STATIC MULTITHRESHOLD RATE CONTROL MECHANISMS IN DOUBLY FINITE QUEUE FOR SUPPORTING ABR TRAFFIC IN ATM NETWORKS

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ABSTRACT:

This paper is concerned with the variety of ABR switch mechanisms available for implementation with the current end system definition. The various switch mechanisms can be broadly classified in two categories depending on the type of feedback information provided to the end system. The feedback mechanism is either binary involving setting the EFCI bit in the data cell header or explicit rate in which case the switch has to calculate a target rate and send it back to the source through an RM cell. Also within each category, different subgroups are formed based on the type of queuing involved (for example per port queuing, per VC accounting, per VC queuing) and the congestion declaration criteria adopted (for example queue thresholding, queue derivative, load monitoring). Our goal is to provide an overview of the various switch mechanisms available and discuss their performance characteristics accordingly. The performance metrics used include queue length, source flow-controlled cell rate dynamics, fairness of bandwidth allocation in the steady state and ramp-up time in the network transient state. Simulations of the different switch algorithms' dynamics for network configurations and scenarios of interest are presented. Each link is assigned a weight that reflects its current utilization. Using these weights, the VPs on highly utilized links are rerouted to less congested physical paths. The algorithm makes use of the equivalent bandwidth concept, which provides an efficient method to estimate capacity requirements of connection requests such that QoS requirements are met. The quality of the solutions achieved by the proposed method is compared to several competitors under varying network topologies and traffic conditions. The observations on the algorithm performance show that the developed method is able to facilitate an efficient use of network resources through the introduction of VPs

Key Words: *ATM, Rate Based Mechanism, Network Link Utilization, Threshold.*

1. INTRODUCTION

The Asynchronous Transfer Mode (ATM) provides the required flexibility for supporting heterogeneous services in a B-ISDN environment [1]. Very large capacity fiber-optic transmission technologies have significantly reduced the transmission cost portion of the total network cost, making the node cost relatively high. This leads to the conclusion that the total network cost would be reduced most effectively by node cost reduction. Simplification of network architecture and node processing is the key to developing a cost-effective, flexible network. This will be possible by implementing the virtual path (VP) concept. The fundamental advantage of this concept is that it allows the grouping of individual connections, also known as virtual circuits (VC), sharing common paths through the network to be handled and switched together as a single unit. Network management actions can then be applied to a small number of groups of connections instead of a large number of individual connections, resulting in smaller total processing requirements, faster processing per VC, and in general, a significantly better use of network resources. More than 90% of processing time can be saved when VCs are routed on VPs rather than processed individually [2].

Congestion control is important in enabling ATM networks to maintain the quality of service (QoS) required by end users [3]. The QoS is measured by parameters such as end-to-end delay (at the cell, burst or message level), cell delay variation and cell loss ratio. Cell delay variation (CDV), also called jitter, is due to the clumping (bunching together) of cells as they travel through the network switches and are delayed in queues. The inter-arrival time between adjacent cells shortens to the point where it affects the network and impacts on other traffic. Also, cells which set out from the source as an evenly spaced sequence arrive at the destination in unpredictable bursts, which affects the playback of speech and video traffic. Excessive cell delay variation (jitter) can cause serious problems for packetized video leading to loss of synchronisation between audio and video components. This manifests itself in the form of breaks in the picture, audible gaps in sound and even loss of lip synchronisation. Cell loss causes significant visual glitches for video (blocking/mosaic patterns, sparkles, blurring etc. [4]) and annoying clicks during telephone conversations. Cells may be lost due to data corruption (error) or deliberately dropped in times of network congestion. Error rates for digital transmission over optical fibres are generally very low. Thus effective congestion control is vital.

To identify the key characteristics of VP routing and capacity allocation, some performance issues affected by the design of a VP network have to be considered. A good VP layout is characterized by achieving a good performance trade-off among them [5].

1. The number of VPs used for routing a VC (termed VP hop count) should be small in order to reduce the VC setup complexity introduced by bandwidth allocations and changes in the VC routing tables.

2. The chosen route for a VC should be short in terms of the number of physical links it uses, or in terms of propagation delay, to efficiently utilize the communication network.

3. The number of VCs handled by a VP should be small to keep the number of occupied entries in the VP routing tables (termed the load on the table) low.

4. The number of VPs that share any link should be small so that if a link is disconnected, the number of VPs that need to be rerouted in order to by-pass the faulty link is small. The purpose of this study is to develop a method of VP routing and bandwidth allocation in ATM networks.

The proposed method applies dynamic capacity control in order to meet QoS requirements such as limited delay and bounded cell loss probability. It also tries to distribute the network traffic as evenly as possible, since a balanced traffic load helps: (a) limiting the effect of link failures; (b) decreasing the chance of link saturation; and (c) increasing the network robustness. As a result, the method can facilitate an efficient use of the network resources.

2. RELATED WORKS

Point-to-multipoint ABR algorithms A rate-based algorithm for pt-mpt ABR service was first proposed by Roberts [6]. In his algorithm, a switch at the branching point returns a backward RM cell to its upstream node whenever it receives a forward RM cell. Also, for every received backward RM cell, the switch will consolidate the congestion information in the received backward RM cell to local variables maintained by the switch on a per-VC basis. This algorithm guarantees that the root gets exactly one feedback RM cell for every forward RM cell regardless of the number of leaves in the multicast tree, in order to avoid the ‘‘implosion’’ of RM cells. It also achieves fast transient response, especially in WAN environment where link propagation delay is considerable. This is due to the fact that congestion information can be immediately returned from a branching point without waiting for the backward RM cells from the destinations. However, such congestion information may sometimes be inaccurate and may lead to persistent rate oscillation. Siu and Tzeng [7,8] presented a fundamental study on pt-mpt ABR service, with emphasis on the protocol performance in terms of fairness. In particular, they established a unified framework to derive a pt-mpt congestion control protocol from a given pt-pt rate control protocol. Moreover, the resulting pt-mpt protocol preserves the fairness characteristics of the underlying pt-pt protocol. A more conservative scheme modified from [6] was also proposed in [7,8]. In the scheme of [6], a switch will return a backward RM cell triggered by each forward RM cell; this backward RM cell may contain congestion information only about the current switch but not that of that downstream nodes.

Thus, the source may receive backward RM cells with incorrect congestion information, and some ‘‘noise’’ will be introduced into the computation of the appropriate emission rate at the source.

The key idea in the scheme of [7,8] is that a switch will return a backward RM cell to its upstream node only when it receives at least one backward RM cell from its downstream nodes. Consequently, the congestion information contained in the backward RM cells is more accurate in the sense that it reflects the bottleneck rates among all links traversed by the VCC.

However, this scheme will inevitably exhibit a slower transient response compared with the scheme in [6]. Note that both algorithms discussed above require that the switches at the branching points generate backward RM cells. Such a requirement imposes significant complexity and cost to the switch implementation. In this paper, we introduce a new approach to consolidate the feedback information in a pt-mpt connection. The ABR service specified by the ATM Forum [9] requires destination end system to return (or generate) a backward RM cell for every received forward RM cell in the pt-pt case. Similarly, in our approach for pt-mpt ABR service, the backward RM cells are generated solely by the destinations but not by the switches, as in the pt-pt case.

Multiway communication in ATM. The ATM Forum has formed a working group on multiway communication to discuss issues in mpt-mpt communication in ATM and possible approaches to some of the key problems involved. To resolve the cell-interleaving problem associated with AAL5, it is suggested that one possible approach is to regulate the transmission from all leaves in a mpt-mpt connection. According to this approach, only one leaf is allowed to send data at any time, and therefore, no cell interleaving could occur. Some traffic control techniques need to be used to ensure that all leaves would obey their media access agreement. This approach requires no change in the switches, but it shifts the burden to the application, signaling, and traffic management. The requirements for mpt-mpt communication in ATM are also discussed in [10].

3. QUEUING POLICIES

The type of queuing discipline used in an ABR switch, is another option that is left vendor implementation dependent. This particular feature was at the heart of the debate between credit-based and rate-based both considered for ABR flow control. It was felt that the credit-based did not scale well since it required per VC queuing.

Per port queuing – With this queuing discipline, all VCs on the same port share the same buffer pool. Although, attractive for its simplicity, this discipline lacks fairness, especially if used in combination with EFCI marking. This is due to the fact that the switch does not differentiate between the different VCs and doesn't keep track of their individual flows. On the other hand, if used in ER switches, both fairness of bandwidth allocation and source rate convergence to fair share can be achieved.

Per VC accounting **–** This is a hybrid version between per VC aggregation and separation. First appearing in [11], it borrows simplicity from per port queuing and intelligence from per VC queuing. All cells are still queued per port, but the switch keeps information on each VC in look-up tables. This solution can provide better results than pure per-port queuing.

Per VC queuing – It is clear that separating VC flows into different queues can provide not only information on each VC but also accurate control means to regulate traffic. In this case, even with EFCI feedback a Round Robin scheduler can be used to achieve fairness. Although not very popular because of the cost and complexity involved, this discipline can give close to optimal performance results. It can be used with any type of feedback and congestion criteria.

3.1 Explicit rate computation

The effectiveness of the ER approach depends on how a switch computes the feedback rate. The main goals are (1) to provide a fair share to all users or VCs, (2) to maximize the link utilization up to a predefined cap, and (3) to keep the buffer utilization within a reasonable range. One of the biggest challenges is to provide an algorithm that is responsive to network changing conditions but demands little processing power of the switch, as a switch may be required to support hundreds of ports and thousands of VCs. In order to compute the fair share for individual VCs, one needs to use a combination of policies in areas such as input rate computation, congestion criteria and queuing policies which are described in the previous sections. The following discusses some general factors that apply to the ER rate determination.

Target link rate (TR) – TR is normally a fraction of the available link capacity for the ABR service. This fraction can be constant, for example 95% of the link capacity or it can vary as a function of the queue length for example.

Fair share (FS) – FS is generally computed according to the Max–Min fairness criteria [11] with the assumption that all VCs have their MCR=0. For a given link, the VCs can be divided into two groups: constrained and unconstrained. The constrained group are those VCs that cannot use all their fair share of the bandwidth because of a limited demand or a bottleneck somewhere else along the path. FS is then computed according to:

$$FS = \frac{AC - \sum_{i=1}^{m} C_i}{n - m}$$

where AC is the available ABR capacity which is the same as the target link rate TR, C is the constrained VC capacity or rate, m is the number of i constrained VCs, and n is the total number of BR VCs. Other fairness criteria with MCR not equal 0 suggested in [12] include techniques such as allocation proportional to MCR, MCR plus equal share, maximum of MCR or Max-Min share, and weighted allocation.

Mean ACR(MACR)– MACR is an average ACR computed at the switch and used for the estimation of ER. The MACR is updated using an exponential averaging whenever a FRM cell is received during ACR promotion (under load condition with MACR-ACR .or ACR suppression (overload condition with MACR>ACR):

MACR=MACR+(R-MACR.) *AV

where R is equal to CCR of the FRM cell , and AV is an averaging factor controlling the sensitivity to current changes. In [8] the MACR is updated every FRM cell and R=CCR*min(1,TR/IR) where IR is equal to the input rate (per port) during one measurement interval.

Load factor (LF) – LF is the link load factor computed at fixed time intervals such as every round trip time [7,16,17] or every N data cells [10] according to:

$$LF = \frac{IR}{TR}.$$

3.2 Queue Response-Time Analysis

The RTA presented in this section assumes that each output queue (in the source nodes and ATM switches) uses priority queuing with "First In, First Out" (FIFO) for cells with equal priority. Thus, the analysis is also applicable to strict FIFO queuing (which can be considered as a priority queue with a single priority level). Figure 1 shows the general structure of the considered network. In the figure, TGi denotes a traffic generating process, which generates messages to be transmitted. These messages are segmented into cells, which are queued in a cell queue associated to TGi. A cell spacer moves cells with a rate of 1/ti from the cell queue to the prioritized output queue. The reason for using a cell spacer is to reduce the burstiness of the traffic and thereby reduce congestion and increase network utilization. From the output queue, cells are sent to the first switch. In the switch, cells are placed in the appropriate output queue, and transmitted further to the next switch until the destination is reached.

It should be noted, that in this paper we focus on the delay in the output queues and in the cell spacer. In this we show how to incorporate other delays into the RTA, and also how the maximum buffer need can be calculated. To understand the different delays involved, it is important to understand the worst-case traffic pattern for a periodic stream experiencing jitter. The largest number of cells in a given time interval appears if the first cell, which should arrive in the very beginning of the interval, experience a maximum delay caused by jitter, and if all subsequent cells experience minimum delays. To calculate the end-to-end delay (the response-time), we first derive the worst-case delay for a message to pass through the cell spacer. Then we characterize the worst-case cell arrival pattern to an output queue.
From this pattern we derive the worst-case

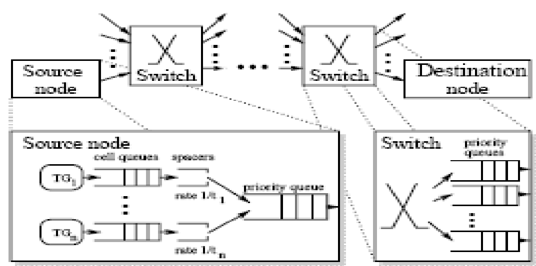


Figure 1. The network architecture

queuing time for a cell. Finally, we describe how the queuing delay changes the shape of the traffic, i.e. we derive the new traffic parameters which can be used to calculate the queuing delay for a subsequent output queue.

3.3 Queuing and packet scheduling mechanisms

The scalability requirement for mpt-mpt connections implies the need to use a single VCI per link for the transmission of packets from all sources of the multicast group. A switch at a merging point needs to map multiple incoming VCIs into one or more outgoing VCIs, and it needs to perform this without interleaving cells of different packets, while not reassembling the packet and suffering the associated delays. We propose the following mechanism be implemented at a merging point.

1. A separate queue is implemented per neighbor upstream switch along the VCC to isolate cells from different packets before the cells are merged to a single VC link.

2. Packet level scheduling is used to transmit cells from each queue for each mpt-pt connection. This mechanism ensures that one packet is fully transmitted before any cell from the next packet is sent, thus avoiding interleaving of cells. However, there is no need to reassemble the cells into packets so that the delay for processing the packet headers will not be incurred. For example, we can use a counter to keep track of the number of packets for a particular mpt-pt VCC buffered at each queue. For AAL5, each End-of-Message (EOM) cell received will increment the counter by one for the corresponding queue and connection, and after each EOM cell is forwarded to the downstream node, the counter will be decremented by one. Before one queue is served, a scheduler will check the corresponding counter first. If the number of packets counted for this queue is zero, the scheduler will skip this queue and serve the next corresponding queue immediately (Figure 2). Note that each (VCi=is1,2,3) described in Figure 2 corresponds to a different VC link.

Since the queuing and scheduling mechanism above can avoid the cell interleaving problem associated with AAL5, it can be used to support mpt-pt connection over the unspecified bit rate (UBR) service for data traffic, where no flow control mechanism is employed. However, more sophisticated packet-level weighted fair queuing (WFQ) scheme [14,15,16] other than simple round-robin scheduling will be needed to ensure the fairness of bandwidth allocation among connections with packets of different size. However, WFQ is relatively expensive to implement, and such additional costs defeat the purpose of a low-cost UBR service for data traffic. On the other hand, a fair ABR switch algorithm combined with a simple round-robin packet scheduling at the merging point is sufficient to provide fair bandwidth allocation to different senders.

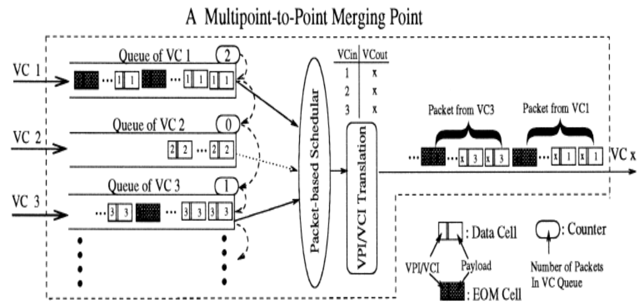


Figure 2: Queuing mechanism plus packet-based scheduling at a merging point.

4. PROPOSED MODEL

4.1 Doubly Finite Queue Algorithm

Let C be the capacity of a link and γ be the desired utilization at the link. The MSVDR presented in routers, at a router works as follows:

• The router maintains a doubly finite queue whose capacity $C^1 \leq C$ and whose buffer size is equal to the buffer size of the real queue. Upon each packet arrival, a fictitious packet is en queued in the Second queue if there is sufficient space in the buffer. If the new packet overflows the Second queue buffer, then the packet is discarded in the second queue buffer and the real packet is marked by setting its ECN bit or the real packet is dropped, depending upon the congestion notification mechanism used by the router.

• At each packet arrival, the second queue capacity is updated according to the following differential equation:

$$C^{l} = \alpha(\gamma C - \lambda), \quad (2)$$

where λ is the arrival rate at the link. The rationale behind this is that marking has to be more aggressive when the link utilization exceeds the desired utilization and should be less aggressive when the link utilization is below the desired utilization. We now make the following observations. No actual enqueueing or dequeuing of packets is necessary in the second queue also, we just have to keep track of the second queue length. Equation (2) can be thought of as a token bucket where tokens are generated at rate $\alpha\gamma C$ up to a maximum of C and depleted by each arrival by an amount equal to α times the size of the packet. Define

B = buffer size
s = arrival time of previous packet
t = Current time
b = number of bytes in current packet
AQ = Number of bytes currently in the second queue
Then, the following pseudo-code describes an implementation of second queue in MSVDR scheme:

The Doubly Finite Queue Algorithm

At each packet arrival epoch do
$AQ \leftarrow \max(AQ - C^l(t - s), 0)$ /* Update Second Queue Size = Primary Queue Size */
If $AQ+b > B$
To find alternative path call MSVDR Algorithm
else
$VQ \leftarrow VQ+b$ /* Update Scondary Queue Size */
endif
$C^l = \max(\min(C^l + \alpha * \gamma * C(t - s), C) - \alpha * b, 0)$
/* Update Secondary queue Capacity */
$s \leftarrow t$ /* Update last packet arrival time */

4.2 On-Demand Path Selection

When a connection setup request arrives, it is the responsibility of the source node to choose a path that is most likely able to support the required QoS. The source node makes a routing decision based on its local knowledge of the network topology. As a result of the PNNI configuration, the knowledge base of a node contains full information about its own peer group, aggregated information about its parent group, more aggregated information about its grandparent group, and so forth. Therefore, source routing to a destination node outside the peer group of the source node is actually to find a path up the hierarchy to the level that source and destination nodes are in the same logical group.

Assume that a connection from the congestion node ***S*** to the destination node ***D*** is to be setup. The parent logical node of the congested node ***S*** and the destination node ***D*** at level i in the hierarchy is denoted as ***Si*** and ***Di***. Let us consider the case where two QoS parameters need to be satisfied: bandwidth ***BW*** and segment-to-segment delay ***DY***, which stand for the attribute and metric parameter types respectively. The Multi-Source Dynamic Routing (MSVDR) decisions can be made in the following steps:

Step 1: Calculate Shortest Path from congested Switch to Destination Node using Hierarchical Least Loaded Routing Algorithm

Step 2: Send reroute information to destination with new source address, routing path and starting cell number

Step 3: Send message to original source to select the second shortest path using OSPF Algorithm without cross the congested link

Step 4: Original source send message to destination with new route path and starting cell number

Step 5: Destination rearranges the cells after receiving the cells from multi source

Step 6: Destination send to the acknowledgement signal to respective sources

Step 7: Read the Cell form Queue with using priority scheduling algorithm

Step 8: Normal Data Flow

5. REROUTING METHOD

The rerouting mechanism is to provide fast recovery in the cases of link failures or topology changes that affect an ongoing session. Since ATM is a connection oriented technology, a new connection should be setup between the source and the destination if the current connection is broken. Link or node failures may occur in different places:

At level i, at which a logical group $PGi(x)$ contains the ancestors of boththe source and destination nodes. Rerouting is started from the very beginning in this case.

In the peer group which contains the source node at level 1. At the levels between level 1 and level i. wherever the node or link failure occurs, doubly queue finite queue maintain at the extreme state the rerouting starts from that particular level. The dead node or link is pruned from the set of pre-calculated paths. After that, phase 2 on-demand routing is performed again from the level at which this failure occurred down to level 1. The doubly finite queue management is shown in the (figure 3), and also explain the maintenance of the finite queue flow diagram is given in the (figure 4). Sampling data and simulation results are discussed in the coming fourth session.

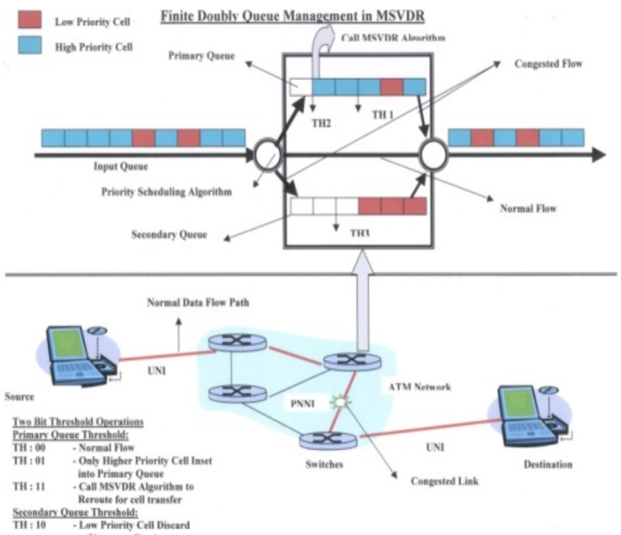


Figure 3. Doubly Finite Queue Management

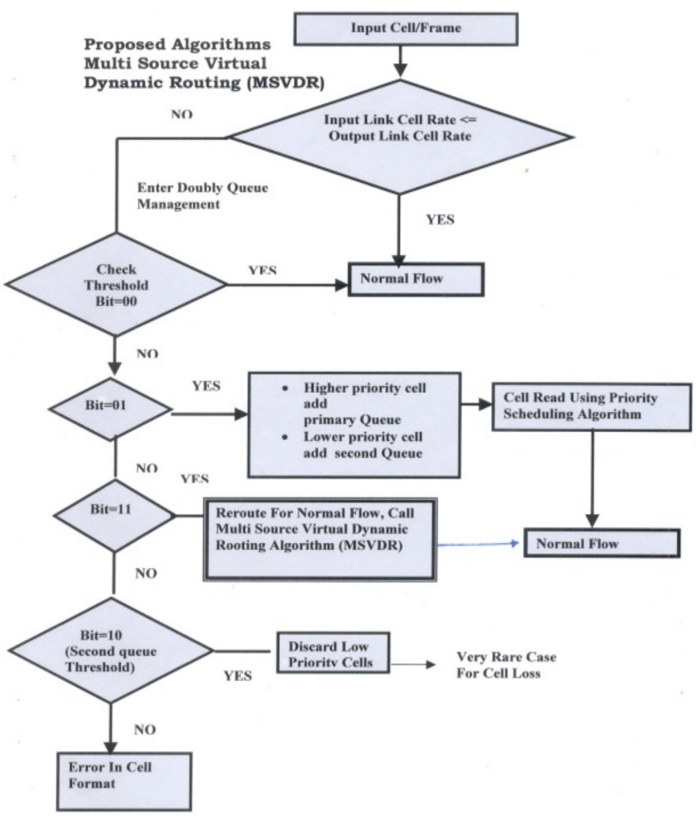


Figure 4. Flow Diagram for Doubly Finite Queue Management

6. SIMULATION RESULT AND DISCUSSION

In this section, we present simulation results which demonstrate the performance of the MSVDR scheme. We evaluate the performance of the scheme based on the following items.

1. Fairness in the equilibrium state allocation of the bandwidth.

2. Transient behavior experienced in the presence of sudden traffic changes.

3. Maximum and mean buffer length in different network scales.

The queuing model of the simulations is illustrated in (Figure 5). The transmitter (T) always serves the RM cells in preference to the data cells. In all simulations, we assume that all VCSs are persistently greedy i.e., all VCSs will attempt to transmit data cells at their peak cell rate. Each transmitter (T/R) will independently remove data cells from the VCS buffer and transmit them to the receiver (R). The intelligent holding will be performed by a bottleneck switch if the number of cells queued in the buffer (data cell and RM cell buffer) in the forward direction exceeds a threshold value. In all simulations, the threshold value is considered 50 cells. The VCS parameters used in the simulation are listed in (Table 1).

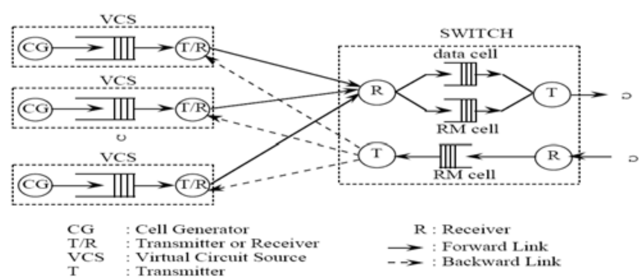


Figure 5. Queuing Model

PCR (Mbps)	150	150	150	150	150
MCR (Mbps)	0.15	0.15	0.15	0.15	0.15
ICR (Mbps)	7.5	20	0.5	0.5	0.5
MAX_AIR (Mbps)	2	8	1	1	1
MIN_AIR (Mbps)	0	0	0	0	0
MDF	4	4	5	5	5
NC	32	32	32	32	32
DTP (cell times)	128	64	128	128	128
DTT (cell times)	16	16	16	16	64
no_of_RM	1	1	1	2	4

Table 1: VCS Parameters

Fairness Performance

The Generic Fairness Configuration (GFC) is used as a benchmark for the assessment of the fairness performance. The GFC is shown in (Figure 6). There are 23 connections, grouped into 6 classes (A-F). The number of connections belonging to class A, B, C, D, E, F are 3, 3, 3, 6, 6, 2, respectively as specified in parenthesis after the class label. The network consists of 5 switches with links of various capacities. We consider fairness performance in a LAN environment. The distance between VCS/VCD and its nearest switch is about 100 meters and the distance between two switches is about 400 meters.

Let us consider to connections, class B and E, which share the same bottleneck link L4. Since the VCS of class E is closer to the link and passes more congested switches than that of class B, class E will get more of the link bandwidth than class B. This is analogous to a parking lot scenario in which E is closer than B from the exit. However, in a good ATM network with separate VCs for class B and E, they will share the available bandwidth of the link fairly (in this case 100 Mbps). Thus, each connection in class B and class E should take about 1/9 of the bandwidth (or 11.1 Mbps).

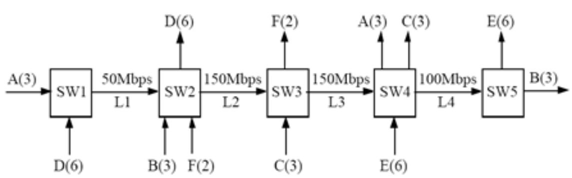


Figure 6. Generic Fairness Configuration

Considering the above fairness objective, we can allocate the bandwidth of each link to each connection as follow.

1. The VCs on the most congested link will share the link bandwidth equally, and this determines the rates to be set for these VCs.

2. Then apply the procedure to the other VCs with the remaining bandwidth of the network.

3. Continue repeating the procedure until rates for all the VCs have been assigned.

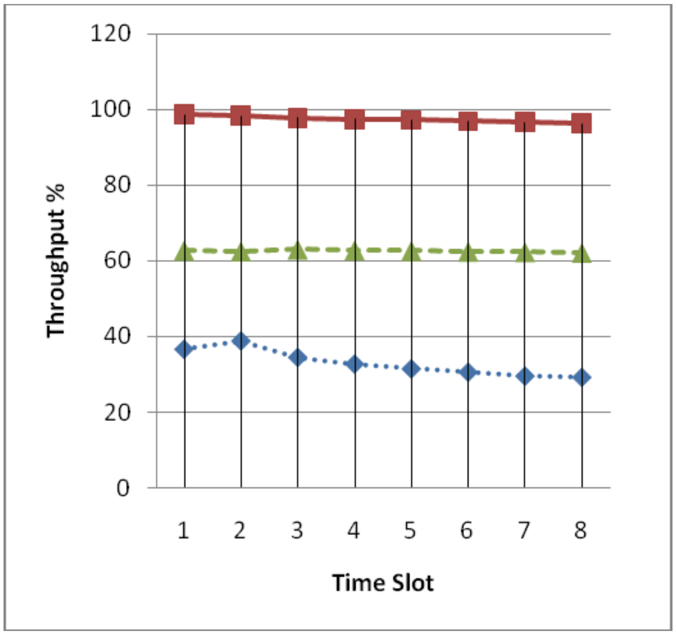
Using the above fair rate-setting procedure, the bandwidth assigned to individual connection classes is shown in (Table 2).

Connection class	Allocated bandwidth	Bottleneck link	Bottleneck switch
A	5.56 Mbps	L1	SW1
B	11.1 Mbps	L4	SW4
C	33.3 Mbps	L3	SW3
D	5.56 Mbps	L1	SW1
E	11.1 Mbps	L4	SW4
F	50.0 Mbps	L2	SW2

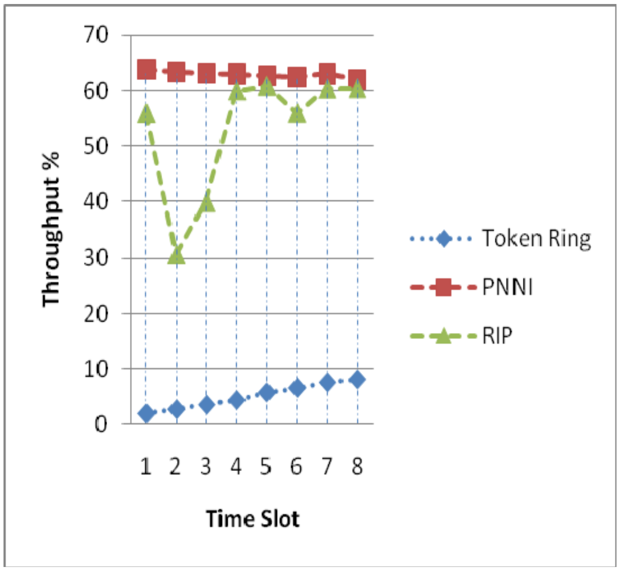
Table 2. Expected Bandwidth

Maximum throughput

Using the doubly finite queue mechanism the ATM switches, the link utilization is very high because of at that time of congestion occur, the cell are added in the primary queue, the primary queue first threshold bit set 01 then the high priority cell added to primary queue and lower priority cell added to second queue. Whenever the primary queue second threshold bit is set to 11 then we call the MSVDR algorithm. The algorithm search the least loaded link from congested place to destination node, the maximum data transferred through this least loaded link to destination, so the maximum link utilization is done by using this algorithm is shown in the (graph 1,2).



Graph 1 : Throughput comparison of PNNI, RIP & Token Ring Protocols with MSVDR algorithm

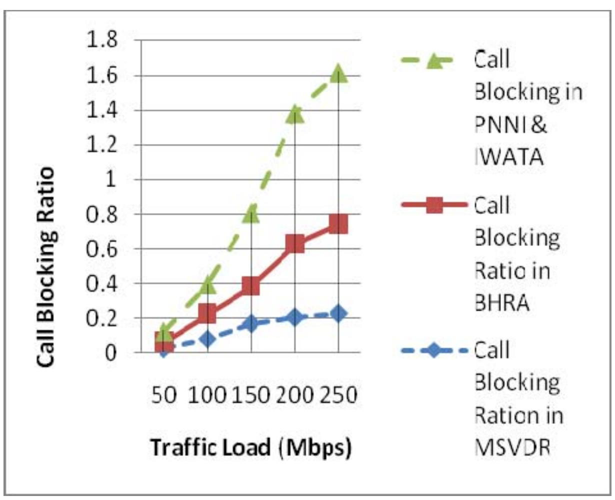


Graph 2 : Throughput comparison of PNNI, RIP & Token Ring Protocols without using MSVDR algorithm

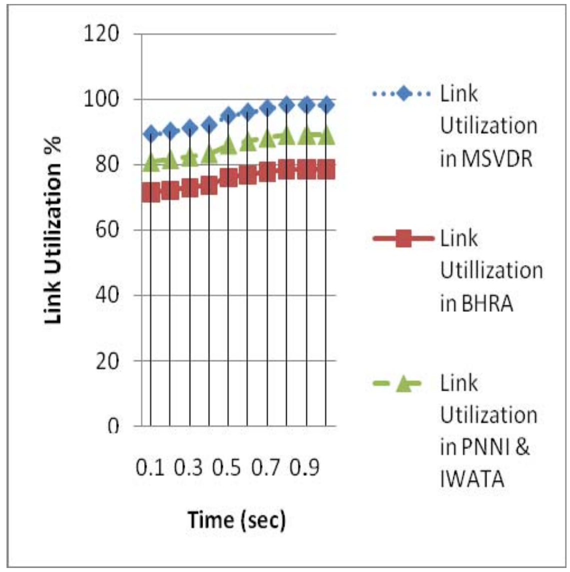
Minimizing Queue Delay time

Graph 3) describe the delay time and queue length for finite queue is minimize, the loss of the data is automatically reduced, then avoid the same cell retransmission then call blocking ratio automatically reduced.

(Graph 4) describe the maximum link utilization within the minimum medium access time in the doubly finite queue management compare with other queue management.



Graph 3. Comparison of Call blocking ratio in CBR traffic load



Graph 5. Link Utilization in Simulation model

7. CONCLUSION AND FUTURE WORK

In this paper, a new QoS-based routing algorithm called the Multi-Source Virtual Dynamic Routing Scheme (MSVDR) has been proposed. The algorithm is compliant with the PNNI protocol and supports multiple QoS requirements, using an adaptive and iterative path search approach that takes advantage of the PNNI hierarchical structure to reduce path computation complexity and maximize network throughput. The MSVDR show that call setup time is significantly reduced, and the computational overhead and call blocking probability are lower, compared to other PNNI routing algorithms. Network throughput is also improved by evenly distributing the traffic among several eligible paths.

We proposed a new queue mechanism to use inside the ATM switches before calling the MSVDR algorithm. This scheme depends on therate-based scheme for coping with congestion control in ATM networks. The MSVDR scheme uses the positive feedback rate control and intelligent holding in each switch in order to resolve the problems which happen in FECN, BECN and PRCA. The behavior of the MSVDR scheme was evaluated by simulations. From the simulations results we conclude:

1. In transient behavior the MSVDR scheme ramps up quickly to the fair rate.
2. There are not noticeable rate oscillations. This means, the MSVDR scheme improves

bandwidth efficiency and reduces the buffer length.
3. For MAN and WAN environments the MSVDR scheme in which each VC has more than one RM cells offer adequate performance.

In this work, we consider that each VC knows its bottleneck switch. However, in real environments the bottleneck switches of the VCs may change dynamically. Therefore, we are investigating a method to determine the bottleneck switch of a VC in a dynamical environment.

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