A SCHEDULING ALGORITHM FOR WDM OPTICAL NETWORKS

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ABSTRACT

This paper proposes a scheduling algorithm for time-slotted WDM broadcast-and-select optical networks. The algorithm is free from collision and supports a particular class of quality of service (QoS), namely constant bit rate (CBR). The running time complexity of the algorithm is $O(M\log^2 N)$ [1], where M and N are the number of packets used for scheduling and the number of nodes, respectively. This running time can be improved to $O(\log^3 N)$ by parallel processing.

Keywords: WDM optical networks, Scheduling algorithm

1.0   INTRODUCTION

Wavelength-division-multiplexing (WDM) is emerging as the most promising approach to exploit the huge bandwidth of fibre optic [1]. This approach divides the optical spectrum into many different channels, where each channel corresponds to a different wavelength. The WDM networks may comprise of several channels up to several tens of channels at different wavelengths, each may be operating at the peak electronic rate. All end-user equipment need to operate only at the bit rate of a single channel.

The simplest physical architecture to implement a WDM network is based on broadcast-and-select approach. Using this approach, the transmission signal from a node is broadcasted to all the nodes in the network. At the receiver end, the desired signal is then extracted from the entire signal spectrum. For example, a node could transmit at an optical wavelength different from the other nodes, and then at the receiver, a tuneable optical filter can be used to select the desired wavelength for reception. Alternatively, the networks may either have tuneable transmitters and fixed tuned receivers, or have both transmitters and receivers tuneable. In general, this architecture is "all optical" in nature, whereby once information enters the network, it remains in the optical domain until it is delivered to its destination.

Communication networks are undergoing rapid development. In the past, the largest networks were electronic, circuit switched networks carrying mostly voice traffic. However, with the explosive growth of the Internet and World Wide Web, electronic packet switched networks have become ubiquitous. At the same time, fibre optic transmission technology has advanced from lower rate multimode fibre links to single mode fibre links capable of carrying multiple channels of several gigabits per second per fibre over a longer distance [1]. However, the optical technology only contributed at the physical layer of the typical multi-layer protocol stack, as the higher layers used are currently still electronic based. Such architecture does not maximally utilise the unique advantages offered by the WDM optical networking. This realisation has led to a recent wave of research being aimed at eliminating the electronic layers that come between a protocol of packet switched network and WDM. The focus is to design a packet switched network that takes full advantage of the unique capabilities of the optical (WDM) layers.

Considering that optical packet switching and buffering technology are still not a matured technology, the future of WDM-based packet networks is much brighter in local area networks (LANs), because LANs span short distances and are inherently based upon a shared medium. Due to the limited nodes and relatively lesser congestion, there is only minimal need for switching and buffering within the network. This makes it possible to design optical LANs that eliminate the electronic layer processing and transmit the packet directly over the optical light-paths. Typically, WDM-based LANs assume the use of a broadcast-and-select architecture. The WDM broadcast-and-select networks have been tested in many testbeds around the world. For instance, DARPA AON has shown that its time-

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[1] Polylogarithmic time, it is the same with $O(M(\log N)^2)$.

slotted WDM broadcast-and-select network testbed is capable of providing time-slotted service, where each time slot is two microseconds [2].

In a time-slotted WDM broadcast-and-select network, the transmitter must know when to transmit a packet at which wavelength, while the receiver must know when to tune to the appropriate wavelength to receive the packet, which requires some form of transmission coordination. A number of scheduling algorithms that perform these coordinations have been proposed. They can be broadly categorised into two categories, viz. 1) scheduling algorithms that consider forms of hardware constraints, such as tuning latency and limited tuning range, and propose various scheduling approaches to overcome these constraints [3, 4, 5], and 2) scheduling algorithms that did not consider these kinds of hardware constraints [6, 7, 8]. The algorithms proposed in the second category were aimed at achieving optimality, such as optimal schedule length and network throughput, and small running time complexity. All scheduling algorithms in both categories are found to be efficient (i.e., having polynomial time complexity), while running on sequential machines.

This paper proposes a scheduling algorithm for time-slotted WDM broadcast-and-select optical networks. The algorithm uses transmission requests or queue length information of the transmitting nodes as the input information and produces a collision free transmission schedule. A technique in graph theory known as edge colouring bipartite multigraph is used as an essential part of the algorithm. The algorithm is able to guarantee a constant bit rate (CBR) quality of service (QoS) to a requesting node. The algorithm can be easily executed on a parallel computer to improve its processing speed.

2.0   SYSTEM DESCRIPTION

2.1   Model Network

The model network consists of N nodes, each connected to a passive star coupler via a two-way fibre as shown in Fig. 1. Each fibre supports W+1 WDM channels; W channels are used for data transmission and the remaining channel is dedicated for transmission control.
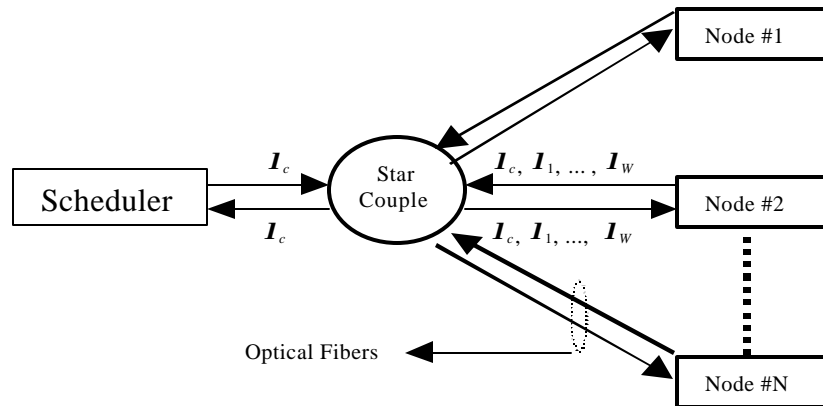

Fig. 1: The conceptual representation of the model network

The data channels are denoted as l_1, l_2, . . . , l_W and the control channel is denoted as l_c. The data channels and control channel are divided into fixed-size time slots such that data and control messages are transmitted in fixed-size packets, where a packet can be sent on one wavelength in a time slot. The size of the time slot of the control channel could be smaller than the data channels depending on the amount of information to be transmitted.

Each node is equipped with a tuneable transmitter and a tuneable receiver to access any of the data channels. The tuning latency of these tuneable transceivers is assumed negligible compared to the length of the time slot. In addition, each node is equipped with a fixed tuned transmitter and a fixed tuned receiver, both of which are tuned to the control channel. The packets that arrive to a node from the upper layer are queued in the buffer of the node.

Although it is assumed that each node is equipped with only a tuneable transmitter and a tuneable receiver for data transmission, it is noteworthy that certain nodes that require higher bandwidth, such as a server or a router, could be

equipped with multiple tuneable transceivers. As the scheduling algorithm will treat all such cases as multiple nodes, it is transparent to the algorithm and can be implemented easily.

The scheduler continuously checks the network to see how many packets each node has to transmit. It selects these packets for scheduling based on certain criteria. The scheduler has a scheduling algorithm to compute the schedule using the selected packets. Then it broadcasts the schedule to all the nodes via the control channel. The node will then transmit the packets based on the schedule. While these transmissions are being carried out, the nodes are re-examined and the schedule is re-determined. This process is repeated, and each such process is called a *cycle*, and the collection of all the time slots within a schedule is called a *frame*. Frames may be of variable size depending on the algorithm's implementation and the number and patterns of the selected packets. All the transmissions related to each node's status and schedule broadcasting are transmitted via the control channel.

There are many ways for the scheduler to gather network status via the control channel; the simplest is based on the round robin reporting (or polling) [8]. The N nodes may report in a fixed round-robin fashion using l_c, and each node's report contains the information of queued packets at the node and the destination address of these packets.

Generally, the number of nodes is much more than the number of channels, $W << N$. This paper considers the case of $W \leq N$; since the N nodes in the network possess N transmitters and N receivers, there are a maximum of N packets that may be transmitted in a time slot. Thus, for the case $W > N$, the situation is similar to $W = N$.

It is assumed that each transmitter and receiver is individually synchronised such that packets arrive at the star coupler at the time slot boundaries. For instance, if the propagation delay between node i and the star coupler is d_i, and the first time slot in schedule s start at time t, then the transmitter at node i may start to send a packet on the first time slot of s at time $t - d_i$, and the receiver at node j may start to receive a packet from the first time slot of s at time $t + d_j$.

2.2 The Scheduling Algorithm

The traffic pattern of a WDM optical network can be represented by a traffic matrix, where (i, j) entry represents the traffic load from node i to node j, and this traffic load may be either a transmission request or queue length information.

The traffic pattern can also be modelled by a bipartite multigraph $G(U, V, E)$. U is the set of source nodes, V is the set of destination nodes and E is the set of edges. Every edge $e \in E$ from a $u \in U$ to a $v \in V$ represents the packet that u intends to transmit to v. Multiple packets from the same source node to the same destination node can be denoted as parallel edges. Fig. 2(a) shows an example of a 4×4 traffic matrix, where $T1$ to $T4$ are the source nodes, $R1$ to $R4$ are the destination nodes, and the number in the matrix denotes the number of packets from a source node that are to be transmitted to a destination node. Fig. 2(b) shows how the traffic matrix in Fig. 2(a) is modelled by a bipartite multigraph.

The bipartite multigraph G can be edge coloured to produce an assignment of colours to its edges such that the adjacent edges are assigned distinct colours. For example, the graph in Fig. 2(b) can be edge coloured with colours $c1$, $c2$, $c3$ and $c4$. This example is a minimum edge colouring, as it uses as few colours as possible to colour the graph. Minimum colouring is also known as optimal colouring. The collection of all edges that belongs to the same colour is called a colour class. For instance, the graph in Fig. 2(b) consists of four colour classes.

Each colour class is a matching that consists of a collection of pair wise non-adjacent edges. Thus, all the transmissions represented by the edges in a colour class can be carried out concurrently without collision using different wavelengths. When the number of wavelengths is not a constraint in the network, i.e., $W \geq N$, each colour class can be transmitted in a time slot. When the number of wavelengths is a constraint, i.e., $W < N$, each colour class can be divided into a number of time slots such that the number of edges in each time slot is not more than W. The collection and the specific order of these time slots is the transmission schedule. Fig. 3 shows the transmission schedule produced by edge colouring the graph in Fig. 2(b), assuming $W \geq N$.
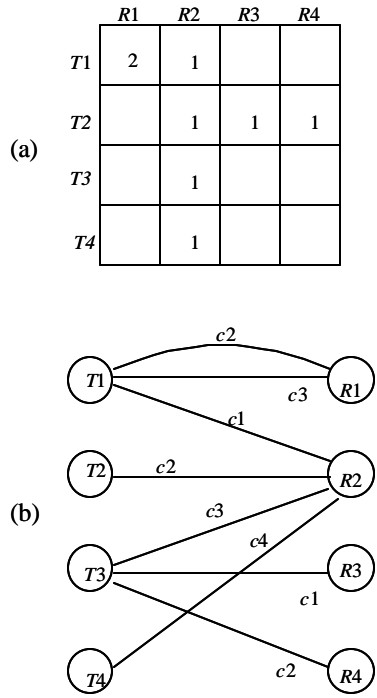
Fig. 2: (a) a 4×4 traffic matrix, and (b) the traffic matrix is modelled by a bipartite multigraph
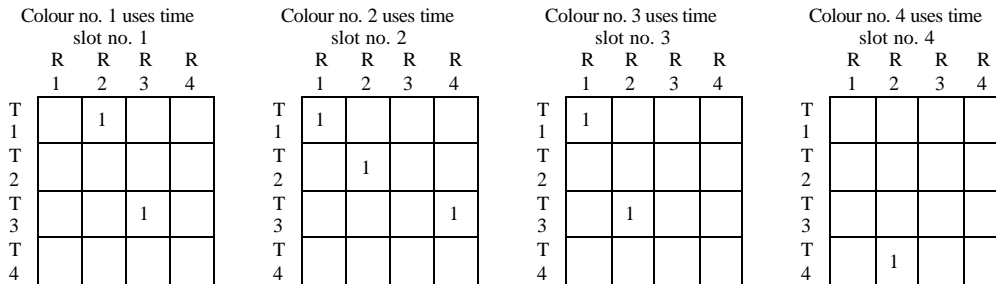

Fig. 3: The transmission schedule produced by edge colouring the graph in Fig. 2(b), assuming $W \geq N$

Since edge colouring bipartite multigraphs produce collision free transmission schedules, the problem now is to find a suitable algorithm that computes optimal colouring. Optimal colouring is desirable as it does not impose any extra or unnecessary length to the frame. Let D denotes the maximum degree of a bipartite multigraph G. It is well known that the chromatic index c of G is equal to D, i.e., $D = c$ [9]. As such, an optimal colouring of a bipartite multigraph is D-colouring.

There are a few efficient edge colouring algorithms for bipartite multigraph that have appeared in the literature [10, 11, 12]. *Colour-by-pair* algorithm [12] is adopted in this work due to its suitability for parallel processing. The time complexity of *colour-by-pair* is $O(M \log 2 \ N)$ in serial, where M is the number of packets used for scheduling. Meanwhile, the parallel algorithm based on *colour-by-pair* runs in $O(\log 3 \ N)$ time in parallel using $O(M)$ processors based on the *Parallel Random Access Machine* model of parallel computation [13].

2.3   Implementation

Using the parallel edge colouring scheduling algorithm, a major factor which must be considered for implementation is that the algorithm should use $O(M)$ processors in parallel. Thus, it is important to ensure the number of packets M selected in each cycle of scheduling is bounded as the number of processors in a parallel computer is fixed. Two

methods of implementation have been studied here, namely the variable frame size (VFS) scheme and the limited frame size (LFS) scheme.

The VFS scheme is very simple; it selects a fixed number of packets, say k packets from each node in each cycle of scheduling, thus $M \leq kN$. All these packets are scheduled by the scheduling algorithm. This scheme allows each node to transmit k packets in each cycle. Similarly, each node transmits $\dfrac{k}{f}$ packets in each time slot, where f is the frame size of the schedule, which depends on N, W and $\boldsymbol{D}$. Since there will be $\boldsymbol{D}$ colour classes produced by the scheduling algorithm, and each colour class consists a maximum of N packets, thus a maximum of $\dfrac{N}{W}$ time slots is sufficient to schedule each colour class. As such, $f \leq \boldsymbol{D}\dfrac{N}{W}$. While N and W are fixed, $\boldsymbol{D}$ varies depending on the traffic pattern of the network. Therefore, this scheme is unable to provide any QoS guarantee to the nodes, and supports only best effort transmission.

In view of the VFS scheme's inability to provide QoS guarantee due to the frame size variability, the LFS scheme is proposed to overcome this problem by ensuring $\boldsymbol{D}$ of the input graph is bounded. In this scheme, the scheduler still selects k packets from each transmitting node in a cycle of scheduling. It also ensures that a maximum of k packets are designated to each receiving node, thus $\boldsymbol{D} = k$. This can be easily done by maintaining a transmitter counter and a receiver counter for each node. When a packet is selected, the appropriate counter is increased by one. A packet will not be selected if the counter has reached the maximum limit of k. As with the VFS scheme, the total number of packets selected in each cycle is also bounded to kN. In this scheme, the input graphs are always k-colourable, thus $f \leq k\dfrac{N}{W}$, and the maximum frame size of LFS scheme $f_{\max}$ is $k\dfrac{N}{W}$.

Since the frame size is bounded, the LFS scheme can be used to guarantee CBR bandwidth to the nodes. Considering each node can transmit and receive the maximum of k packets in each cycle, thus the maximum bandwidth that can be reserved is also limited to k packets per cycle for both transmitting and receiving ends. Similarly, the maximum bandwidth that can be reserved to each node is k packets per $k\dfrac{N}{W}$ time slots, or $\dfrac{W}{N}$ packets per time slot. This bandwidth is the fair share of the data channels by nodes. For example, let the data channel transmit at 1 Gb/s, and $W = 20$ and $N = 100$. Thus, the maximum reservable bandwidth per node is:

$$B = \frac{W}{N} \times 1 \text{ Gb/s}$$
$$= 200 \text{ Mb/s}.$$

The bandwidth reservation is granted to the nodes as virtual circuits (VCs), and each VCs ranges from one to k packets per node per cycle. Multiple VCs can be granted to a node as long as the total reserved bandwidth to the node does not exceed the limit of k packets per cycle.

The CBR bandwidth guarantee can be implemented by maintaining a bandwidth allocation table. During a transmission setup, a new VC may be requested for reserving bandwidth. The scheduler will check the bandwidth allocation table if the requested rate plus the total rate allocated to the transmitting node is not more than k packets per cycle; then the scheduler will check if the requested rate plus the total rate allocated to the receiving node is also not more than k packets per cycle. If both conditions are complied, the new VC is added to the table. Otherwise, the request is rejected. Fig. 4 shows an example of bandwidth allocation table, where $T1$ to $T4$ denote the transmitters for node 1 to node 4 and $R1$ to $R4$ denote the receivers for node 1 to node 4. The (i, j) entries denote the allocated bandwidth in terms of the number of packets per cycle. Let $k = 8$, a new VC from $T4$ to $R2$ with the requested bandwidth of four packets per cycle will be granted as the total reserved bandwidth for both $T4$ and $R2$ is still not more than eight packets per cycle. Meanwhile, a new VC from $T4$ to $R3$ with any bandwidth will be rejected as the reserved bandwidth at $R3$ has reached the maximum limit of eight packets per cycle.

	*R*1	*R*2	*R*3	*R*4
*T*1	0	3	5	0
*T*2	4	0	1	0
*T*3	0	1	0	2
*T*4	0	0	2	0

Fig. 4: Bandwidth allocation table.

The packet selection process may be performed in two passes. In the first pass, all packets allocated with guaranteed bandwidth are selected. In the second pass, all others packets are considered at best effort basis as long as the transmitter counters and receiver counters are each still less than k. In such a selection, it ensures all packets with reserved bandwidth are considered while the remaining bandwidth is used for other packets. The bandwidth reservation allows VCs to enjoy a steady transmission, but if they miss a transmission opportunity due to idleness, no future compensation is given.

For both the VFS and LFS schemes, k must be properly chosen to ensure efficiency and effectiveness of the network. A large k causes delay in the network, while a small k causes inefficiency. From the scheduling algorithm's perspective, k can be any integer that is larger than one. However, the algorithm will be most efficient if k is an exact power of two, i.e., $k = 2^x$, with x being a positive integer. In fact, when k is an exact power of two, the time complexity of the *colour-by-pair* algorithm reduces to O(M log N), while the parallel algorithm based on the *colour-by-pair* reduces to O($\log^2 N$) in parallel using O(M) processors [11],[13].

3.0   SIMULATION MODEL

The proposed scheduling algorithm is written using standard C language and executed on a Pentium II 350 Mhz personal computer. The parameters used in the simulation are divided into two categories, namely, the system parameters and performance metrics.

3.1   System Parameters

The network consists of N=100 nodes and W=20 wavelengths. In each cycle, the scheduler selects k=10 packets from each node for scheduling. The buffer size is chosen to equal to f_{max}, which is 50 packets. The transmission speed of each data channel is assumed to be 1 Gb/s. The following parameters are varied in the simulations.

1)   Implementation scheme
      Both the VFS and LFS schemes are used for simulation.

2)   Reserved bandwidth
      When the LFS scheme is used for simulation, different percentages of reserved bandwidths can be adopted, ranging from zero to one hundred percent.

3)   Traffic Arrival Pattern
      Four different traffic arrival patterns considered for the simulation are as follows:

      •   Bernoulli traffic
           In each time slot, one packet may arrive independently from the previous arrival with probability p. The probability p is varied to generate different arrival rates. Bernoulli traffic is suitable to represent the non-bursty nature of traffic load.

      •   Poisson traffic

The inter-arrival time of the packets is exponentially distributed with rate p. As with Bernoulli traffic, p is varied to generate different arrival rates. Poisson traffic is suitable to represent traffics with some bursty nature.

- 2-state traffic
  Each node may be in one of two states, which is the bursting or resting state. While in the bursting state, a packet arrives at every time slot for the duration of its burst length, and then it changes to resting state. The burst length is varied exponentially and its average is chosen to equal to k. While in the resting state, no packet arrives. The state transition probability from resting to bursting is determined by the average burst length and p, where p is varied to generate different arrival rates.

- Reserved traffic
  The packets arrive according to the reserved rates granted. The reserved traffic is applicable only to the LFS scheme.

4) Arrival Rate

The arrival rate per node is chosen from the range of zero to the fair share of the network capacity, i.e., $\dfrac{W}{N}$.

For instance, the data channel is transmitting at 1 Gb/s, thus the arrival rate could be from zero to 200 Mb/s per node. An arrival rate higher than this range is not considered, as the total arrival rate will exceed the network capacity.

3.2   Performance Metrics

For any choice of the control parameters, the simulation is repeated for at least 200 cycles, and then the performance metrics are obtained. The following are the performance metrics considered in the simulation.

1) Blocking Probability
    A packet arrives from the higher layer and is stored in the buffer of the node before transmission. Blocking probability refers to the possibility that the packet is unable to be stored in the buffer due to the finite buffer space, and the packet is discarded. Thus, the blocking probability is obtained by dividing the number of packet lost with the total number of packets generated for the duration of the simulation.

2) Channel Utilisation
    Channel utilisation refers to the percentage of a data channel which is used for data transmission. The channel utilisation is measured as the percentage of the number of packets scheduled, divided by the number of time slots and W for the duration of the simulation.

4.0   RESULTS AND DISCUSSIONS

The VFS and LFS schemes are simulated for different traffic patterns, and then the results are compared for channels utilisation and blocking probability.

Figures 5, 6 and 7 show the channel utilisation of the VFS and LFS schemes for Bernoulli, Poisson and 2-state traffic, respectively. The results show that LFS scheme has higher utilisation compared to the VFS scheme especially during high traffic loads for all traffic arrival patterns. At lower loads, the number of packets in the buffer of the nodes would be limited. As such, when the scheduler schedules for transmission, it may not have enough requests to fill the frame. Therefore, the impact of the frame size was not seen and the performances of the two schemes were identical. However, when the arrival rate increases, reaching the mean service rate of the network, the VFS scheme begins to lag as compared to the LFS scheme. At heavy loads, due to the variability of the frame size for the VFS scheme, it tries to accommodate as many packets as possible in a cycle. Consequently, its cycle becomes longer and packets start to fill up the buffer space. Thus, when the buffer is full, packets would be dropped and this contributes to lower utilisations. As for the LFS scheme, even at higher loads, the frame size is limited. Therefore, the scheduler would be able to schedule faster than the VFS scheme. Thus, it will be able to transmit more packets in the same interval, and consequently, results in higher network utilisations. It can be seen in the plots that the LFS scheme enables more than 90% of network utilisation for any traffic pattern, except for 2-state traffic. This is evident that the scheme only spends minimal bandwidth for the overhead processing.
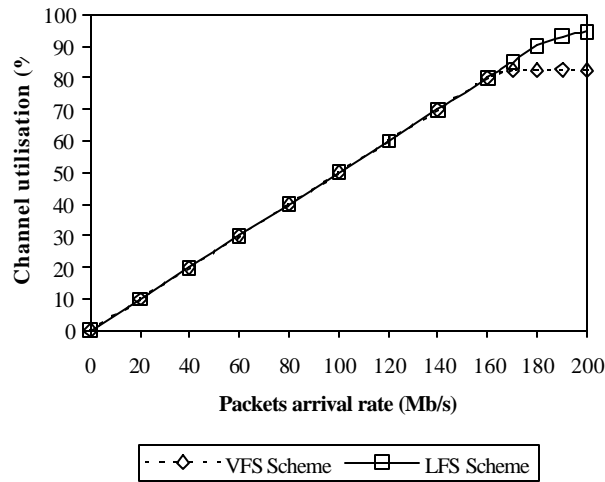
Fig. 5: Channel utilisation of the VFS and LFS schemes for Bernoulli traffic
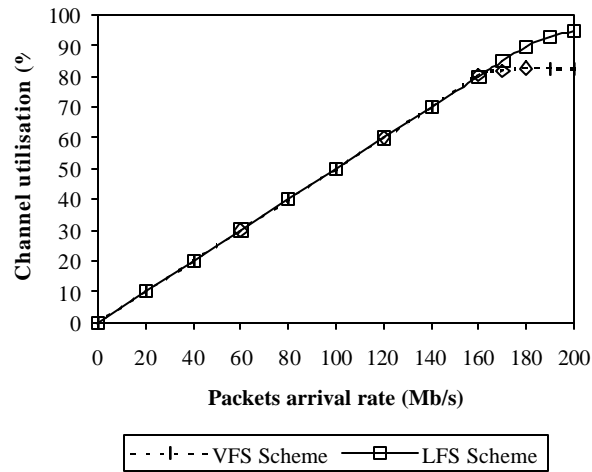

Fig. 6: Channel utilisation of the VFS and LFS schemes for Poisson traffic
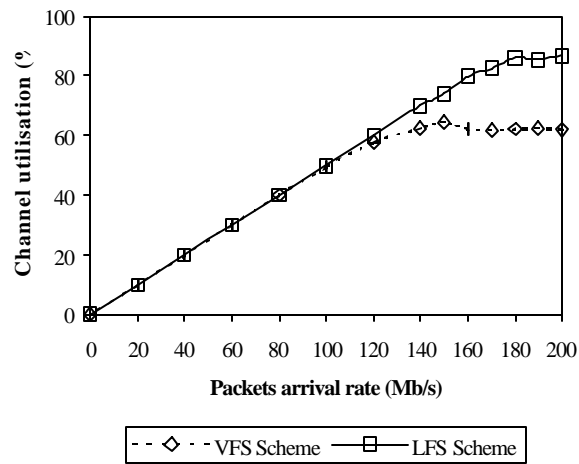

Fig. 7: Channel utilisation of the VFS and LFS schemes for 2-state traffic

Figures 8, 9 and 10 show the blocking probability of the VFS and LFS schemes for Bernoulli, Poisson and 2-state traffic, respectively. The results show that the LFS scheme has a lower blocking probability compared to the VFS scheme during high traffic load for all types of simulated traffic. It can be seen in the plots that the blocking probability of the LFS scheme is close to zero for most of the arrival rates. It starts to increase exponentially only when the arrival rate reaches about 90% of the mean service rate. These results are consistent with the previous results. As a matter of fact, the blocking probability metric is inversely related to the utilisation metric.
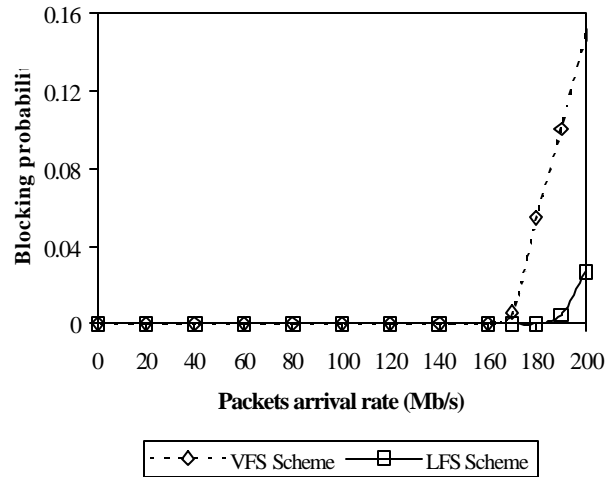

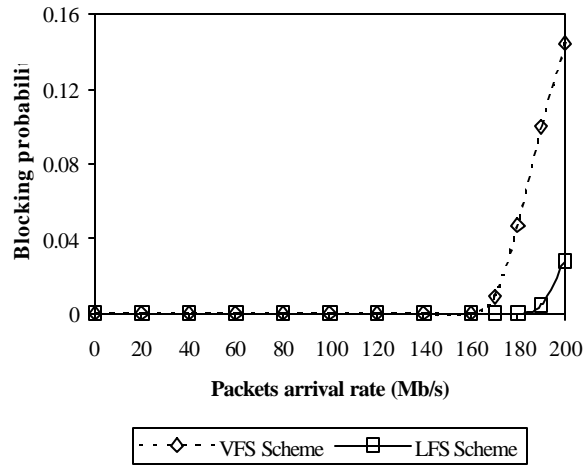Fig. 8: Blocking probability of the VFS and LFS schemes for Bernoulli traffic

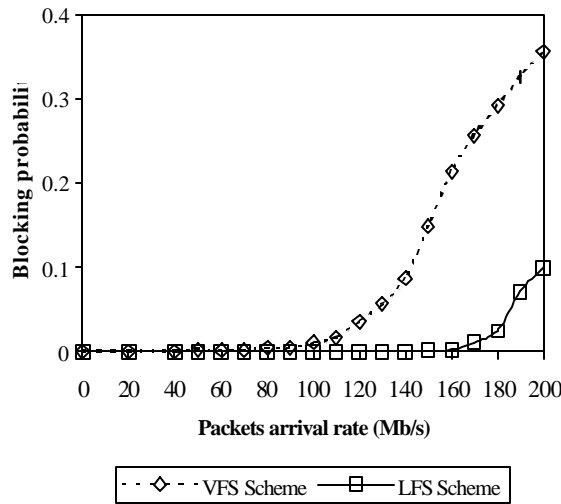Fig. 9: Blocking probability of the VFS and LFS schemes for Poisson traffic

Fig. 10: Blocking probability of the VFS and LFS schemes for 2-state traffic

Fig. 11 shows the blocking probability of the LFS scheme with different ratios of reserved traffic. The figure presents results of three different scenarios. In the first scenario, only reserved traffic is used, and the results show that the LFS scheme is able to guarantee any level of bandwidth guarantee without blocking. This is due to the nature of bandwidth reservation that the reserved bandwidth never exceed k packets for each node in each cycle such that the maximum degree D of the input graph is not more than k. Therefore, this enables all the reserved packets to be served in each cycle.
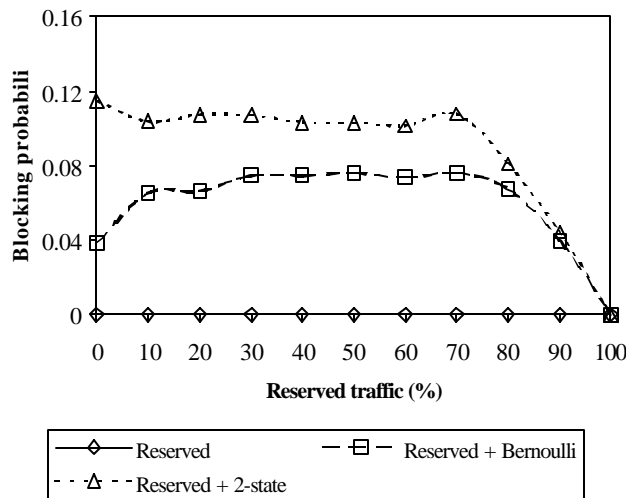

Fig. 11: Blocking probability of the LFS scheme with different ratios of reserved traffic

In the second and third scenarios, Bernoulli and 2-state traffic patterns are used to fill the unused bandwidths, respectively. The results show that 2-state traffic has higher blocking probabilities than Bernoulli traffic due to the bursty nature of the 2-state traffic. Meanwhile, the reserved traffic in the second and third scenarios is again observed without blocking. This is due to the nature of two-pass packets selection procedure, where all reserved packets are selected before the other packets.

Fig. 12 shows the blocking probability of the LFS scheme with 50% of reserved traffic and different arrival rates of Bernoulli and 2-state traffic. Again, the results show that the 2-state traffic has higher blocking probability due to the bursty nature. It is also observed that the blocking probability is higher compared to without reserved traffic. This is due to the transmission priority being given to CBR traffic and other packets are only served on best effort.
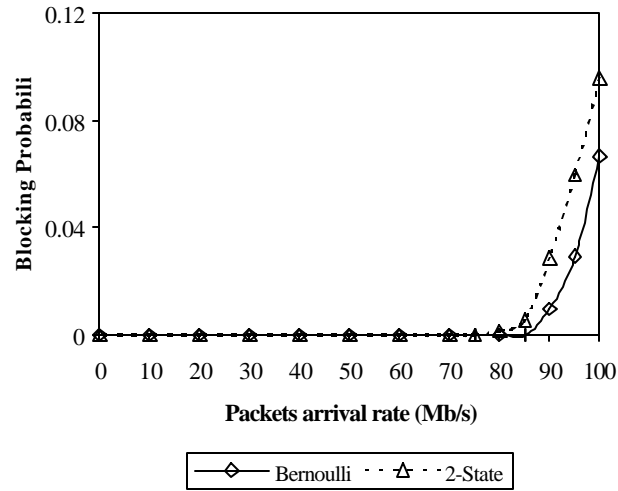
Fig. 12: Blocking probability of the LFS scheme with 50% of reserved bandwidth and different arrival rates
of Bernoulli and 2-state traffic

5.0   CONCLUSION

In this paper, a scheduling algorithm for WDM optical networks has been proposed. The algorithm capitalises on the unique features of the WDM networks. It is collision free due to the nature of the edge colouring technique. The algorithm also supports provision of CBR guarantee up to the fair share of the network capacity of each node. The algorithm has a small running time complexity, which is $O(M\log^2 N)$ in serial. Moreover, the algorithm can be executed in a parallel computer, which would improve the time complexity to $O(\log^3 N)$ in parallel using $O(M)$ processors.

Two different approaches of implementation have been studied. The simulation results show that the LFS scheme performs better than the VFS scheme in terms of the channel utilisation and blocking probabilities, especially at heavy loads. Besides, the LFS scheme is able to respect any level of CBR QoS guarantees, while the unused bandwidth can be used for best effort transmissions. The LFS scheme also spends minimal bandwidth in overhead processing and enables more than 90% of network capacity be used for data transmission for most traffic arrival rates and patterns.

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BIOGRAPHY

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