Simulation Study of a Run-time Bandwidth Assignment Technique for Delay Sensitive Traffic in High-speed Network

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Abstract

This paper proposes a run-time monitor and bandwidth assignment scheme for delay-sensitive traffic in ATM(asynchronous transfer mode) network. The traffic monitor function is thought to be indispensable to support the QoS(quality of services) of the accepted calls and carry out the network resource management efficiently at the same time. The proposed scheme monitors user data cells by its counter which is driven by system clocks and makes a report on the monitor results to the upper plane such as control plane or network management plane. This feature makes it possible to do the run-time bandwidth assignment and can moderately reduce the size of cell buffers which is inevitable to observe input cell rates in existing traffic controls. Furthermore, any processing delays that are resulted from usage parameter controls can also be eliminated. The proposed scheme is conformed to efficiently manage the calls and reliably improve the overall system performances without a drop in QoS by computer simulations.

1 Introduction

In order to service the delay sensitive traffics such as audio and video informations many protocols have been proposed, and the ATM network[1] has attracted attention as the most suitable high-speed network that can support the traffics having various characteristics. The peculiar functions of ATM network are that it can carry out the statistical multiplexing along with the capability of processing broadband services at a very high speed.

This paper has two objects. The one is to improve the utilization of the network resources in addition to support the delay sensitive traffics by making the most of the statistical multiplexing. To do this, it is a prerequisite to accurately monitor the user data and the proposed scheme should be configured as simple as possible. The other is to provide the first step for the integrated traffic management such as connection admission controls and congestion controls by making reports on the monitor results to network control or management planes. The term integrated means that all traffic control functions are to be united at one place and operated with mutual interactions. This can be achieved by signaling traffic control informations to control blocks in the form of look-up tables.

While it seems likely that the policing[2] and congestion control[3] functions can be handled by some variations of the leaky bucket algorithm[4] or several window mechanisms, there appears to be no general consensus as to the best method to implement the tasks of ATM cell control and bandwidth allocation. One reason is that the considerable buffers for incoming data cells are necessary to accurately do the average cell rate control for the accepted connections. This makes the ATM traffic controls more complicated and the overall cell processing delays are increased consequently. Another reason is that the network resources are wasted to support the pre-determined level of QoS for the variable bit rate or bursty traffics[5]. For example, the leaky rate means that the fixed amount of bandwidth is continuously assigned to the accepted connections even when there is few(or no) incoming cells. Although we can, of course, solve this problem by adopting buffer monitor scheme, this again complicates the traffic controls in the end. Furthermore, it is too difficult to measure the total resource usage rate of all the accepted connections in real-time.

In order to correct these shortcomings and attain the above-mentioned purposes, a run-time monitor and bandwidth re-assignment scheme is proposed. The scheme is largely composed of three parts1. The first part is TMP(traffic monitor part) that monitors current input cell rate and reports its result. The second part is MDPP(monitored data processing part) that calculates the traffic parameters according to the informations which is reported from the TMP. The final one is TCP(traffic control part) that actually processes the traffics by referring the control informations which is set by MDPP. TMP is made up of counters that are driven by the system clocks and designed to transfer the monitor results on conforming the predefined rules that reflect the effects of cell delay variation and suppress its reporting to excess.

In consequence, MDPP can calculate the run-time traffic generation rates of all the accepted calls so it can assign the appropriate amount of bandwidth to arbitrary virtual connection during some time intervals. This feature may be combined to be used for the case that traffic generators can notify their traffic rates in advance just like MPEG2. The additional merit of

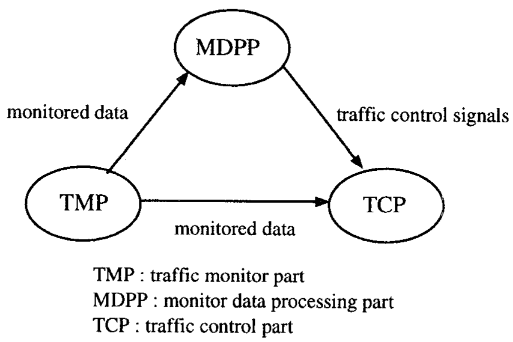

Figure 1: Basic configuration of the proposed scheme

the proposed scheme is that it makes it possible to determine the optimal number of acceptable user connections for a given bandwidth. We can use that value as a threshold one to protect the network from being congested and so support the user informations with the negotiated parameters.

Experiments focus on the facts that how much the proposed scheme reduces the buffer sizes and improve the statistical multiplexing efficiencies by monitoring user traffic in run-time and re-assigning the bandwidth. The general functional requirements for the run-time monitor and bandwidth assignment are stated in section 2 and the details for design is described in section 3. The simulation results under several traffic conditions are described and analyzed in section 4 and section 5 is the conclusion.

2 General requirements for run-time operation

Most traffic sources in ATM networks have bursty characteristics. It is shown that with longer bursts, statistical multiplexing becomes less effective, and thus fewer active sources can be supported for a given amount of bandwidth[6]. Since such a bursty traffic does not require continuous allocation of bandwidth at peak rate, statistical multiplexing technique has been applied to ATM networks in order to manage resources efficiently. Therefore the more traffic sources can share the limited bandwidth.

The general requirements of ATM cell control algorithm are summarized as follows[4].

- capability of detecting any illegal traffic situations
- selectivity over the range of checked parameters
- rapid response time to parameter violations
- simplicity of implementation

The representative ATM cell control algorithms proposed by I-series recommendation and some interesting simulation studies can be found in [7]. The problem when implementing ATM cell control functions using these algorithms is the degree of strictness with which to design them. The complex structure of considerable volume of hardware for monitoring the average bit rates of the incoming cells from bursty traffic sources as well as for accurately controlling data cells also arises as a problem. Most cell control algorithms require input cell buffers and have the concept of obtaining control informations from the data cells waiting in these buffers. Therefore the necessity of considerable cell buffers for monitoring and the consequent delays act as big factors that deteriorate the overall system performances.

In order to overcome the mentioned shortcomings of the existing algorithms, run-time monitor and bandwidth assignment scheme is introduced. For the ATM user data cells generated from the accepted connections, monitor function is executed according to control informations registered in the look-up table and control function is also done simultaneously. Once the monitor and control actions for data cells are made, these informations are reported to control plane as well as to physical layer selectively. Such a configuration minimizes the hardware burdens of the proposed scheme by performing statistical procedures on incoming cells at control planes, and makes real time usage parameter control possible by limiting cell processing delay of TCP within one cell time. The functional requirements of the proposed scheme are are stated below.

Besides the cell clock that is the system synchronization signal, 5 bytes cell header that is synchronized with 5 byte clocks just after the one cell clock is received. The received cell header is separated according to their functions and used to monitor and control user data cells. By the inspection of cell header, the validity judgment of the current data cell is performed to see if it is the cell from the accepted connections. This is accomplished by transmitting the extracted 8bits VPI and 16 bits VCI values to the management plane and receiving the match flag signal.

In this scheme all the user data cells are monitored by counters that are driven by system synchronization clocks. The size of the counters can be adjusted by the minimum cell rate of the traffics that is to be monitored. It is determined by calculating the inter-cell time $T_p = line\ speed/minimum\ cell\ rate$. Upon receiving a user data cell, TCP reads the cell control informations set by MDPP through the look-up table. The control information C_c(counter control value) is determined approximately as $C_c = cell\ size/current\ rate \times\ one\ cell\ time$.

The control action for the current data cell is determined by comparing theC_m(monitored counter value) with the C_c. If the C_m value of a cell is not smaller than the C_c value, it is judged a compliant cell. Otherwise it is judged a non-compliant cell. The unique feature of this scheme is to report this cell monitor information C_m to MDPP so that it can get the statistical data of data cells and update cell control informations C_c for the corresponding connection when necessary. For example, in the case of variable bit rate or bursty traffics, C_m is much greater than C_c in most of the time. Thus C_c is set by MDPP according to the monitored counter value C_m(very large value compared to peak cell rate R_p). When C_m is smaller than C_c, it is reported to MDPP and cell control informations are updated. This is explained more precisely in section 3.

3 Configuration of the run-time monitor and bandwidth assignment

The state diagram of run-time monitor and bandwidth assignment scheme conforming to the general requirements that are discussed in the previous section is shown in figure 2.

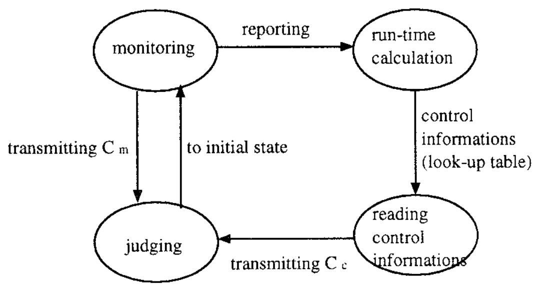

Figure 2: Basic configuration of the proposed scheme

First of all, VPI/VCI and cell loss priority(CLP) bit are extracted from 5 bytes cell header received from the physical layer, and whether the cell is a valid one that originated from an accepted connection, is judged by referring to the control informations in the look-up table. If it is verified as a valid data cell, whether it violates the negotiated parameters is examined by comparing the counter control value with the monitored counter value. If a cell is judged a non-compliant cell, it is discarded or tagged according to the value of CLP bit. The determined cell control action for a valid data cell is transmitted selectively to MDPP and immediately to physical layer where practical processing on that cell is done. Finally cell monitor counter of the corresponding connection is reset.

The proposed scheme is composed of a look-up table where control informations are stored and *three parts* as depicted in figure 1 in section 2. Run-time cell monitor and control are executed using control informations set by MDPP which performs statistical processing of all the transmitted cells. The details about these parts are described below.

3.1 TMP(traffic monitor part)

This part simply reports the monitored value C_m of the current cell to *TCP* and *MDPP*. At first, VPI/VCI, PTI, and CLP bits are extracted from 5 bytes cell header synchronized with the byte clocks just after a cell clock that is generated every 2.726μsec from the physical layer, and sent to TCP. Negotiated parameters of all the accepted connections are monitored by counters which are driven by cell clocks. Counter value for the current valid data cell is extracted and used by TCP to decide whether the cell has violated the negotiated parameters. All counters are driven by cell clocks which are generated when cells(whether they are valid or invalid) arrive on the physical layer. The counter for the current cell is reset at the same time the monitored counter value is transmitted to TCP and MDPP. Of course the reporting is done only when C_m and C_c differ in their values and the difference is larger than a particular limit(allowable range of cell delay variation).

3.2 TCP(traffic control part)

The control information C_c for valid user data cell is read from the look-up table and compared to the monitored value C_m to determine usage parameter control action on that cell. The determined control action signal is transmitted to the physical layer. As the actual control action is performed in this part, procedures to control operation of TCP can be added. That is, in congestion case, by updating control informations in the look-up table such as match flag, C_c, and A_I according to the properties of traffic sources, MDPP can protect high priority cells from being tagged or discarded. And TCP reset signal can be used when TCP malfunctions.

3.3 MDPP(monitored data processing part)

This unit has a close relationship with connection admission control, network resource management function, and bandwidth reassignment function. The peak cell rate negotiated at the connection admission phase is converted to the counter control information C_c and recorded in the look-up table. During transmission phase monitored information C_m is reported. Using the reported informations, statistical traffic parameters are calculated and the control informations are to be updated to according to the current bandwidth utilization.

Monitoring ATM data cells and reassigning bandwidth based on these informations plays an important role in increasing statistical multiplexing gain of the ATM system. That is, by getting reports on cell transmission rate of the accepted connections, remaining bandwidth can be assigned efficiently to a new call or an existing call to the extent that the negotiated quality of service is maintained. If congestion occurs, congestion notification signal is immediately transferred to TCP, and PTI field of the next incoming cells is altered according to this congestion notification signal. In this way the rate-based congestion notification scheme can also be operated.

Bandwidth reassignment is done using a suitable computation method[8] depending on traffic types that users send or can be done dynamically[9] based on the utilization of network resources. Since traffics that actually enter ATM network have various characteristics, a proposal has been made to service traffics having same characteristics in the same band in order to manage finite network resources efficiently. Therefore, in this paper, every traffic is assigned its minimum bandwidth $B_1,\ldots,B_n$ from the total bandwidth according to its characteristics, and the remaining bandwidth

VPI/VCI (24 bits)	Match Flag (1bit)	C_c (14bits)	A_I (1bit)	CNS (1bit)	Co_id ($\leq$ 24bits)
#10	0				
#20	1	xx	0	1 or 0	xx
#30	1	xx	1	1 or 0	xx

Table 1: A Simple Configuration of Look-up Table

is to be shared according to the control by MDPP. The minimum bandwidth of each traffic class is by no means to be assigned to the traffics of different classes and can be taken as the one that is chosen from the computation by the network manager.

The method to assign unused remaining bandwidth to each traffic must be performed in such a way as to maximize performance by efficiently assigning the network resources. MDPP has been designed in such a way that any class of traffic can use the remained bandwidth as long as the current cell arrival rate conforms to the negotiated parameters.

3.4 Look-up table

Table 1 shows a simple configuration of the look-up table. VPI/VCI values of the accepted connections, match flag for verifying the valid cell Match Flag, counter control value C_c, TCP control signal A_I, congestion notification signal CNS, and counter identifier Co_id are stored in the look-up table and used by TCP to control incoming data cells.

When a cell whose VPI/VCI value is #10 arrives, TCP immediately sends cell discarding signal to physical layer since the match flag is set to 0. This is the case that the cell is an invalid cell. If a cell with VPI/VCI value is #20 arrives, since the match flag is 1, it is judged a valid data cell and compared to counter control value C_c. Although C_m is less than C_c, cell tagging or discarding is not executed as A_I signal is low. This corresponds to the case that even if a bursty traffic requiring higher bandwidth than the one currently assigned arrives, it does not violate the negotiated peak cell rate. Of course C_m for the current arrival rate is reported to MDPP and C_c for this connection is updated. When the traffic violates the negotiated peak cell rate, A_I signal as well as the current C_c is updated and the control actions are executed automatically on the following cells that violate C_c. In this case the control informations for the virtual channel #20 are same as for virtual channel #30.

Co_id has been used because the number of virtual connections is small enough to easily find a counter that monitors a virtual connection. But as the number of virtual connections increases, Co_id needs 24 bits(equal to VPI/VCI bits), and becomes useless. Therefore, except the VPI/VCI values at most 17 bits of additional control informations are necessary for each virtual connection. Leaky bucket or window control algorithms must contain several informations; the threshold values of input buffers for each connection, control information for token manager, and predetermined measurement time or window sizes. Although different sizes of control informations are required for different connections, it can be easily seen that the amount of the necessary control informations on a same traffic is much smaller and the hardware complexities are also reduced in the case of the propose run-time scheme.

4 Discussions on simulations

The purposes of the proposed scheme are to improve the utilization of the network resources and provide some bases for the integrated traffic controls. The capability of the exact control over data cells using control informations in look-up table can be verified. The scheme has been designed by using *WorkView* of *VIEWlogic Systems Inc.* and simulated in ALS environments. As a result, it is possible to send a signal indicating whether or not the corresponding valid data cell has violated the negotiated parameter and report the monitored counter value to MDPP within one cell time assuming the transmission line speed to be 155Mbps. More detailed explanations have relation to the complicated hardware facts and so are omitted here(can be referred in [10]).

Therefore, in this paper, the effects of resource utilization is firstly focused on the assumption that all data cells are properly controlled. And it is considered that how the reported informations from TMP would be used for connection admission control and congestion control. To simplify simulations several assumptions are made as follows.

- Data cell processing delays at the physical layer is negligible and homogeneous traffics are multiplexed in a same bandwidth of 50Mbps.
- Cell delay variations of data cells are also assumed to be negligible as the main goal is to assign bandwidth efficiently and improve multiplexing effects through run-time traffic processing.
- In order to reassign the bandwidth to a virtual channel, control informations in the look-up table are updated by MDPP within at least $T_r(\leq \textit{cell size/arrival rate})$ which depends on the current cell rate of that connection.
- The user traffics are generated on conforming to their negotiated parameters and calls are accepted with no limitation. This is to see how many calls can be serviced without the degradation of QoS by merely appending some buffers.

- Dual leaky bucket algorithm is considered for performance comparison of resource utilization and so a fixed size of buffers is used to do peak and average cell rate controls. The buffer size is calculated at a minimum to monitor average cell rate when cells are generated at peak cell rate.

- MDPP manages the network resources by setting A_I signal low as long as there is some extra in total bandwidth, and by making cell loss rate close to zero. Of course A_I signal is changed to high in the case that the traffic violates the negotiated parameters.

Under these assumptions, simulations are done on four variable bit rate traffics.

video telephony with the bandwidth range of 64Kbps ~ 2Mbps, burstiness of 2.5, burst length of 2 ~ 10Kbytes

video conference with the bandwidth range of 128Kbps ~ 14Mbps, burstiness of 2.5, burst length of 1.6 ~ 40Kbytes

videotex with the bandwidth range of 64Kbps ~ 10Mbps, burstiness of 10, burst length of 1Mbits

medical image with the bandwidth range of 1.5Mbps ~ 10Mbps, burstiness of 25, burst length of 5 ~ 8Mbits

Though the last two traffic types may be better serviced by protocols like fast reservation protocol, they have been included in simulation for performance comparison. The first items concerned are multiplexing efficiency and gain. The existing algorithms allocate certain amount of bandwidth computed in advance for each traffic fixedly during a period, and can change window size or leaky rate according to the number of input cells monitored in the previous period. Despite all these, it is very difficult to monitor and control a bursty traffic, and long monitor time is necessary for accurate controls. Unrealistic input cell buffer and token pool size to monitor and control the exact average cell rate of ATM traffics is also pointed out as a big weakness.

Differently from these, the proposed scheme can control traffics having very large burstiness values accurately without being affected by monitor time or buffer size. It is compared with leaky bucket algorithm which is the most well known one among the existing cell control algorithms on the following performance evaluation categories.

For the homogeneous case we can define R to be the maximum bandwidth allocated to some traffic class, m to be the maximum number of virtual connections that can be serviced simultaneously at their peak cell rates in R, M to be the maximum number of virtual connections that can be multiplexed in R, and μ the average cell rate of virtual connections. And we also define the multiplexing efficiency Mux_{eff} to be Mux_{eff} = M $\times \mu$ / R and the multiplexing gain Mux_g to be $Mux_g = M/m$[9]. The multiplexing efficiency Mux_{eff} means the relative bandwidth usage rate of traffics that are multiplexed in the basic bandwidth B_x set 100%, and Mux_g shows how exactly the control scheme can monitor input cells and how efficiently it can reassign the bandwidth according to the monitored cell rates.

The maximum number of virtual channels M that can be serviced simultaneously is, from video telephony, 59, 6, 42, 90 respectively and m is determined 25, 3, 5, 5 respectively according to the peak cell rate. For the traffics having large burstiness values such as medical image(cell delay tolerance of 2sec), it may be more desirable to put in practice the fast resource management scheme than the general cell control techniques. However, a private bandwidth of 55Mbps is allocated to each traffic class in the case of leaky bucket algorithm and basic bandwidth of 50Mbps with the shared bandwidth that averages 5Mbps is for the proposed scheme for the performance comparison.

From the simulation results, run-time monitor scheme produces multiplexing efficiency which averages 77.5% for the pre-defined traffics, while leaky bucket algorithm produces about 56%(Fig. 3). For leaky bucket algorithm to satisfy some level of cell loss requirement as in run-time scheme, larger input cell buffer and more high leaky rates should be used. This means that a certain amount of bandwidth must be allocated to a connection even when the cells are not generated.

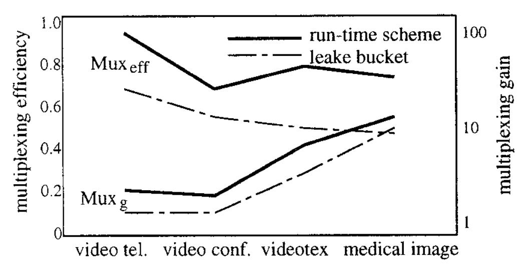

Figure 3: Multiplexing efficiency and gain

While the multiplexing efficiency is determined according to the average cell rates of traffics and basic bandwidth R, multiplexing gain Mux_g has a close relationship with burstiness ranges of traffics. The multiplexing gain increases as burstiness range increases. This is because a large burstiness value means a relatively low average cell rate compared to the peak cell rate. In the same way as the results of multiplexing efficiency, run-time scheme exhibits better performances(about 1.3times) than leaky bucket algorithm as for multiplexing gains.

Some strategies for effective bandwidth management is described from now on. And the method to determine the threshold value of acceptable number of calls is considered. That value may be used to indicate the congestion state and to do connection admission controls. Since the basic(minimum) bandwidth is 50Mbps, when the sum of the bandwidth requirements of all the accepted virtual channels is above B_x, some shared bandwidth is to be allocated or some buffer control policy should be used. This is when more virtual channels than the number of virtual channels that can be serviced with the negotiated QoS are connected. In this case, MDPP assigns the remaining bandwidth of the shared bandwidth appropriately. The required bandwidth of any virtual channel m $BW_{req}(\mathrm{m})$ is computed as $BW_{req}(m) = \sum_{n \neq m} BW_{assg}(n) + R_c(m) - B_x$, when denoting n the total number of virtual channels, $BW_{assg}(\mathrm{n})$ the currently used bandwidth, and $R_c(\mathrm{m})$ the current cell arrival rate of virtual channel m. Such a bandwidth requirement arises when doing simultaneous services for 45 video telephony, 4 video conferences, 27 videotex, or 56 XRay image virtual channels respectively.

Items to be considered together with required bandwidth are time duration of this bandwidth requirement TD_{req} and time interval between requirement originations TI_{req}. The bandwidth may not be used exclusively since the bandwidth $BW_{req}(\mathrm{m})$ should be shared with other traffic types. Therefore, MDPP makes a decision on the allocation of shared bandwidth according to the time duration of bandwidth requirement and its interval. Fig. 4 shows the required bandwidth BW_{req}, requirement duration TD_{req}, and the interval of requirement originations TI_{req} when the basic bandwidth of 50Mbps is given to video telephony sources(can also be referred in [11]).

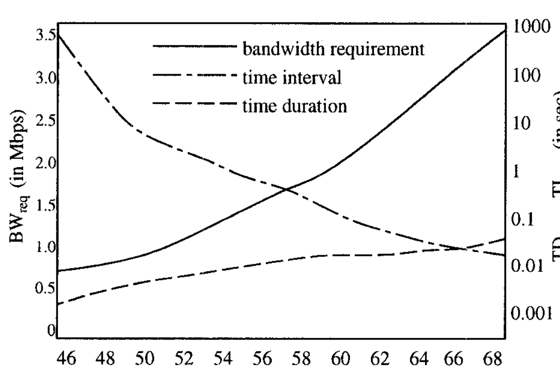

Figure 4: Bandwidth requirement and its time relation

From the simulation results, it can be seen that the QoS of all the accepted connections less than 66 is supported. That is, when the number of connections is 66, run-time monitor and bandwidth assignment scheme can do well only if it is possible to allocate bandwidth of average 2.592Mbps during 42msec with time interval of 44msec. But as the number of virtual channels increases, time interval of bandwidth requirement TI_{req} exceeds the duration TD_{req}. From this time on, allocation of shared bandwidth alone cannot satisfy the QoS of several accepted connections and thus additional private bandwidth must be allocated.

Defining the average shared bandwidth to be 10% of the basic bandwidth, the maximum possible number of connections for video telephony is determined 66. However, considering the possibility of cell loss and congestion, there may be a suitable point(number of connections) that is preferable to 66. It is the number of virtual connections which simultaneously satisfy the followings.

- to maximize the statistical multiplexing gains
- to support the QoS of all the accepted connections
- not to have the shared bandwidth all to itself
- to be appropriate to make it a threshold point for congestion state

Therefore, on coming to this point, network manager will reject new virtual connection requests and run a congestion control mechanism(activating the CNS signal). In the case of video conference, it is possible to support up to 10 virtual channels by allocating the shared bandwidth of average 10.32Mbps for 259msec with a frequency of 306msec. Considering, however, the usable limit of the shared bandwidth(average 5Mbps), it is desirable to service 6 virtual channels by using the shared bandwidth of average 4.219Mbps for 197msec every 4.152sec because the shared bandwidth must be also allocated to other traffics when necessary. In the case of videotex, even when the number of connections is 49, only the shared bandwidth of average 4.3Mbps is required. But since the time interval of bandwidth requirements is almost equal to the requirement time duration, the number of simultaneously serviced virtual connections must be set smaller than this.

Assuming 4 types of traffics and dividing the total bandwidth into 4 basic bandwidths of B_1, B_2, B_3, B_4 and one shared bandwidth of average 5Mbps, run-time scheme can service 42 virtual channels of videotex(BW_{req} = 3.674Mbps, TI_{req} = 2.89sec $\geq$ 4 $\times$ (TD_{req} = 0.513sec)). In the case of XRay traffics, up to 90 connections can be serviced under the same assumption(BW_{req} = 4.229Mbps ($\leq$ 5Mbps), TI_{req} = 2.762sec $\geq$ 4 $\times$ (TD_{req} = 0.666sec)).

Finally, figure 5 shows the buffer sizes required to guarantee the no cell loss at the cell monitor or control blocks. From this simulation results, we can also see that the required buffer sizes for run-time scheme is less than those for leaky bucket algorithm. They range from 1.512Mbits to 3.822Mbits. The run-time monitor scheme needs additional buffers from the time when the number of connections exceeds 46 while leaky bucket needs its monitor buffer from the first time.

5 Conclusion

The key to the success of the ATM based B-ISDN will be if network performance requirements can be supported and statistical multiplexing of user data cells on limited network resources from wide variety

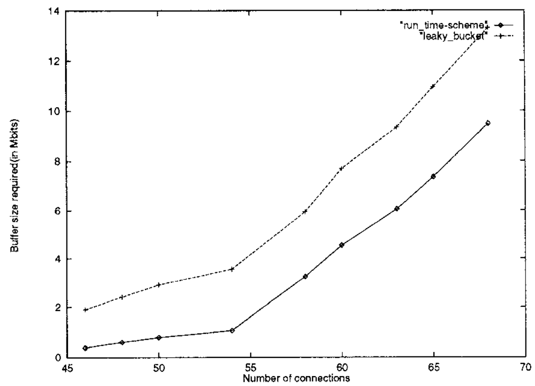

Figure 5: Required Buffer Size

of traffic source types can be done without violating the QoS requirements for individual connections. To achieve these goals, traffic management scheme should be able to monitor input traffics accurately and control them correctly according to the negotiated parameters.

The run-time monitor and bandwidth assignment scheme has been verified to accurately monitor the user traffics and efficiently reassigns the network resources at the basis of control informations set by MDPP. The unique feature of this scheme lies in the fact that TMP reports its monitor result to MDPP located in control plane. In addition to the efficient bandwidth assignment, the reporting function is also helpful to reduce the size of system buffers from the simulation results. This is achieved by mutual interaction between TMP and MDPP, and finally affects TCP to reflect the control informations on user traffics. In this way, run-time processing on data cells is possible and the bandwidth is assigned to each traffic source efficiently. Furthermore, solutions to hardware complexities such as buffer requirements and the cell processing delays due to the input buffers needed in the existing cell control algorithms have been found.

An interesting fact in managing the shared bandwidth is when the time duration of bandwidth requirements grows close to the time interval of requirement originations. In such a case, the proposed scheme results in the loss of transmitted data cells or cell delays as it must store the input cells in the buffer until extra shared bandwidth becomes available. To solve these problems, network manager should limit the number of virtual connections in advance according to the traffic classes in order to maintain the ratio of TI_{req} to TD_{req} greater than any threshold value. This is for further study.

References

[1] Le Boudec J.-Y., "The Asynchronous Transfer Mode : a Tutorial," *Computer Networks and ISDN Systems*, Vol. 24, pp. 279-309, 1992

[2] Catherine Rosenberg and Bruno Lague",, "A Heuristic Framework for Source Policing in ATM Networks," *IEEE/ACM Transactions on Networking*, Vol. 2, No. 5, pp. 387-397, 1994

[3] L.Trajkovic and S.J.Golestani, "Congestion Control for Multimedia Services," *IEEE Network*, Vol. 6, No. 5, pp. 20-26, Sept. 1992

[4] *ITU-T SG XVIII*, Recommendation *I.371*

[5] M.Murata, Y.Ohba and H.Miyahara, "Analysis of Flow Enforcement Algorithm for Bursty Traffic in ATM Networks," *IEEE INFOCOM*, pp. 11c.2.1-11c.2.10, 1992

[6] Wolfgang Fischer, Eugen Wallmeier, Thomas Worster, Simon P. Davis and Andrew Hayter, "Data Communications Using ATM: Architectures, Protocols, and Resource Management," *IEEE Communication Magazine*, Vol. 32, No. 8, pp. 24-33, Aug. 1994

[7] E.D.Rathgeb, "Modeling and Performance Comparison of Policing Mechanisms for ATM Networks," *IEEE J.Sel. Areas Communication*, Vol. 9, No. 3, pp. 325-334, Apr.1991

[8] T.Okada, H.Ohnishi and N.Morita, "Traffic Control in Asynchronous Transfer Mode," *IEEE Communication Magazine*, Vol. 29, No. 9, pp. 58-62, Sept. 1991

[9] Jonathan S.Turner, "Managing Bandwidth in ATM Networks with Bursty Traffic," *IEEE Network*, Vol. 6, No. 5, pp. 50-58, Sept. 1992

[10] Intae Ryoo, "Design and Performance Analysis of Integrated Traffic Control Technique for Broadband Network," *Doctor Thesis from Yonsei Univ., Korea*, Feb. 1994

[11] I. Ryoo, C. Shim, H. AIDA and T. SAITO, "Traffic Monitoring and Bandwidth Management for VBR Traffics in ATM Networks," *Proceedings of the ICCC'95 Conference*, Aug. 1995