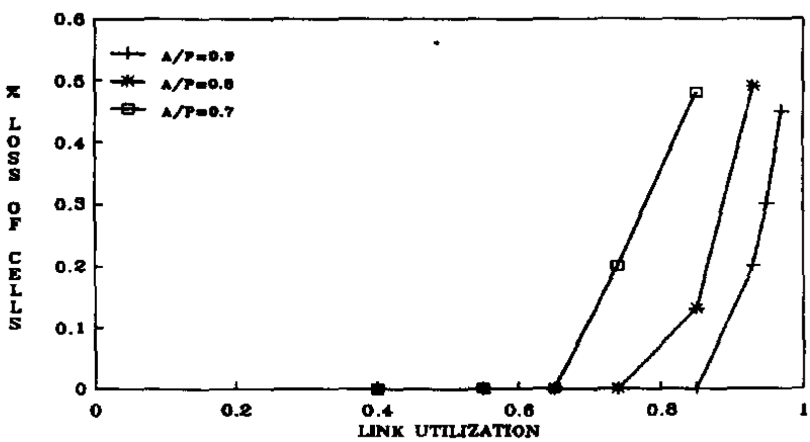

Fig. 5. Output link utilization versus % cell loss for low-burst sources.

Sources with medium burstiness factor

For medium burst sources multiplexed, we have shown delay versus utilization curves for A/P=0.2 and A/P=0.5. Fig. 6 shows the delay versus output link utilization for sources with A/P =0.2 and 0.5 respectively. The output link utilization is obtained by continuously loading the channel by adding more input sources to the multiplexer. Fig. 7 shows the packet loss pattern for these cases. An output link utilization of 33% is reached for A/P=0.2 and 62% for A/P=0.5.

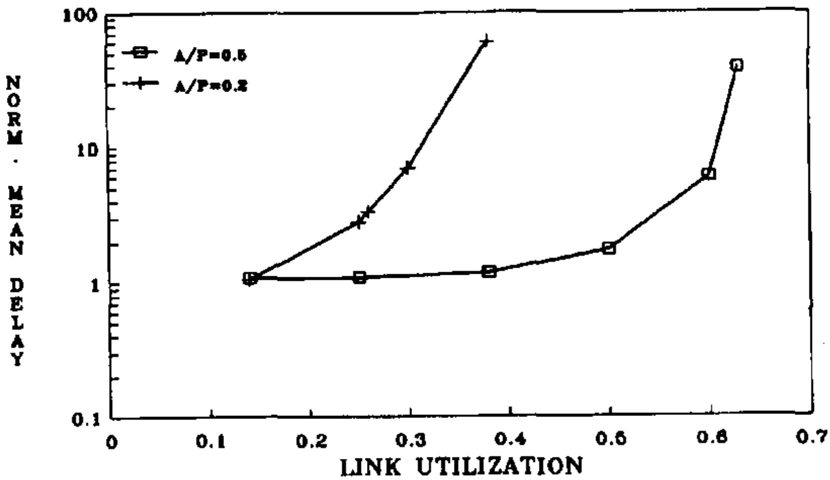

Fig. 6. Output link utilization versus delay for medium-burst sources.

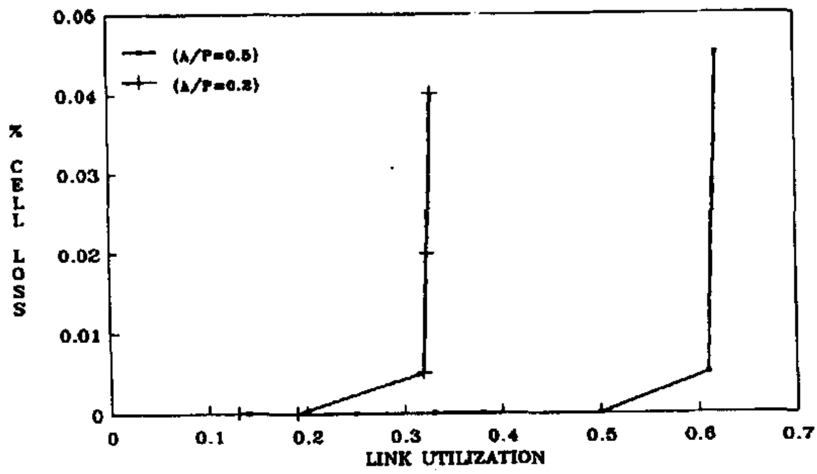

Fig. 7. Output link utilization versus % cell loss for medium-burst sources.

Sources with high burstiness factor

We have simulated the cases of A/P = 0.01 and A/P = 0.05 . The results are shown for these cases in figures 8 and 9. In Fig. 8 the link utilization versus delay curves are shown. The output link utilization remains very low under the cell loss constraints. The cell loss rate rises sharply at output link utilization as low as 0.04 for A/P=0.01 and at 0.09 at A/P = 0.05, respectively. Fig. 9 shows the cell loss rate. It was noticed that under the same operating conditions a longer burst length causes higher percentage of cell loss and lower link utilization.

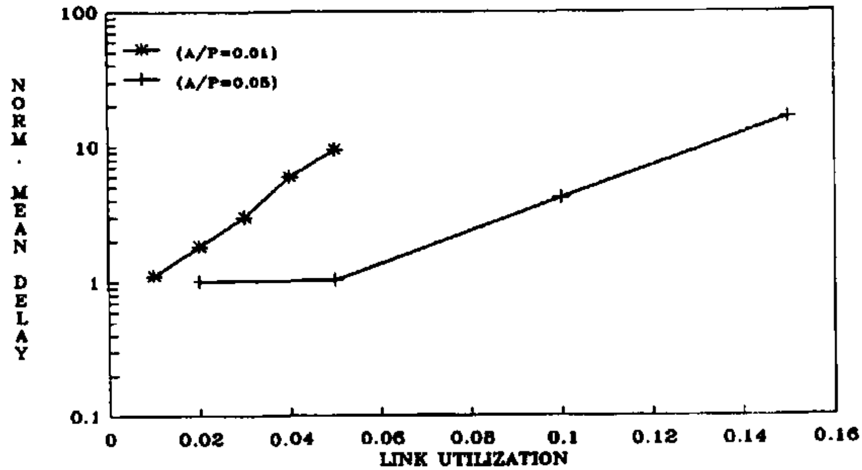

Fig. 8. Output link utilization versus delay for highly bursty sources.

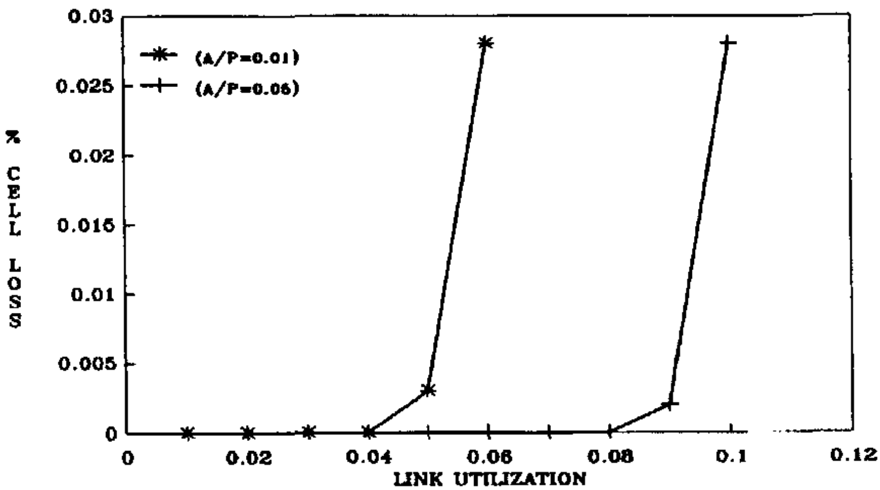

Fig. 9. Output link utilization versus % cell loss for highly bursty sources.

Mixed sources

Heterogeneous input sources are multiplexed in obtaining the results shown in figures 10 and 11. Ten input sources are multiplexed with different burstiness factors. Moderate link utilization is obtained for all the combinations. Different A/P values are set for different source types. The multiplexer input is slowly increased to load the output link as shown in Fig.10 for different combinations of the input sources. It has been observed that the highest loss occurs in case of low-burst sources while least loss occurs in case of highly bursty sources when different sources are multiplexed.

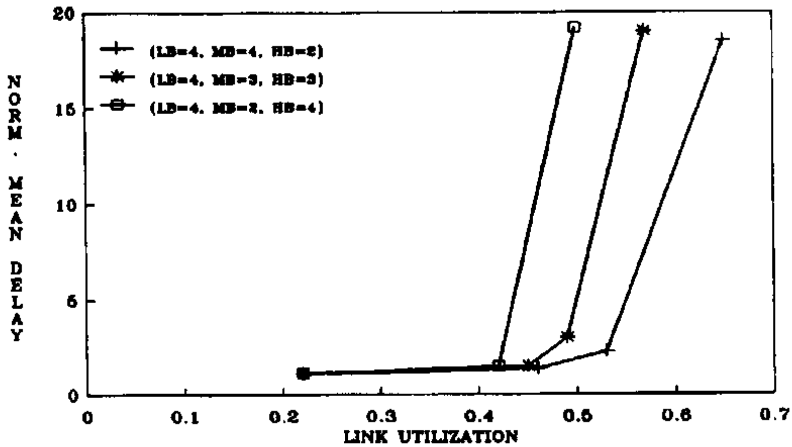

Fig. 10. Link utilization versus delay for mixed sources (Low-bur, A/P=0.9; med-bur, A/P=0.2; high-bur, A/P=0.05).

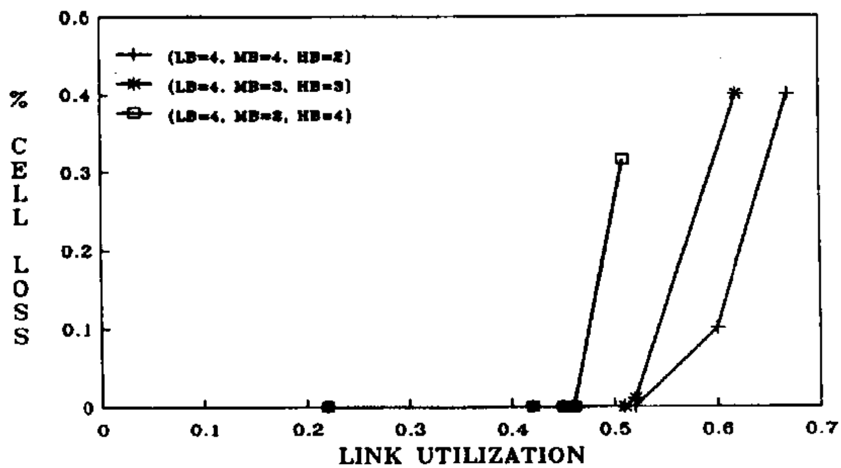

Fig. 11. Link utilization versus % cell loss for mixed sources (Low-bur, A/P=0.9; med-bur, A/P=0.2; high-bur, A/P=0.05).

Figures 12-16 provide a comparative evaluation of performance between analytical means and simulation. We used equations 6-7 to evaluate the queuing delay using the GI/D/1 approach and compared with the simulation results for sources of different burstiness and hence different squared coefficient of variation C_a^2. It is clear from the figures that the approximate queuing equations agree well with the simulation results at heavy load while at low offered load there is a difference between the results. It is also observed that for the mean burst length (B) of up to 30 cells/burst the simulation and analytical results agree quite well for the range of A/P ratios from 0.1 to 0.9. For longer burst lengths with A/P ratios less than 0.7 the results differ and analytical delays computed using equations 6 - 7 are found to be always higher than the delay obtained by simulation. The reason for this discrepancy is mainly due to a very high value of C_a^2 as computed by using equation 11. The squared coefficient of variation as computed by equation 11 does not truly represent the true variation of the arrivals. The use of a bias factor to scale down the value of coefficient of variation has been proposed in [20]. Further investigation is necessary to determine a heuristic base for the evaluation of a suitable formula to accommodate the whole range of ATM traffic.

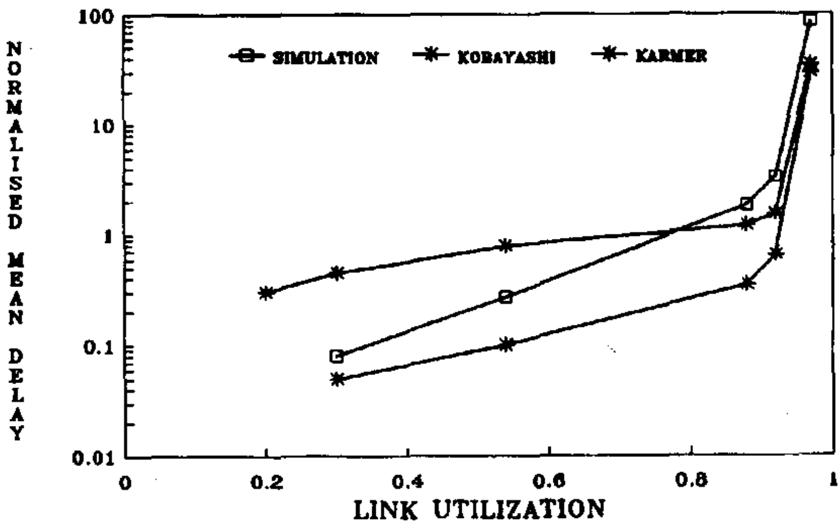

Fig. 12. Output link utilization versus delay (A/P=0.9, B=10, SQ. CV.=0.19).

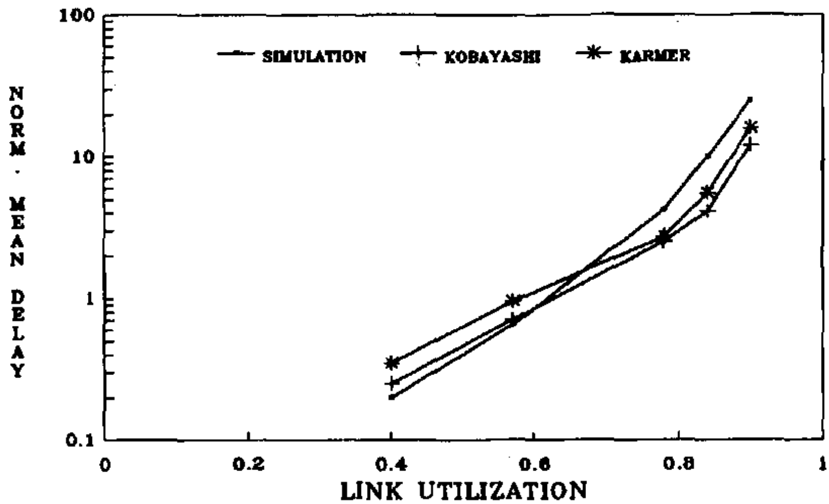

Fig. 13. Output link utilization versus delay (A/P=0.7, B=10, SQ. CV.=0.71).

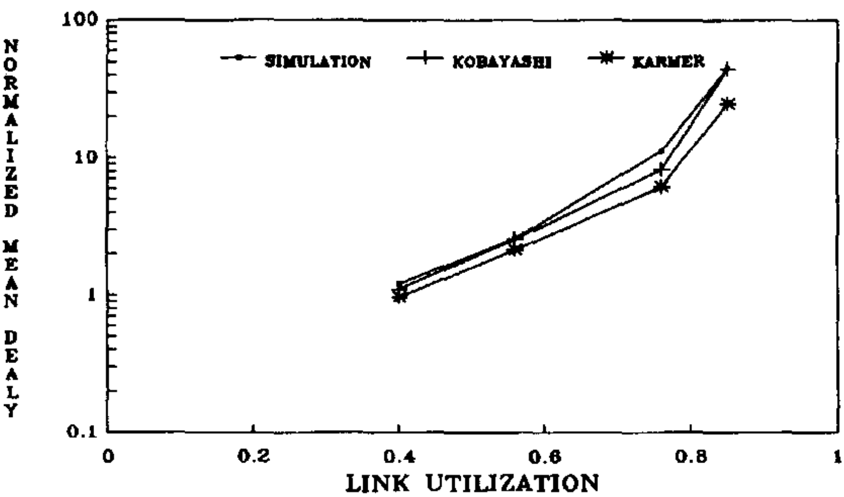

Fig. 14. Output link utilization versus delay (A/P=0.5, B=10, SQ. CV.=4.75).

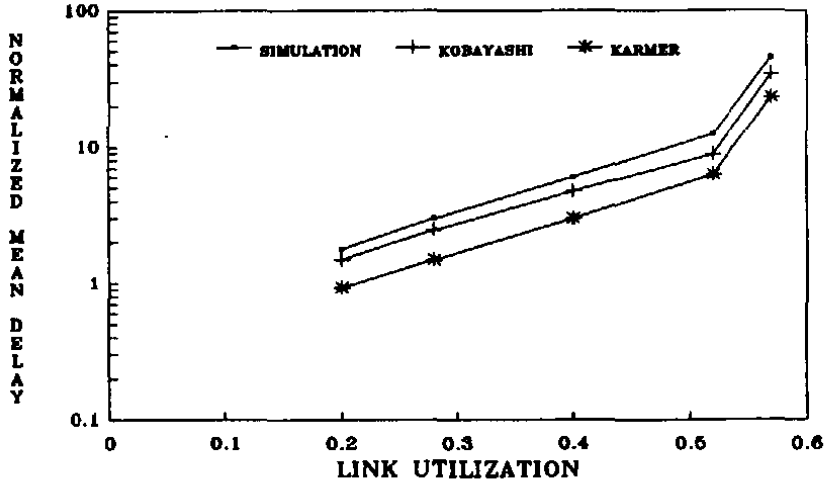

Fig. 15. Output link utilization versus delay (A/P=0.1, B=10, SQ. CV.=15.39).

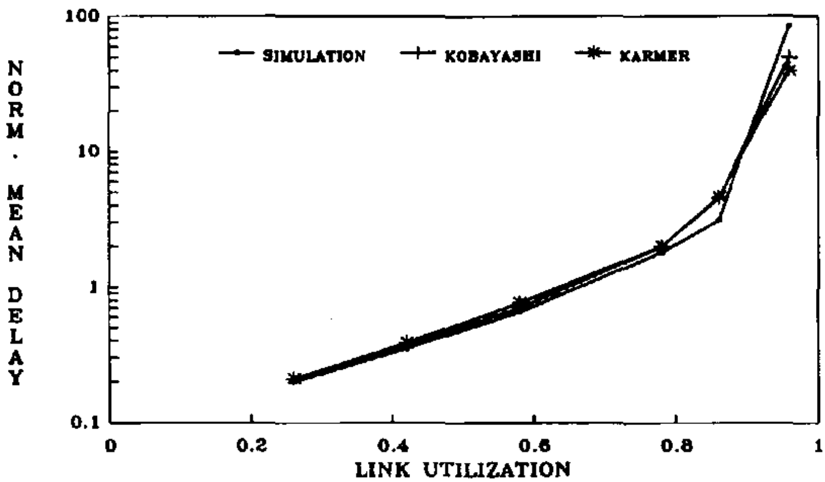

Fig. 16. Output link utilization versus delay (A/P=0.9, B=100, SQ. CV.=1.99).

6. Admission Control Policy to Avoid Cell Loss

Due to a very high speed transmission of cells in ATM links, reactive flow control mechanism is not recommended for ATM as it can cause severe congestion problems before an action is taken [2]. Preventive flow control is the best solution to limit the flow of cells in the ATM links. This can be implemented at the entry point by blocking new call admissions if the required GOS cannot be guaranteed for the accepted calls as well as the ongoing calls.

We have studied the ATM multiplexer characteristic in the previous section for different types of traffic sources. It is noticed that sources having high A/P ratio (less bursty) offers higher output link utilization under the same delay and loss constraints. Sources having very low A/P ratio (more bursty) perform the worst. In case of low-burst sources an output link utilization of up to 90% can be attained where the total peak input rate can reach up to 180 MB/sec.

In case of medium-burst sources the output link utilization reaches 33% for A/P = 0.2 and 62% for A/P = 0.5 respectively. Total peak input reaches 270 Mb/sec for A/P =0.2 and 195 Mb/sec for A/P =0.5.

Sources with A/P =0.01 gives an output link utilization of only 4% under the given delay and loss constraints. It reaches up to 9% for A/P =0.05. The total peak rate is 570 Mb/sec for A/P = 0.01 and 295 Mb/sec for an A/P =0.05.

From our studies of the ATM multiplexer we can adapt a guideline for accepting new calls to maintain a desired grade of service at the ATM node. Fig. 17 shows the operating region (A+B) for an ATM multiplexer loaded with multimedia traffic. Region A is the non statistical operating region, where the total peak input rate can never exceed the output link capacity (L). Region B is the statistical operating region where the overall input data rate can exceed the output link capacity and hence can accommodate more calls simultaneously. To guarantee the required GOS and to avoid congestion we must not cross the boundary of the operating region i.e. area (A+B) is the operating region.

As for an example, if we consider the case of homogeneous traffic with an A/P=0.2 and operating in the statistical mode, the number of sources that can be multiplexed can be computed from Fig.17, if the peak source rate is available. For A/P=0.2 we can read Q*P=220 Mb/sec (the ordinate) from curve B in Fig.17. For P=20 Mb/sec. we can accommodate up to 11 video sources (Q=11) and can operate safely within the operating region. In case of highly bursty sources with A/P=0.05 and P=30 Mb/sec, Q*P=300 Mb/sec (from curve B, Fig. 17), which gives Q=300/30=10 i.e. 10 image sources can be multiplexed.

Figure 17 can also be used in case of heterogeneous multiplexing. In this case a knowledge of the number of ongoing calls of each type can provide us the information about the used BW and the rest can be assigned to the incoming call if sufficient to meet the demand.

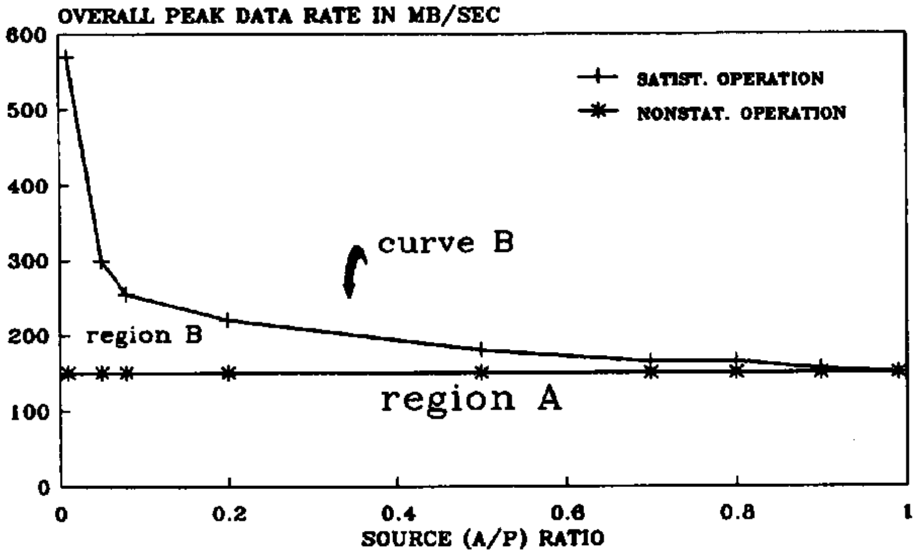

Fig. 17. Source A/P ratio versus combined peak data rate for different sources.

For an example of heterogeneous multiplexing, let us take the case of multiplexing medium and highly bursty sources at the same ATM node. Our problem is to decide about the acceptance of a call from an highly bursty source if there are 7 medium burst sources, each having P=20 Mb/sec and A/P=0.2, already in service from the same ATM node. To decide about the acceptance of the call from a source having P=30 Mb/sec and A/P=0.05, we need to calculate the total BW occupied by the medium burst sources and hence to find out the available BW for the incoming call. We enter Fig. 17 with an A/P=0.2 and read the overall peak data rate for the video multiplexing i.e. 220 Mb/sec (the ordinate). As for L=150 Mb/sec (out trunk capacity) there is a multiplexer gain factor of 220/150=1.466 for the multiplexed sources, the available BW becomes 150 - (7*20)/1.466=54.50 Mb/sec. As the requested call, from the highly bursty source, has a peak rate of 30 Mb/sec, we can easily accept the new call set up request.

7. Conclusion

We have simulated an ATM node accepting cells from multimedia traffic sources and transmitting them onto the output trunk under given delay and loss constraints. Cell loss and delay characteristics for different traffic sources has been evaluated and an admission control policy is proposed for the safe operation of an ATM node serving multimedia traffic. This procedure of admission control can avoid cell loss at the multiplexer buffer and can avoid traffic congestion inside the network. The cases of both homogenous and heterogeneous traffic mixes are considered in our study and the results are provided for both the cases.

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تقويم أداء معدّد إرسال ذي أسلوب نقل لاتزامني وحركة مرور متعدّد الوسائط

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ملخص البحث . إن هدف هذا البحث هو النظر في أداء منتق (معدّد إرسال) ذي أسلوب نقل لاتزامني محمّل بحركة مرور متعددة الوسائط. ولقد تمّ تطوير نموذج محاكاة حاسوبي لتقديم أداء هذا المنتقي. ويتم تزويد مداخل هذا المنتقي بحركة مرور من وسائط متعدّدة مثل الصوت والبثمرئيات والصور، بغية تقويم الفقد في الخلية (الرزمة) وكذا دراسة تأخير الرزم المرسلة عبر المنتقي، بالإضافة إلى تقويم الانتفاع بمخرجات الدارة الرئيسية ذات أسلوب النقل غير التزامني. ولقد تمّت مقارنة نتائج نموذج المحاكاة الحاسوبي مع النتائج التحليلية التقريبية، وكانت نتائج المقارنة طيبة ومبشّرة. وعلاوة على ذلك تمّ إعطاء موجز لكيفية قبول استدعاءات جديدة بناءً على عامل تفجر (اندفاعية) المصدر وكذا على معدّل ذروته لتفادي كل من ضياع الرزم والتأخير الزائد عبر المنتقي.