DYNAMIC BANDWIDTH ALLOCATION IN ATM NETWORKS

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Declaration

I declare that this thesis is my own work. Where collaboration with other people has taken place, or material entered by other researchers is included, the parties and/or material are indicated in the acknowledgement or references as appropriate.

This work is being submitted for the Degree of Doctor of Philosophy in Electrical Engineering at the University of Cape Town. It has not been submitted to any other university for any other degree or examination.

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Date

Synopsis

This thesis investigates bandwidth allocation methodologies to transport new emerging bursty traffic types in ATM networks. However, existing ATM traffic management solutions are not readily able to handle the inevitable problem of congestion as result of the bursty traffic from the new emerging services. This research basically addresses bandwidth allocation issues for bursty traffic by proposing and exploring the concept of dynamic bandwidth allocation and comparing it to the traditional static bandwidth allocation schemes.

In this research work we suggests the available bandwidth be allocated dynamically in order to control the burstiness and consequently improve overall network utilisation. This goal is achieved by employing two approaches. In the first approach, the adaptability of bandwidth allocation is achieved by seeking the required extra bandwidth to accommodate the incoming bursts on the same transmission path that was established during the call setup phase. The traffic source used in the renegotiation-based dynamic bandwidth allocation protocol is scalable compressed video on multilayer format. The proposed protocol periodically renegotiates the assigned bandwidth. The protocol uses on-line predictors to forecast the subsequent and future traffic requirements. The prediction is derived from measurements of the traffic that has already been observed prior to the current connection. The proposed protocol assumes that the traffic source can accept a graceful degradation of QoS in the case of network congestion when the required bandwidth cannot be provided.

The second approach is based on multipath routing strategy and does not involve any renegotiation of QoS parameters. The multipath routing-based dynamic bandwidth allocation protocol searches for the required extra bandwidth to accommodate the incoming burst on alternative routes in parallel basis. With this protocol, the bandwidth is allocated and managed dynamically at multilevels (i.e. call, burst, and cell levels).

The performance of the renegotiation-based dynamic bandwidth allocation protocol is evaluated and compared with traditional static bandwidth allocation schemes by simulation using MPEG-4 traces. The simulation results show a significant improvement in network performance and utilisation for our protocol in comparison to the traditional static bandwidth allocation schemes. The performance of the multipath routing-based dynamic bandwidth allocation protocol is evaluated analytically. The analytical results show that the burst blocking probability could be reduced when more than one path is added to the original path and searched for extra bandwidth to accommodate the incoming burst in parallel. While the multipath routing-based dynamic bandwidth allocation approach improves network performance and utilisation by controlling the traffic burstiness it however will suit non-interactive applications because of routing costs on *burst-by-burst* basis. However, the renegotiation-based dynamic bandwidth allocation approach works for real time applications if the renegotiation interval is carefully tuned. This is because the renegotiation-based dynamic bandwidth allocation protocol does not incorporate any routing or housekeeping records at burst level.

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Chapter 1

Introduction

The emergence of high-speed optical transmission is likely to fuel an already growing demand for interactive image communications, multimedia applications, and real time video services, including video conferencing, TV, and High Definition TV (HDTV). Coupled with advances in storage technologies, high–speed communication networks can be used to build "on-demand" services that are expected to penetrate residential, organizational, and educational premises in a manner similar to existing cable TV or telephone networks. Further, computing environments are migrating fast from centralized to distributed processing; high-speed networks will allow multiple computing resources to be treated as a single system rather than a network of computer systems.

The diversity of traffic types in distributed multimedia applications introduces hard problems on the underlying network. No ultimate solution has yet been reached for providing integrated services to multimedia applications. The difficulty lies in the nature of the multimedia traffic, basically video, which is highly bursty. The key success of high-speed networks will be their ability to support a diverse range of applications on a single platform.

The single platform that can transport a variety of traffic applications is a Broadband Integrated Service Digital Network (B-ISDN). There are couple of reasons why B-ISDN will be eventually needed. First, all of the above mentioned applications currently either require dedicated networks or allow only one application at a time. The introduction of broadband integrated networks will lead to a consistent user interface and will simplify operation and maintenance for network service providers. Second, new services cannot always be accommodated by an existing specialized network and therefore require a new network to be installed before they can be offered. The flexibility of B-ISDN ensures, given sufficient capacity, that new services can easily be implemented without costly network hardware.

Traffic management and control of B-ISDN present numerous challenges, which must be overcome before their full potential can be realized. Controlling access to network resources is essential to guarantee quality of service for each application. High transmission speeds render the traditional approaches based on reactive control mechanisms only of little use.

1.1 Asynchronous Transfer Mode Networks

The promise to provide multimedia applications with diversity in the quality of service (QoS) requirements in terms of loss, delay and service cost is the challenge of B-ISDN. Asynchronous Transfer Mode (ATM) has been chosen by ITU-T as the transport mechanism for B-ISDN [12][13][14]. ATM technology is expected to provide an acceptable, cost-effective means for meeting the requirements of future broadband networks. The ATM technique is sufficiently versatile and flexible to provide a bearer service capable of integrating a wide spectrum of traffic sources. These sources exhibit a diverse mixture of traffic characteristics with different 'correlations' and 'burstiness' parameters. ATM-based networks are also well suited for high-speed LANs.

Specifically, ATM is a high-speed, virtual circuit-oriented packet-switching technique that uses short data packets of a fixed length of 53 bytes, called cells[1]. Five of the 53 bytes are reserved for the cell header. All the nodes on the network are connected via one or more ATM switches, which route the cells to their various destinations. A virtual circuit in ATM-based B-ISDN is established using a traffic contract between the network and customer to deliver traffic of specified statistical characteristic to the destination with a specified quality of service. Fixed length cells simplify the design of an ATM switch at the high switching speeds. The selection of the short fixed length reduces delay and more significantly jitter for delay-sensitive applications, such as voice and video. In addition, it leads to considerable simplification in hardware processing and in buffer and queue management. Hence, minimal functionalities are required in the switching nodes, thus facilitating switching with low delay[2].

1.2 ATM Statistical Multiplexing Gain

ATM networks provide an efficiency gain through statistical multiplexing of dynamic user traffic. A necessary number of cells are allocated on demand, which means that bursty applications could be statistically multiplexed. This will result in bandwidth-sharing and, consequently, a higher resources utilisation and lower cost per bit of transmission capacity. The statistical multiplexing also could lead to a severe traffic congestion unless optimal congestion control mechanisms and bandwidth allocation schemes are implemented. Consider, for example, two connections that merge at one output queue in an ATM switch. The bandwidth requirement of each connection fluctuates, producing peaks and troughs in the aggregate bandwidth demand. To achieve an efficient multiplexing gain, the ATM switch allocates an aggregate bandwidth that is less than the sum of the peak cell rates. This means that the entire bandwidth is divided among the network nodes active at any time. This enables ATM networks to implement services with variable bandwidth requirements, such as applications with high varying bit rates, real-time applications, fixed bit rates and time-critical applications, with a high degree of efficiency. However, the "statistical multiplexing" gains do not come without their associated costs. Cells have to be buffered at the ATM switches and may have to be dropped as a consequence of buffer overflow as shown in Figure 1.1[26]. Further, statistical multiplexing introduces queuing delays at intermediate nodes; cells reaching the destination after excessive delays may have to be discarded. Furthermore complicating the picture is "jitter", i.e. the variability in the delay of successive cells belonging to the same source. An increase in delay jitter will increase the number of cells requiring buffering at the destination, causing buffer overflow.

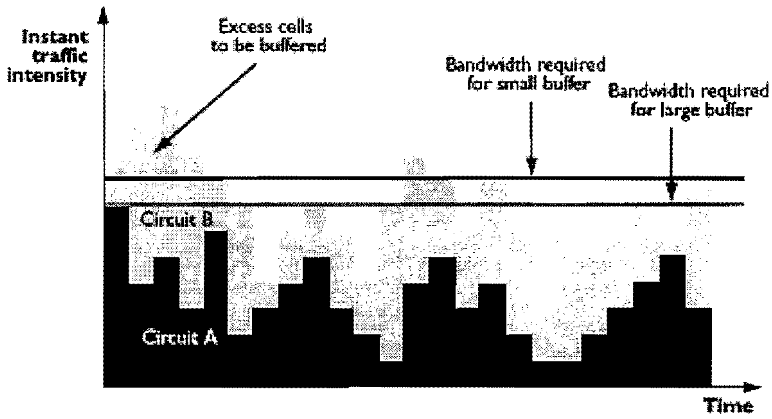


Figure 1.1: Statistical multiplexing gain [26]

1.3 ATM Service Categories

In the ATM standard [1], several QoS classes are currently defined, as illustrated in Table 1.1. Constant bit rate (CBR) and variable bit rate (VBR) are classes that give performance guarantees to the application, providing that the application complies with parameters on the traffic contract. For CBR applications, the negotiated parameter is the peak cell rate (PCR). For VBR applications, the negotiated parameters are the PCR, the sustainable cell rate (SCR), and the maximum burst size (MBS). The network for CBR and VBR services can allocate only a predetermined portion of the link bandwidth. Available bit rate (ABR) applications use the residual bandwidth not allocated to CBR and VBR services. Since residual bandwidth fluctuates over time, ABR applications must be able to control their transmission rates.

The unspecified bit rate (UBR) class is a "best effort" service, where no guarantees are given and the applications use the bandwidth not currently in use by CBR or VBR connections. However, CBR transmission is easier to implement from the network management point of view, but at cost of variable quality of service, and/or at the cost of network efficiency. Compressed video is variable bit rate (VBR) type in nature, and hence, it is efficient at transmitting compressed video streams by VBR service with a guaranteed quality of service.

Table 1.1: ATM service classes category

Attribute	ATM Layer Service Category				
	CBR	rt-VBR	nrt-VBR	UBR	ABR
Traffic Parameters:					
PCR,CDVT$_4$	Specified			Specified$_2$	Specified$_3$
SCR, MBS, CDVT$_{4,5}$	n/a	Specified		n/a	
MCR$_4$	n/a			n/a	Specified
QoS Parameters:					
Peak-to-Peak CDV	Specified	Specified	Unspecified	Unspecified	Unspecified
Maximum CTD	Specified	Specified	Unspecified	Unspecified	Unspecified
CLR$_4$	Specified			Unspecified	See Note 1
Other attributes					
Feedback	Unspecified			Unspecified	Specified

Notes:

1. Cell Loss Ratio (CLR) is not signalled for ABR because it has a low value.
2. Should not be subject to Call Admission Control (CAC) and Usage Parameter Control (UPC) procedures.
3. Represent maximum source rate.
4. Parameters are either explicitly or implicitly specified for Permanent Virtual Connections (PVCs) or Switched Virtual Connections (SVCs).
5. Cell Delay Variation Tolerance (CDVT) is not signalled.

1.4 Traffic and QoS Parameters

In ATM networks applications are often allowed to customise their required QoS parameters. Each ATM connection may request a different QoS parameter than that specified on the five ATM service categories explained on Table 1.1. The negotiation of these QoS parameters is a function of provisioning (for PVCs) or signalling (for SVCs). So far, there has not been a good way to specify the bursty traffic, like video. One way to specify the traffic is to use leaky bucket parameters, such as burst length, sustained bit rate, and peak bit rate. Another way is to use traffic models to describe traffic. Traffic parameters, however, are important to the service providers to allow them to allocate resources and calculate charges. There are three QoS parameters:

• Cell Loss Ratio (CLR)

• Maximum Cell Transfer Delay (maxCTD)

• Cell Delay Variation (CDV)

CLR is the ratio of lost cells to total transmitted cells. Cells may be lost due to an ATM switch malfunction, but cells are usually lost because the ATM switch explicitly discards them. An ATM switch will discard cells that belong to noncompliant traffic flows at the network edge switches, or in response to network congestion. The method by which an ATM switch discards cells in the face of congestion is critical to a network's performance. A switch may discard cells in such a way that the congestion is not resolved or is increased. A discard algorithm based on efficiency and fairness provides optimal traffic performance.

CTD is defined as the elapsed time between a cell exit event at the source and the corresponding cell entry event at the destination. CTD is the sum of the total inter-ATM node transmission delay and the total ATM node processing delay between source and destination. End systems using CBR or VBR-rt service indicate their CTD requirement by negotiating with the network a maxCTD.

CDV is sometimes called cell jitter. It refers to the fact that while cells may be sent into a network evenly spaced, a variety of factors may contribute to cell clumping and gaps in the cell stream. CDV is mostly an issue for end systems running voice, video, and multimedia applications over CBR and VBR-rt connections. If the network cannot properly control CDV, some of these real-time services may have their communications distorted. An end system using CBR or VBR-rt service indicates its end-to-end cell delay variation requirement by negotiating a peak-to-peak

CDV with the network. Figure 1.2 explains QoS requirements for different ATM traffic applications.

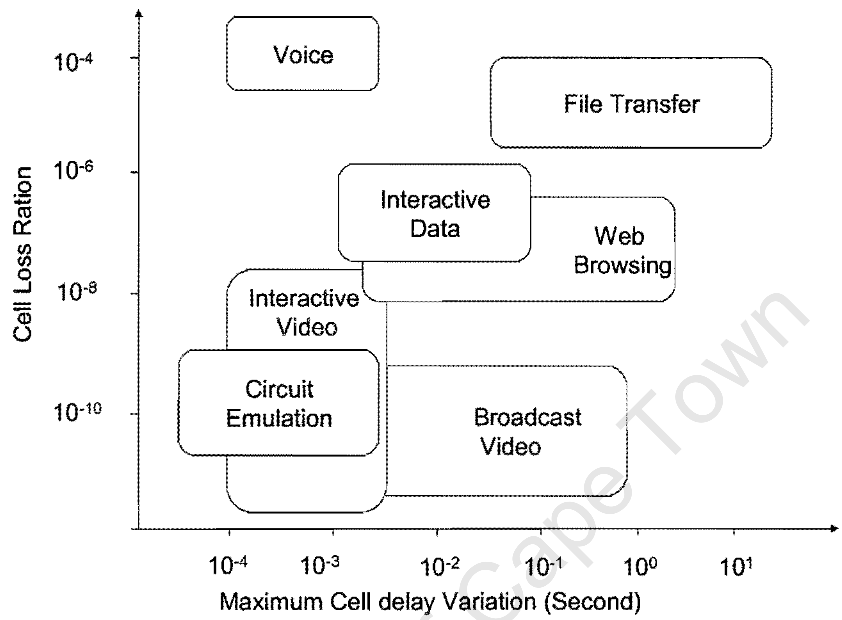


Figure 1.2: QoS requirements for different traffic applications

1.5 Background on Bandwidth Allocation in ATM Networks

Modern high speed integrated services digital networks have to support applications with diverse traffic characteristics and performance requirements. Real-time voice and video applications generate network traffic that requires per-connection performance guarantees in terms of throughput, delay, delay-jitter, and cell loss rate. To guarantee such performance requirements, the network must be able to control its resources tightly. Generally, traffic patterns generated by data sources and VBR real-time applications such as compressed video and audio, tend to exhibit burstiness over multiple time-scales. Correspondingly, resource allocation algorithms should have mechanisms at different time-scales to achieve an acceptable level of utilisation [27][28][29][30].

Providing deterministic services for real-time traffic necessitates bandwidth allocation according to a worst-case scenario. Generally, a time-invariant worst-case constraint function that bound the arrival process is used to characterize the traffic stream. Because of the long-term variations in real-time traffic, this time-invariant function is not representative of the whole stream. Fixed bandwidth allocation that is based upon this function will not achieve acceptable utilisation.

To compensate burstiness on long time-scales, several renegotiation and prediction schemes based upon changing the allocated resources when the traffic rate changes over long term were proposed [3][24][25]. However, these schemes can provide only statistical performance guarantees.

In [10],[18] they have studied dynamic bandwidth allocation based on measured cell loss ratios (CLR). The measurement time interval of these methods is inversely proportional to the CLR. When the target CLR is small, a very large time window will be required to collect the statistics for CLR estimation. For example, measuring a CLR of 10^{-5} with meaningful statistics requires at least 10^{7} cell arrivals to be monitored. However, it is not feasible to use very large time windows to dynamically allocate bandwidth since the traffic conditions may have changed significantly before accurate measurement can be obtained. Furthermore, their mechanisms do not control short-term QoS violation, which may lead to poor picture quality in video transmission for a short period of time.

In order to validate whether the network is providing the QoS that it has committed to a connection, the transfer characteristics of the network are measured either on a single ATM cell or a sequence of ATM cells during the lifetime of an ATM virtual connection. The results of this measurement are compared against the ATM QoS parameters included in the traffic contract. At this point, it is important to realize that these QoS parameters are statistical in nature, i.e. they are measured over a sample of cells that is appropriately large for the particular QoS parameter. It is equally important to note that while the CLR, CTD, and CDV QoS parameters can be negotiated, the Cell Error Ratio (CER) and Cell Misinsertion Ratio (CMR) QoS parameters are not negotiated. The difference is that it is possible for the negotiated QoS parameters to be different among connections depending on how the connections are handled. However, the non-negotiated QoS parameters are dependent on the inherent characteristics of an ATM network and are therefore the same across all virtual connections, i.e. they cannot be changed during the operation of the network. Depending on the capability of individual implementation of end-systems, QoS can be negotiated in terms of individual numeric parameters or at least QoS classes. Whichever way is used, it is very important for applications, especially video applications, to receive an appropriate QoS from the network. From the point of view of an end-user or a video service provider, if the QoS requested for a video connection is too strenuous (i.e. the QoS requested is well above what is appropriate), the connection request may be refused due to a lack of enough resources in the network. Even if there is enough resources in the network, the connection may have to pay more due to the strict QoS. On the other hand, if the QoS request is too loose (i.e. QoS is below what is required), the quality of video delivered at the destination may be unsatisfactory. From the point of view of a network provider, if the QoS of many connections is too strict, network resources will not be utilised efficiently. As a result, the network will not be able to provide services to as many users as it should and, depending on the pricing strategy, the revenue generated, might decrease as a result. Conversely, an unsatisfactory network service will be provided if the QoS is too low. Therefore, in order to utilise the network resources a dynamic bandwidth allocation is used to provide extra resources for bursty traffic sources. This is can be achieved either via renegotiation of QoS parameters as proposed in Chapter 5 or seeking the required resources on multipath routing as proposed in Chapter 7.

1.6 Problem Definition and Motivation

The majority of traffic in broadband networks is likely to be generated by video applications, whether it is for interactive videophone conversations or videoconferencing, consultation of pre-recorded video sequences in multimedia databases, or simply watching a movie. However, the existing ATM traffic management solutions are not ready to handle the congestion problem of such bursty traffics. Variable bit rate video over ATM faces video quality degradation problems due to loss at congestion. Preventive traffic control is based on the notion of a traffic contract [21]. A requested connection is described by means of a set of traffic parameters, and the network must decide if it can accepts the connection without violating quality of service constraints. To ensure that the traffic characteristics of an accepted connection are as declared, the network must police the traffic parameters. The research in this area also has proven that the usage of cell priorities provides better congestion and loss control. The current ATM traffic management model supports only two levels of priority classes [31], which are application independent and completely network performance oriented. Only a single Cell Loss Priority (CLP) bit in the cell header is allocated for priority information. ATM constant Bit Rate (CBR) service is used for loss prioritisation for VBR video.

It proves particularly difficult to identify traffic parameters for variable bit rate connections, which are both, significant in determining required network resources and utilise the network resources efficiently. In accordance with the ATM Forum specification [1], certain traffic parameters are used to setup the traffic contract during the call setup phase. However, it is very difficult to characterize a bursty traffic like a long video sequence with only a few traffic descriptors. Owing to the unpredictable nature of bursty traffic, these traffic descriptors are expected to be imprecise. To take care of this problem, one can reserve resources for the worst-case scenario to guarantee the QoS but this results in low utilisation of network bandwidth [5]. To improve the utilisation of the network, a proper bandwidth allocation and management scheme is needed.

The bandwidth allocation task was traditionally done for the entire duration of the connection at call level (call setup phase) only. In this case, the user must specify its requirements from the network in terms of bandwidth, and QoS parameters. The network uses these requirements to allocate adequate resources for the connection. However, when the traffic offered to the connection is bursty, the widely varying nature of bursty traffic indicates that a simplistic burst reservation mechanism would not suffice [22][23].

However, the problem with the above-mentioned bandwidth reservation (allocation) schemes is that unless reservation success probability is high, the schemes aren't helpful. The main objective of our research is to develop a new ATM traffic management model that will handle the new emerging bursty traffic. Therefore, this research addresses the above problem of bandwidth allocation for bursty traffic by proposing a dynamic allocation of bandwidth to achieve efficient utilisation of network resources. This goal has been achieved in two approaches. In Chapter 5, a dynamic bandwidth allocation was achieved by seeking the required resources on the same transmission path which has been established during call setup phase. The bandwidth allocation protocol is based on a periodic renegotiations of the traffic parameters during connection lifetime. However in Chapter 7, we chose another approach to achieve our goal by seeking the required bandwidth on alternative routes by tuning the input traffic for a given bandwidth using a multipath routing strategy.

1.7 Published work

Original work that we present in this thesis has been published at various International and national conferences. These publications are:

[1] M. Ashibani, and B. Nleya, "Burst level Real time bandwidth allocation schemes in ATM networks," *SATCAM 2000*, Stellenbosch, South Africa, September 2000.

[2] M. Ashibani, and B. Nleya, "Burst control schemes in ATM networks," *SATCAM 2000*, Stellenbosch, South Africa, September 2000.

[3] M. Ashibani, and B. Nleya, "Real time dynamic flow control in ATM Networks," *SATCAM 2000*, Stellenbosch, South Africa, September 2000.

[4] M. Ashibani, and B. Nleya, "Design of fault tolerant networks," *Proceedings of International Conference on Communication System, IEEE ICCS*, Singapore, November 2000.

[5] M. Ashibani, D. Mashao, and B. Nleya, "Multi level Real time Bandwidth Allocation Scheme for ATM Networks," *IEEE, International Conference on Trends in Communications, (EUROCON2001)*, Slovakia, July 2001.

[6] M. Ashibani, D. Mashao and, B. Nleya, "Performance Analysis of Burst level Bandwidth Allocation Using Multipath Routing Reservation," *IEEE, International Conference on Trends in Communications, (EUROCON2001)*, Slovakia, July 2001.

[7] M. Ashibani, B. Nleya, and D. Mashao, "Renegotiation-Based Quality of Service Approach for Scalable Compressed Video in ATM Networks," *SATNAC 2001*, South Africa, September 2001.

[8] M. Ashibani, B. Nleya, and D. Mashao, "Bandwidth Renegotiation Using Hierarchical Compressed Video in ATM Networks" *IEEE 8th Symposium on Communications and Vehicular Technology, SCVT 2001* Netherlands, October 2001.

[9] M. Ashibani, B. Nleya, and D. Mashao, "Dynamic Bandwidth Allocation Protocol for Hierarchically Compressed Video in ATM Networks," *IEEE, International Conference on communications, ICC 2002*, USA, April 2002.

[10] M. Ashibani, B. Nleya, and D. Mashao, "Multilayer Video Transmission System for ATM Networks", *International Symposium on Communication Systems, Networks and Digital Signal Processing, CSNDSP2002*, Staffordshire, UK, July 2002.

[11] M. Ashibani, B. Nleya, and D. Mashao, "Performance Evaluation of Dynamic Bandwidth Allocation Scheme for VBR Video Streaming in ATM Networks," *Accepted at IEEE Region 8 (AFRICON'02)*, George, South Africa, October 2002.

[12] M. Ashibani, B. Nleya, and D. Mashao "Multilayer Video Transmission Model for ATM Networks," *To appear at International Telecommunication World Conference,* WTC 2002, Paris, France, September 2002.

[13] B. Nleya, M Ashibani, and R. Sewusankar, "An overview of Intelligence in DWDM Based Optical Networks," *To appear on IEEE AFRICON'02,* George, South Africa.

1.8 Thesis Outline

Chapter 2 presents an overview on traffic control and bandwidth management in ATM networks. Several network traffic control issues related to our research are presented. It also presents an overview of some ATM traffic control mechanisms and their solutions in the management of bursty traffic over an ATM network. In

Chapter 3, we address the problem of VBR traffic modelling. However, in this thesis we also recognize the importance of understanding traffic statistics and studying the impact that different traffic sources have on quality of service. Any effort to develop a bandwidth allocation and management scheme for multimedia networks has to be preceded by a though study of the traffic that these networks will carry. This is particularly true for VBR video traffic, which has been shown to be highly bursty and significantly more difficult to characterize than traditional voice and data traffic. Since these traffic streams have a complex structure, their effect on network performance may be much more complex than that predicted by simplistic traffic models. Thus, a careful performance analysis effort is required for accurate modelling of multimedia applications via both analysis and simulation.

Chapter 4 provides an overview on transmission of compressed video over ATM networks. We begin with a short history of MPEG video compression standards and proceed to discuss the basic structures of MPEG coding, such as quantisation and interframe prediction. This chapter presents a comprehensive discussion on transportation of MPEG-2 over ATM. The ITU-T has adopted MPEG-2 for video transmission over different ATM AAL layers. The content-based compression technique is investigated also in this chapter where MPEG-4 is a good example for that. MPEG-4 achieves the goal of having a scalable coding of video. Thus a deep understanding of multilayer or hierarchical encoded technique is necessary because the main assumption of the proposed scheme in Chapter 5 is that the video source is compressed on scalable or multilayer format. Such scalable encoding will support receivers with different bandwidth and display capabilities.

In Chapter 5, we have proposed a dynamic bandwidth allocation protocol based on renegotiation of QoS traffic parameters. In this proposal the bursty traffic source is a multilayered compressed video. A multilevel traffic descriptor (profile) is proposed in this chapter, where the traffic source describes its QoS parameter on multilevel format according to the recent Q2963 ITU-T signalling protocol. Chapter 6 evaluates the performance of the renegotiation-based dynamic bandwidth allocation protocol. A number of simulation experiments using MPEG-4 traces have been conducted to evaluate the performance of the proposed protocol. We start with description of the MPEG-4 traces used in the simulation, and then we proceed to present the performance results in terms of cell loss rate, throughput, resources utilisation, and data-rate SNR. The impact of renegotiation frequency on network performance is evaluated as well. Chapter 7, however, proposes a multipath routing-based dynamic bandwidth allocation protocol. This protocol manages network bandwidth at call, burst, and cell levels to efficiently utilize network resources for bursty traffic. The performance of multipath routing-based dynamic bandwidth allocation is evaluated analytically in this chapter. Chapter 8 presents conclusions and recommendations for future work.

Chapter 2
Congestion Control and Bandwidth Management

The task of ATM congestion control is to take various measures to minimize the extent and duration of congestion episodes. The traffic control function is designed to ensure that such situations are avoided as far as possible. This is achieved by optimising the usage of existing network capacity. This chapter covers several network traffic control issues that are related to our research. It also, presents an overview of some of ATM traffic control mechanisms and their solutions in the management of bursty traffic over an ATM network.

Traffic control functions or mechanisms are required for the implementation of traffic monitoring and congestion control in ATM networks. They can be defined as a set of network actions that monitor and control the flow of traffic to avoid congestion. Congestion is defined, as a phenomenon where the amount of traffic injected into the network is more than the current handling capacity of the network, resulting in possible overflow of buffers in the network. The concept of traffic control involves methods of fairly and optimally sharing network resources and balancing different traffic loads by satisfying the required quality of service (QoS) parameters and traffic definitions. It must provide each traffic stream with its desired level of bandwidth and QoS.

Congestion control is one of the primary mechanisms of realising traffic control and is the most challenging as far as traffic control is concerned. The primary roles of a congestion control procedure are to regulate the traffic in order to obtain high network utilisation, avoid network congestion, and provide an acceptable and consistent QoS to the user at low service cost [31][33].

Achieving network efficiency implies operating the network at or near capacity approaching congestion. When a network approaches congestion, it is functioning efficiently and achieving high utilisation. The challenge facing an ATM traffic control is to maintain QoS in the face of high utilisation. A proper traffic control is the key to meeting that challenge.

2.1 Traffic and Congestion Control Functions

A number of ATM traffic and congestion control functions have been defined by the ITU-T and the ATM Forum to support the standard service categories [31][33]. The functions provide a framework for monitoring and controlling traffic and avoiding congestion. Congestion control can be achieved using either open loop or closed-loop schemes. The open-loop scheme is based on designing and configuring the system carefully to avoid the occurrence of congestion; it is basically a preventive approach. However, the closed-loop scheme, which is a reactive approach, is based on feedback from the network to the ABR sources to slow down. ABR sources have traditionally been data traffic. While both approaches have advantages and disadvantages, a trade-off between buffer resources and delay (or overhead, complexity, and cost) must be considered on the traffic control mechanism chosen. However, in ATM networks, a combination of these two approaches is currently used in order to provide effective congestion control. For instance, CBR and VBR services use preventive open loop schemes and the ABR service is based on a reactive closed loop scheme.

The following functions are provided for the implementation of traffic monitoring (control) and congestion control in ATM networks [91]:

1- Traffic control functions:

* Network resource management.
* Connection admission control (CAC).
* Usage/network parameter control (UPC/NPC).
* Priority control and selective cell discarding.
* Traffic shaping.

* Fast resource management (FMR).

2- Congestion control functions:

* Selective cell discarding.

* Explicit forward congestion indication.

The interaction between the above mentioned traffic control functions in achieving the goal of a non-congestional broadband ATM network are illustrated on Figure 2.1.

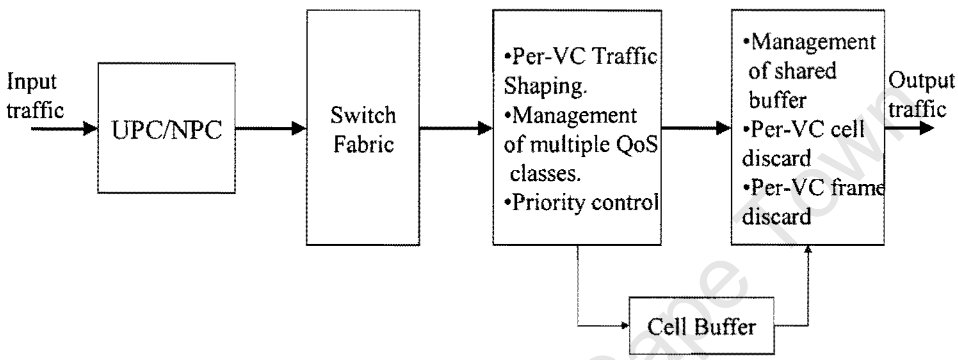


Figure 2.1: Traffic control functions

2.1.1 Management of Network Capacity

The management of network capacity is mainly implemented by means of path management. This allows the switching requirements for the setting up of path connections to be reduced by reserving transmission capacity in paths.

The end-to-end QoS for a given connection is directly dependent on the quality of the series of paths in which the virtual circuit is located [91]. If various virtual circuits are routed via the same path, they will have similar performance and QoS parameters, such as cell loss rate and cell transfer delay. Virtual circuits with similar QoS parameters should therefore be routed over same ATM path by the traffic control. If the virtual circuits (VC) within a path require different QoS classes, the class with the strictest requirements can be used for all the VCs. This saves on management input but will come at the expanse of network utilisation.

2.1.2 Call Admission Control

Call Admission Control (CAC) can be defined as the set of actions taken by the network during the call setup phase to establish whether a VC/VP connection can be made. A connection request for a given call can only be accepted if sufficient network resources are available to establish the end-to-end connection maintaining its required quality of service and not affecting the quality of service of existing connections in the network by this new connection. This involves checking the traffic contract. An ATM traffic contract specifies the characteristics of a connection, as defined by a connection traffic descriptor and a QoS [31][33]. The traffic contract is an agreement between an ATM end-system and the ATM network. As long as the end-system sends traffic across the User Network Interface (UNI) in conformance with the connection traffic descriptor, the network will enforce the negotiated QoS [82][83]. The connection traffic descriptor includes two key elements: the Cell Delay Variation Tolerance (CDVT) and the source traffic descriptor. The source traffic descriptor is a set of parameters that describe the connection's expected bandwidth utilisation. These parameters are: Peak Cell Rate (PCR), Sustainable Cell Rate (SCR), Maximum Burst Size (MBS), and Minimum Cell Rate (MCR). With these parameters, an ATM end-system describes its average and peak bandwidth requirements and its maximum burst size. The network determines whether it can establish the connection and still meet the required QoS level for all connections including the new one. This calculation is referred to as Call Admission Control (CAC). Conformance is determined by a Usage Parameter Control (UPC) function at the network edge switch as will be explained in the next sections.

The CAC question turns out to be complicated, as the issue of call admission control (CAC) is closely related to the other aspects of a network, such as service models, scheduling disciplines, traffic characterization, and QoS specification. A key challenge in the design of a CAC algorithm is to support a large number of sessions with different performance requirements, while minimizing cost as measured by network resources.

The call set up procedure runs on a resource manager, usually a workstation attached to the network node. This resource manager controls the operations of the network node, accepts new connections and tears down old connections. The CAC has to be applied to the buffer of each output port. If a new connection is accepted, bandwidth and/or buffer space in the network node has to be allocated for that connection[90][91][92][93].

Call admission control schemes may be classified according to achievable statistical multiplexing gain as: a) deterministic or peak CAC; b) statistical CAC; or c) best effort CAC. Below we examine these cases.

2.1.2.1 Deterministic Call Admission Control

Deterministic Call Admission Control (CBR- CAC) method is used for CBR traffic, which has a fixed rate. The only information that is required is the peak rate of the new connection. The new connection is accepted if the sum of the peak rates of all existing connections plus the peak rate of the new connection is less than the capacity of the output link. The disadvantage is that unless the connection transmits at its peak rates, the output port link may be grossly under-utilised [94].

2.1.2.2 Statistical Call Admission Control

In statistical call admission control (VBR CAC), the intended connection's bandwidth is not assigned/allocated on a per peak rate basis. Rather, the allocated bandwidth is less than the peak rate of the source (i.e. between the peak and the average). As a result, the sum of all peak rates may be greater than the capacity of the output link. Statistical CAC makes economic sense when dealing with bursty sources, but at the expense of congestion occurrence. Statistical multiplexing is a mechanism for reducing the bandwidth requirement of bursty and VBR traffic sources [88]. It has been used over the Internet to improve network utilisation, but without providing any performance guarantees [95]. It is being used in ATM networks in conjunction with QoS guarantees. In essence, statistical multiplexing is a spatial aggregation mechanism by which several individual streams are asynchronously superposed and transported over the same channel. The resulting aggregate traffic exhibits smoother bit rate behaviour than the original streams. Bandwidth is allocated to the aggregate traffic, resulting in reduction in the per-stream allocated bandwidth. This reduction is proportional to the burstiness of the multiplexed sources. One way, and the easiest way for statistical call admission, is treat the request as CBR request and perform the admission test based on PCR; however this scheme will lead to bandwidth under-utilisation [94].

Another form of admission control would be based on sustainable cell rate (SCR)[90]. This scheme is especially effective when there are large numbers of sources, since the more the number of connections multiplexed on the link; the less is the probability of sources transmitting at their peak rates. The aggregation of bursty sources generates traffic that is less burstiness. However, the delay bounds may not be satisfied all the time due to statistical fluctuations of the traffic.

More sophisticated admission control requires knowledge of the scheduling policy employed by each switch. The conservative assumption that there is a very small or no buffering CAC is called zero buffer approximation (or Rate Envelop Multiplexing-REM). The advantages of these assumption is that cell loss rate (CLR) derivations are easier because there is no need for queuing analysis and there is no need to consider the correlation in the arrival process. Mathematically, let S be a random variable representing the total amount of work arriving in a small time interval including existing and new connections. Let C be the available link capacity, CLR is given by:

$$CLR = E\{|S - C|\} / E\{S\} \tag{2.1}$$

S can be estimated either by traffic measurements or traffic models. Having the distribution of S, CLR will be computed. If the predicted CLR is lower than the required CLR for all existing connections as well as for the new connection the connection will be accepted.

If the switches use large buffers to absorb bursts, the zero buffer approximation CAC will not be efficient. In the rate sharing approach, the CLR predictor does not rely on modelling the traffic but on measurements and estimation of the distribution of the amount of cell arrivals during different time intervals. Let $S(t)$ be a random variable that represents the amount of work arriving in interval time t. Let $CLR(t)$ be a lower bound for the cell loss ratio based on the random variable $S(t)$ given by;

$$CLR(t) \geq E\{|S(t) - C(t) - b|\} / E\{S(t)\} \tag{2.2}$$

where $C(t)$ is the amount of work that can be served during time t and b is the buffer size. Because $CLR(t)$ for all t is a lower bound of CLR, $CLR \geq \max(CLR(t))$. If this inequality is considered as equality then an estimation of CLR is obtained. The key problem here is to have accurate estimation of the distributions of $S(t)$ for different t values. These can be obtained using traffic measurements by which moments or histograms are produced and the required distributions can be estimated [98].

In all the methods, the bandwidth allocation can be based on effective bandwidth definitions. Effective bandwidth is a number W_i associated with connection i such that the grade of service requirement is met if $\sum W_i \leq c$. Effective bandwidth of a given traffic stream is only related to statistical characteristics of the traffic stream and the network resources. It should not be dependent on the traffic characteristics of any other stream. Moreover, the effective bandwidth of the superposition of k streams is equal to the sum of the effective bandwidth of each of the streams. If we have an effective bandwidth value for each connection, the CAC function will be simply to compare the sum of these values in all network links on the route of the new connection with the total available capacity. If they exceed the total capacity on at least one of the links, the connection is rejected, otherwise it is admitted. For example, for zero buffer approximation CAC, the effective bandwidth is obtained using equation (2.1) for a given distribution of S to find the smallest value of C such that CLR is not higher than the required CLR. In the similar way, for rate sharing CAC, the derivation of effective bandwidth requires a formula for loss probability that takes into account the buffer as well as realistic models of the traffic. This is a difficult problem. A practical approach would be to use traffic traces generated by different services and to keep a record of CLR versus buffer size for large range of traffic streams.

2.1.2.3 Best Effort Call Admission Control

In the best effort service, if the call requires no grantees for minimum cell rate (MCR), then it will not be blocked due to a lack of bandwidth or buffer space (UBR service). If the call requires a MCR, then the call admission test is the same as for the CBR traffic but using the MCR only (ABR service).

2.2 Classification of Call Admission Schemes

A variety of different call admission schemes exist, where some require an explicit traffic model and others only require traffic parameters, such as the peak and average bit rate. In this section we review some of these schemes. For presentation purpose, the schemes have been classified as follows:

1. Effective bandwidth
2. Heavy traffic approximation
3. Fast bandwidth reservation
4. Time windows
5. Fast Resource Reservation
6. Dynamic call admission control
7. Multipath call admission control

These classifications were based on the underlying principle that was used to develop each scheme. Below, we discuss the some of the features of each class of CAC schemes and review some of the proposed schemes within each class.

2.2.1 Effective Bandwidth

Let us consider a single source feeding a finite capacity queue. Then, the effective bandwidth of the source is the service rate of the queue that corresponds to a cell loss rate CLR. The effective bandwidth for a single source can be derived as follows [7]. Each source is assumed to be an Interrupted Fluid Process (IFP). Let R be its peak rate, r the fraction of time the source is active, and b the mean duration of the active period. Then, an IFP source can be completely characterized by vector

(R, r, b). Assume that the source feeds a finite capacity queue with constant service time. Let K be the capacity of the queue. The effective bandwidth e is given by:

$$e = \frac{a - K + \sqrt{(a-K)2 + 4Kar}}{2a} R \tag{2.3}$$

where $a = \ln(1/CLR)b(1-r)R$

In the case of N sources and given that the buffer has a capacity K, the effective bandwidth is again the service rate e which ensures that the cell loss for all the sources is CLR. Ahmadi, el al. proposed the following approximation for multiple sources [7] :

$$c = \min\left\{\rho + a'\sigma\sum_{i=1}^{N} e_i\right\} \tag{2.4}$$

where

e_i is the equivalent capacity of the ith source calculated using expression (2.3), and $\sum_{i=1}^{N} e_i$ is the sum of all the individual equivalent capacities,

ρ is the total average bit rate, i.e. $r = \sum_{i=1}^{N} \rho_i$, where ρ_i is the mean bit rate of the ith source,

$\sigma = \sum_{i=1}^{N} \sigma_i$, where σ_i^2 is the variance of the bit rate of the ith source,

$a' = \sqrt{-2\ln(CLR) - \ln 2\pi}$.

Elwalid and *Mitra* [38] showed that the effective bandwidth of Markov modulated fluid source is approximately the maximum real eigenvalue of a matrix derived from source parameters, multiplexer resources, and the cell loss probability. Some studies have clearly indicated the inaccuracy of effective bandwidth methods in some situations [39]. In particular, the effective bandwidth method fails when a bufferless system subject to the same input traffic has a small probability that the traffic load exceeds the link capacity. In the effective bandwidth approach, this probability is assumed to be close to one (and is taken as one in the calculations). *Rege* [40] compares various approaches for effective bandwidth and proposes some modifications to enhance the accuracy of the scheme. *Elwalid, et al.* [41] proposed a method in which they combined Chernoff bounds and effective bandwidth approximation to overcome the shortcomings of the effective bandwidth. This method provides better accuracy than effective bandwidth for the case mentioned above of a bufferless mutiplexer that can achieve substantial statistical gain. However, in some other cases, the method does not solve all the problems with the inaccuracy of effective bandwidth.

2.2.2 Heavy Traffic Approximation

Sohraby [42] proposed an approximation for bandwidth allocation based on the asymptotic behaviour of the tail of the queue-length distribution. It is known as steady-state queue-length distribution and exhibits a geometrically distributed tail. For sufficiently large i, Sohraby suggests;

$$\Pr(queue\_length > i) \approx \alpha(1/z^*)^i \tag{2.5}$$

where

$$z^* \approx 1 + \frac{1-r}{\sum_{i=1}^{N} r_i R_i (1-r_i)^2 b_i} \tag{2.6}$$

where $r = \sum_{i=1}^{N} r_i R_i$. The approximation is good when the traffic intensity γ is $0.8 < \gamma < 1$. The cell loss probability is approximated by; $\gamma(1/z^*)^K$, where K is the buffer capacity, and z^* is given by Equation (2.6). The call admission decision is then quite simple. Accept a new connection if the resulting $\gamma(1/z^*)^K$ is small, or when,

$$\ln[\gamma(1/z^*)^K] < \ln(CLR) \tag{2.7}$$

This is a good approximation when there are a large number of sources with each source peak rate very small compared to link capacity. Only in this case, the system may be operated (efficiently) at the heavy traffic region.

2.2.3 Upper Bounds of the Cell loss Probability

Several other call admission control schemes have been proposed which are based on upper bound for the cell loss probability. Saito [43] proposed an upper bound scheme based on the Average Number of cells that Arrive during a fixed interval (ANA), and the Maximum Number of cells that Arrive in the same fixed interval (MNA). The fixed interval was taken to be equal to $D/2$ where D is the maximum admissible delay in a buffer. Using these parameters, the following upper bound was derived.

Let us consider a link serving N connections, and let $P_i(j)$, $i = 1,2,...,N$, and $j = 0,1,...$ be the probability CLR can be bounded by

$$CLR \leq B(P_1,...,P_N; D/2) = \frac{\sum_{k=0}^{\infty}[k - D/2]^+ P_1 * \ldots * P_N(k)}{\sum_{k=0}^{\infty} kP_1 * \ldots * P_N(k)} \tag{2.8}$$

where $*$ is the convolution operation. Let $\theta_i(j)$ be the following functions:

$$\theta_i(j) = \begin{cases} ANA_i / MNA_i, & j = MNA_i, \\ 1 - ANA_i / MNA_i, & j = 0, \\ 0, & otherwise. \end{cases} \tag{2.9}$$

Then it can be shown that

$$
\begin{aligned}
CLR &\le B(P_1,\ldots,P_N;D/2) \\
&\le B(\theta_1,\ldots,\theta_N;D/2) \\
&= \frac{\sum_{k=0}^{\infty}[k-D/2]^{+}\theta_1 * \ldots * \theta_N(k)}{\sum_{k=0}^{\infty}k\theta_1 * \ldots * \theta_N(k)}
\end{aligned}
\qquad (2.10)
$$

A new call is admitted if the resulting $B(\theta_1,\ldots.,\theta_{N+1};D/2)$ is less than the admissible cell loss ratio. Saito proposes a scheme for calculating $\theta_1 * \theta_2 * \ldots. * \theta_N$ effectively. He also obtained a different upper bound based on the average and the variance of the number of cells that arrive during $D/2$.

The main problem with this method is the absence of burst size in the calculation and thus a worst-case behaviour is assumed for the source. This method works well in the case when the actual source behaviour is close to the worst-case behaviour assumed in the above calculation.

2.2.4 Fast Resource Reservation

The fast resource reservation allocation method (fast resource management) was devised for transmission of bursty traffic. When a virtual connection is established, the path through the network is set up and the routing tables are appropriately updated, but no resources are allocated to the virtual connections. When, the source is ready to send a burst, only at that moment will the network allocate the required resources for the duration of the burst, i.e. on-the fly. The Fast Reservation Protocol (FRP) unit is located at the user-network interface (UNI) points. Further discussion on fast resources reservation is reported in Chapter 7, section 7.1.

An example on fast reservation of resources is Dynamic flow control (Dynaflow) [43][44]. Dynaflow is a class of fast resources reservation, where the resources are reserved on-the-fly. It requires no prior end-to-end session establishment. For example, with the FRP in [43], the bandwidth of each hop/link all the way to the destination needs to be reserved in advance before the source starts sending its burst. As the number of hops increase, this protocol is unlikely to be successful on all the links on a given source-destination pair. On the other hand, Dynaflow eliminates reservation delays, through the use of shared "burst-stores" until the downstream link bandwidth becomes available. The Dynaflow provides effective support for bursty traffic and at the same time allows users to send data without advance end-to-end virtual circuit establishment. However, the virtual circuits are established on a *burst-by-burst* basis by sending a burst setup cell at the beginning of each burst. This cell contains sender and destination addresses, as well as the sender's desired transmission rate. Setup cells must be sent periodically during the burst transmission in order to maintain the virtual circuit alive. At the end of a burst transmission, the sender forwards a release cell to free resources that are no longer required. Resources also released if no set up cells are received during a timeout interval.

The idea of routing and admission control operations of the Dynaflow method is basically an IP lookup. The Dynaflow switch keeps a list of outgoing links that are on paths to the desired destination. The first link in this list with sufficient unused bandwidth to handle the new flow is selected, and the flow is switched through to that link. If none of the outgoing links can currently accept the flow, it is diverted to a central shared buffer called a "burst store" for non-real-time applications; otherwise the burst will be discarded. Figure 2.2 shows an ATM switch that implements the Dynaflow service [46]. The Fast Reservation Protocol used in Dynaflow switch can be illustrated on Figure 2.3. When a set up cell is received on a previously unused VCI, the input port processor (IPP) forwards the setup cell to one of a number of Dynaflow control processors (DCP). It then buffers the arriving data on that virtual circuit in per VC buffer until the control processor makes its switching decision. When a Dynaflow control processor receives a setup cell, it performs a routing operation to determine which links can be used to reach the destination, and selects a link from this set that has sufficient unused bandwidth to accommodate the new flow.

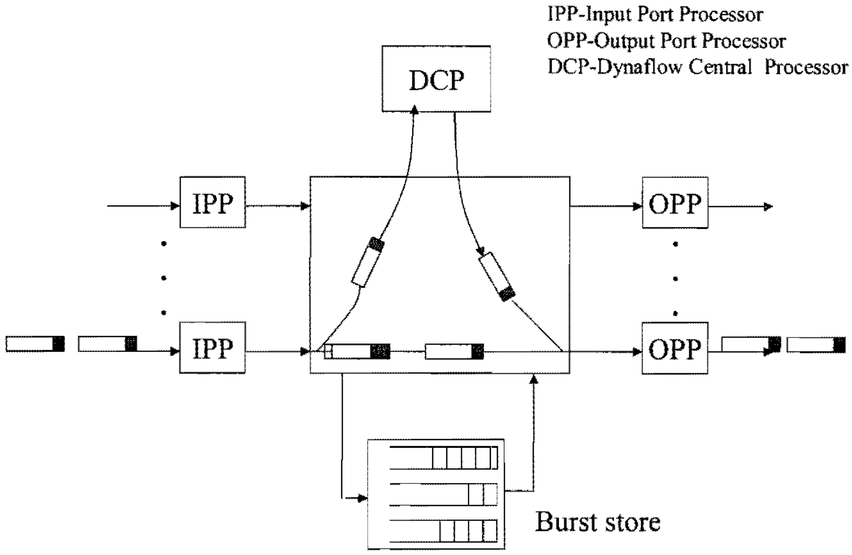


Figure 2.2: Dynaflow switch

It then forwards the setup cell on the selected link using a previously idle virtual circuit and sends a control cell to the IPP, modifying its virtual circuit table entry so that it forwards cells on the proper selected output virtual circuit. In [44], we proposed to modify the Dynaflow protocol presented in [46] to provide three alternative options when there is no available bandwidth in all of the outgoing links of the Dynaflow switch to the destination. The alternative options include storing the burst at burst store, re-routing the burst on alternative route, and discarding the burst. We proposed to use control cells to inform the control processor about the option that must be taken in the Dynaflow switch. If the buffering option is called, the burst is forwarded to one of a set of burst stores. The diversion of flow into the burst store is implemented through control cells sent by the control processor through the switching network to the IPP and to the selected burst stores itself. The bandwidth is allocated on the Dynaflow approach by DCP which keeps track of the rate of all flows using a given output link and allowing the admitting of new flows as long as the aggregate sum of their rates do not exceed that of the outgoing link bandwidth.

The Dynaflow protocol in [46] [47] was not intended to handle real time bursty traffic. However, in [44] we proposed to use a Prior Estimation of the Residual Bandwidth algorithm (PERB) [48] in order to reduce the waiting time of the incoming bursts and to judge immediately the availability/non-availability of the resources in the outgoing links. Also, a burst-scheduling algorithm has been provided to bind the delay and delay jitter.

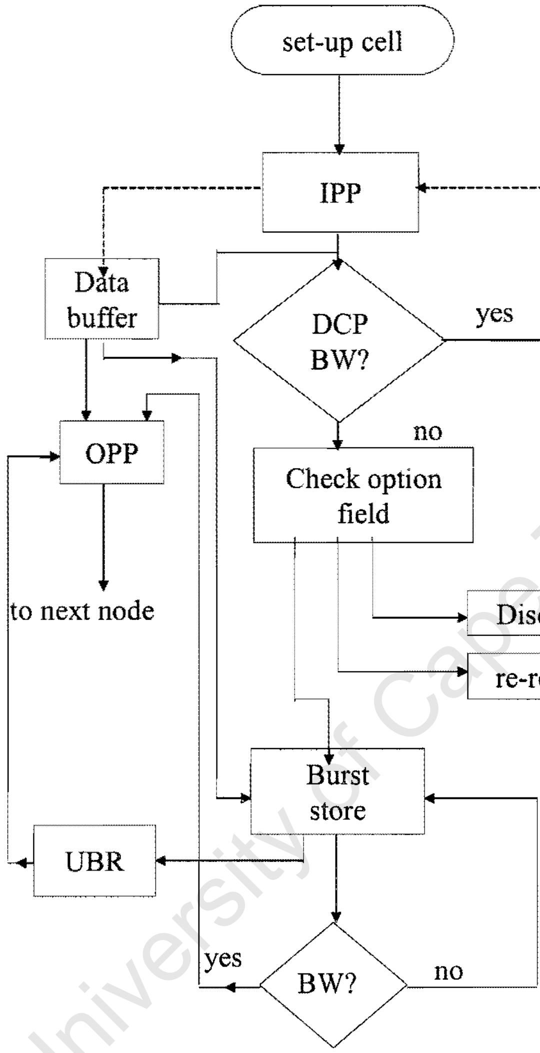


Figure 2.3: Flowchart of Dynaflow Protocol [44]

2.2.5 Time Windows

These schemes have been proposed based on the notion that a source is only allowed to transmit up to a maximum number of bits within a fixed period of time. This fixed period of time is known by different names, such as frame and time window. This notion is similar to the jumping window that has been used by the policing schemes.

Golestani [49] proposed a mechanism whereby for each connection, the number of cells transmitted on any link in the network is bounded. Thus, a smooth traffic flow is maintained throughout the network. This is achieved using the notion of frame, which is equal to a fixed period of time. The frame is not adjustable and it is the same for all links. Each connection can only transmit on a link up to a fixed number of cells per frame. Thus, the total number of cells transmitted by all connections on the same link is upper bounded. On a given switch, time on each incoming and outgoing link is organized into frames. Arriving frames over an incoming link are not synchronized with departing frames over an outgoing switch. A mechanism is proposed so that for each connection, the number of cells per frame transmitted on an outgoing link cannot exceed its upper bound. This mechanism is non work-conserving. However, a cell arriving at an input port in a given frame is guaranteed that it would be transmitted out of the switch at the end of an adjacent frame. This scheme requires buffering. Time windows were also proposed by Faber and Landweber [50].

Vakil and *Singh* [51] proposed a node-to-node flow control mechanism. For each connection, the transmitting node can only transmit up to a certain number of cells every fixed time period. The receiving node specifies the number of cells it can transmit. This is done using credits. The receiver informs the transmitter how many credits it can use for each connection per fixed period of time. If the credits for a particular connection are exhausted before the time period ends, then no more cells from this connection can be transmitted for the remaining of time period. The receiver can dynamically modify the number of credits. This method requires buffering.

2.2.6 Dynamic Call Admission Control

Dynamic call admission control was investigated by *Tedijanto et al.*[52], *Saito* and *Shiomoto* [53], *Bolla et al.* [54] and *Jamin et al.* [55]. In this case, the allocated bandwidth to the connection is dynamically adjusted every fixed period. However, in this research we have proposed two approaches for dynamic allocation of bandwidth in ATM networks (see Chapters 5 & 7) [30][56][57].

The first approach is proposed in Chapter 5; the bandwidth is allocated dynamically based on renegotiation of traffic parameters. The protocol is capable of rapidly adapting to the changes in network conditions, providing high quality video service, and good network resource utilisation.

The second approach is proposed in Chapter 7; the dynamic allocation of bandwidth is based on multipath routing at burst level. In this approach, at the time of burst arrival and there is insufficient bandwidth on the path that was established during call setup phase, a fast routing protocol is applied on *burst-by-burst* basis in order to re-route and accommodate the incoming burst.

Related to the dynamic bandwidth allocation issue, there are various reactive congestion control schemes that have been proposed in the literature [59][61][62]. Contrary to an initial negative reaction towards these reactive schemes, it has been shown that they can be effective in cases where the source has an ON period, which is long compared to the round trip propagation delay. These schemes, though they were developed specifically for burst-level congestion control, lend themselves to an approach for connection control at call level [58].

2.2.7 Multichannel Call Admission Control

Dejean, et al. [59] and *Lorenze et al.* [61] proposed a multipath call admission scheme based on multichannel allocation, which they referred to as the string mode protocol. The principle idea behind this scheme is that each burst is chopped into sub-bursts and each sub-burst is sent over a different virtual circuit. In view of multichannel allocation, this protocol can easily handle bursty sources with high peak bit rates compared to the capacity of a link.

The use of multichannel allocation is inevitable especially for real time applications. For example, multiple ATM virtual circuits that operate in parallel are necessary to interconnect high-speed hosts. It is possible that a few large flows generated by applications, such as remote visualization, high-resolution medical imaging and three-dimensional volumetric animations, drive the available network links to saturation. However, one of multiple paths/links allocation policies is to distribute the cells from a single burst onto a number of ATM virtual paths (VPs). This permits the network resources to be reallocated on a more frequent basis, resulting in an increase in statistical multiplexing gain. We earlier proposed in [62] a call level routing scheme that assigns cells of a session to pre-determined set of multiple paths (multi-channel allocation) according to a deterministic rule. The bursts arrival process of the traffic source was captured by a hyperexponential distribution. The smaller allocation granularity permits a finer control to be exerted and results an improvement in the performance compared to a routing strategy in which all cells of a connection follow the same path (uni-channel allocation) as shown in Figure 2.4. This is the central idea behind the Multi-channel Transmission Group (MCTG) routing architecture that was proposed for ATM networks in [63].

The disadvantage of multichannel allocation at call level is the additional reassemble burden imposed at the destination when cells comprising a given message arrive out of sequence.

2.2.8 Other Call Admission Control Schemes

Bovopous [64] and *Evans* [65] proposed a call admission scheme that can be formulated as an optimisation problem, where a particular reward function is optimised. Also, neural networks have been used for call admission control, as presented on [66][67][68][69][70]. A different approach for call admission control has been proposed by *Gibbens, et al.* [71]. They propose using Bayesian decision theory to provide a simple and robust call admission scheme in the existence of uncertainties in the source average rate. A source is characterised by its peak rate and cell delay variation tolerance. Simple load threshold rules are used for admission control. In this model, buffers are used for cell-scale congestion while burst level congestion control is accounted by a bufferless model. Call admission control schemes for virtual paths (VP) have been examined by *Sato et al.* [72][73].

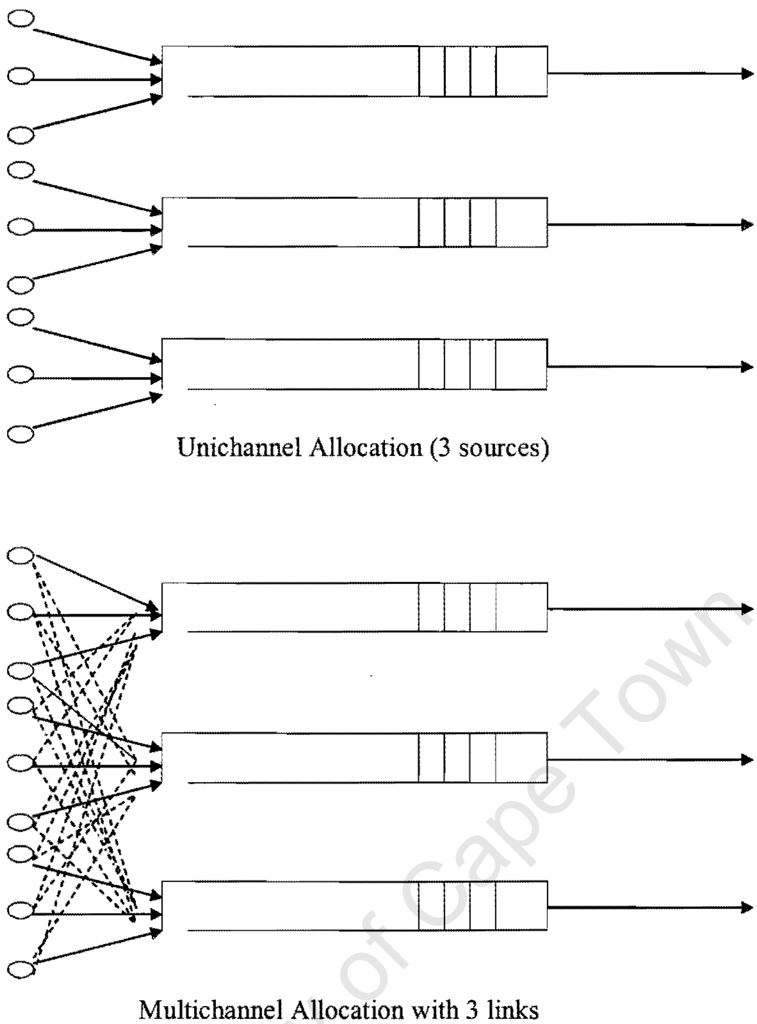


Figure 2.4: Multichannel call admission control

2.3 Tradeoff Between Network Efficiency and QoS

One of major ATM challenges is delivering network efficiency/utilisation while maintaining QoS. ATM promises seamless transport of a range of traffic types voice, video, and data. But QoS guarantees and network efficiency are competing objectives. Figure 2.5 illustrates the difficult tradeoff ATM makes between QoS and statistical gain. The level of traffic management determines the position of the performance curve. To seamlessly transport a range of traffic types and simultaneously maximize network efficiency, an ATM network requires advanced traffic management capabilities. Advanced traffic management leads to a higher performance curve than basic traffic management [96].

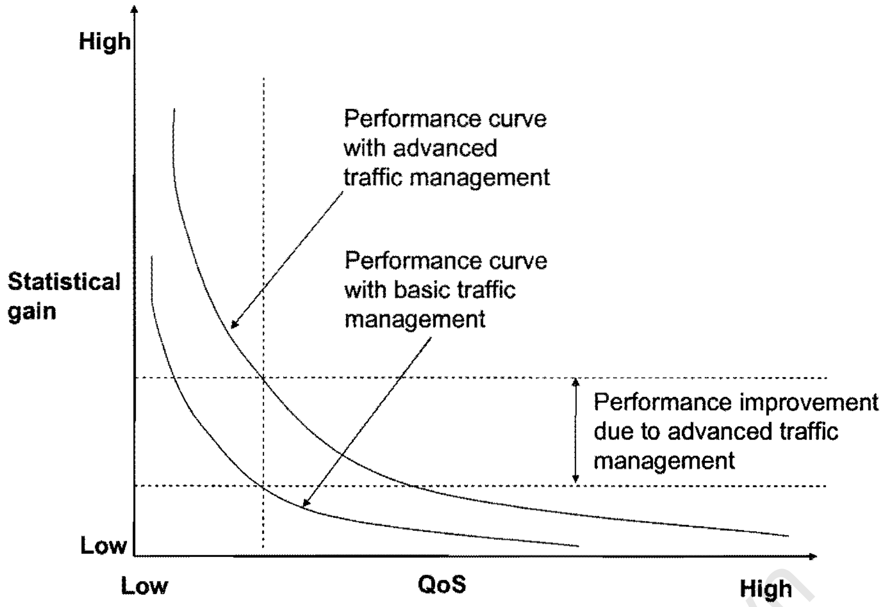


Figure 2.5: Tradeoff between statistical multiplexing and QoS

Because of ATM's asynchronous nature, contention in ATM networks is inevitable. Traffic streams contend for limited resources such as buffer space and bandwidth at every switch contention point. Serious contention, if not quickly and fairly resolved, can lead to network congestion. Specifically, excess traffic from misbehaving users, if not properly confined, can cause system overload and uneven resource sharing. Prolonged system overload can then escalate into congestion as higher-layer applications request re-transmission of lost cells. Congestion at a switch may ripple to the other switches in the network. The congested network cannot guarantee QoS levels, and the performance of higher-layer applications is seriously degraded. Advanced traffic management provides adequate buffering and differentiates among traffic streams. It must also efficiently address the fairness issue so that the behaviour of one user's traffic does not affect the QoS of other users.

2.4 Traffic Policing

The network does not trust the user to keep to the contract unsupervised, so a policing mechanism is implemented on the network side of the user-network interface (UNI), normally in the input port of the first ATM switch or cross-connect at the edge of the ATM network. If the network did not have this policing mechanism, individual users could accidentally or maliciously send excessive traffic causing congestion and consequent damage (cell loss and cell delay) to the traffic of other users in unforeseeable ways. Where permanent virtual circuits (PVCs) are concerned, the policing mechanism is supplied with the usage parameters control (UPC) via the network management system. For switched virtual circuits (SVCs), the contract is negotiated and parameters passed via UNI signalling messages and the call setup procedure to the policing mechanism. Note that the policing mechanism treats each virtual circuit on an ATM link individually and may police many thousands of virtual circuits at an interface. This sounds like an overwhelming job but it is worth remembering that, regardless of the number of virtual circuits, only one cell crosses the interface at any given time and that the more virtual circuits there are, the lower the average bandwidth of each. Most policing mechanisms use context-switching methods based upon the VPI/VCI of the current arriving cell to realize a "virtual" policer.

Usage Parameter Control (UPC) may monitor and control the PCR, SCR, CDVT and BT with the leaky bucket algorithm. The algorithm maintains a counter (bucket occupancy), which is decreased at a constant rate of one unit per time unit with a minimum value of zero. The counter is increased by an amount I for each arriving cell unit up to a maximum count value of $I+L$ bucket size, as can be shown on Figure 2.6. Any cell that arrives and causes the count to exceed $I+L$ is dropped or tagged. If the leaky bucket is used to monitor the PCR and CDVT then, $I = 1/PCR$ and $L = 1/CDVT$ (the new arrival may arrive at most L time units early without causing the bucket to exceed its capacity). The PCR of the CLP=0+1 cell flow is subject to UPC for all types of connections at the private and public UNI, whereas the enforcement of SCR and MBS is network specific.

Several experiments have shown that VBR sources can pass though a leaky bucket without being tagged [74]. To overcome the limitation of traditional policing mechanisms, multilevel policing mechanisms were defined. In each state of multistate policing mechanism, there is a single-state mechanism. For instance, each state of the multilevel leaky bucket mechanism has a leak rate, a counter, and a threshold. Whenever the state I threshold is reached, the mechanism jumps to the next state, which has higher leak rate. Whenever the counter goes back to zero, the mechanism jumps to the previous state [74].

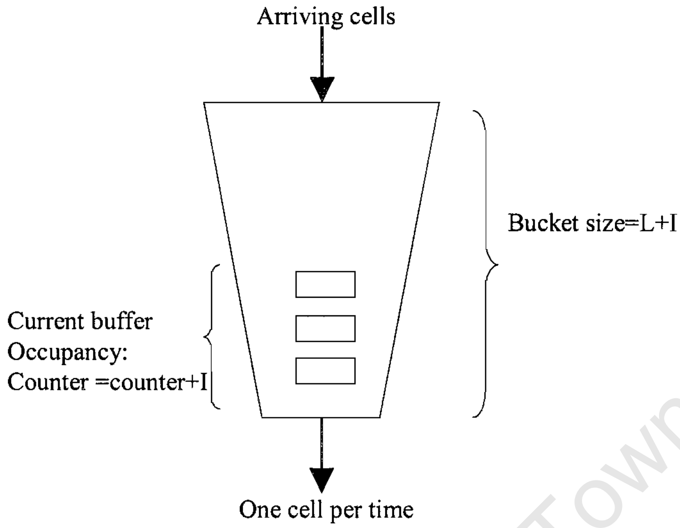


Figure 2.6: Leaky bucket UPC

2.5 Conformance to the Traffic Contract

The ATM Forum and the ITU-T have defined sets of options for policing different kinds of traffic and the different capabilities of different networks [31]. It is important to distinguish between the traffic characteristic offered to the network and what the network is prepared to tolerate; these may be essentially the same or they may be different. For example, if the user provides bursty traffic, which has characterised as ON-OFF model, the contract is easy to specify and police. Good conformance to the traffic contract depends on the offered traffic characterisation. Thus the difficulty rises from specifying accurate traffic parameters and will often result in inefficient use of the network resources, as more resources may have to be reserved to ensure a sufficiently good QoS.

2.6 Traffic Shaping

Traffic shaping is an alternative to traffic policing, where non-conforming cells are shaped and regulated instead of being dropped or tagged. Traffic shaping is about regulating the burstiness of the transmission. Flow specification is necessary for traffic shaping to describe the traffic pattern, thus QoS parameters desired by the application play an important role in the specification. The common way is to give the source flow specification to the subnet for approval. The specification can be accepted, rejected or changed by negotiations. Once a specification is accepted, transmission can start. An example flow specification includes characteristics of the input traffic such as maximum packet size, maximum transmission rate, token bucket size, token bucket rate and the service desired including delay, jitter, and loss. One of the best-know traffic shaping algorithms is the Generalized Cell Rate Algorithm (GCRA).

The common version of GCRA is based on the leaky bucket algorithm [33], [35]. The main idea of the algorithm is to allow a specified amount of data to be transmitted by each clock tick of the network (no matter what the input rate of the traffic is) and keep the rest of the traffic in a queue. If a cell arrives at the queue when the queue is full, the cell is discarded; otherwise it is appended to the queue. Thus a specific rate is maintained and the burstiness of the traffic is controlled. Leaky bucket can be implemented on special hardware or part of the operating system. One version of leaky bucket is the byte count leaky bucket algorithm. When variable-sized packets are used, it is better to allow a fixed number of bytes per tick rather than a fixed number of packets. Thus if the rule is 1024 bytes per tick, a single 1024 byte packet, two 512 byte packets, four 256 byte packets, and so on can be admitted on a tick.

Another version of leaky bucket is token bucket algorithm. The leaky bucket algorithm enforces a fixed output rate, no matter how bursty the traffic. Sometimes it is better to allow the output to speed up when large bursts arrive. In the token bucket algorithm, the leaky bucket holds token, generated by a clock at the rate of one token every ΔT seconds. For a packet to be transmitted, it must capture and destroy a token. A potential problem with token bucket algorithm is that it allows large bursts, although the maximum burst interval can be regulated. It is desirable to reduce the peak rate, but without going back to the low value of the original leaky bucket. One way to get smoother traffic is to put a leaky bucket after the token bucket. The rate of the leaky bucket should be higher than the token bucket's but lower than the maximum rate of the network. A basic difference between leaky bucket and token bucket techniques is that the token bucket scheme allows idle sources to save up permissions up to the maximum size of the bucket, in order to send large bursts later. Also the token bucket algorithm discards token when the bucket fills up but never discards packets, whereas the leaky bucket algorithm discards packets when the bucket fills up.

A source may shape its traffic before it enters the network to ensure that it meets its agreement and that all cells delivered to the network are confirming to the traffic descriptor. The network may also employ regulators or traffic shapers within the network to avoid clustering of cells that may occur after the traffic streams traverse several hops. Essentially, the traffic shaper tries to ensure that the traffic that flow into a scheduler is statistically similar at each scheduler within each switch from source to destination.

2.7 Congestion Control at the End System

When congestion control is implemented at the end system, the sender adjusts the traffic source rate based on the level of network congestion. It is different from rate control at the end system in the sense that congestion control at the end system is aimed at helping the network to recover from congestion, whereas rate control is an end-to-end process. Feedback from the network can be used by the sender as an indication of the level of network congestion. Alternatively, the buffer occupancy at the sender can also be used as a measure of the level of congestion in the network. The buffer occupancy can be used to control the transmission rate from the source [75][76]. Feedback from the server buffer has the advantage of a much smaller delay when compared to the delay in obtaining feedback from the network. A number of mechanisms are used to implement the congestion control at end system. Some of them are included in the following subsections.

2.7.1 Flow Control

Flow control is used to regulate the traffic rate between the sender and the receiver so that a fast/slow transmitter will not result in overflow/underflow at the receiver. Flow control mechanisms should prevent congestion in a network by sending signals back to the traffic sources in order to regulate their transmission rates. The problem with such closed loop controls in a B-ISDN environment is that the latency due to the finite speed of light makes them very sluggish for wide area networks [77]. In the time it takes to send a signal back to a source in the event of congestion, that source could have already transmitted thousands of cells. These cells will only contribute to the congestion which already exists. As a result, researchers have focused more on the open loop rate controls, such as the leaky bucket mechanism that has been mentioned in section 2.4.

For example, in multimedia transmission over ATM network, it is extremely important to maintain the traffic flow at an optimum rate. Too fast a rate will result in buffer overflow (leading to loss of multimedia traffic) at the sender. A very slow rate will result in buffer underflow at the receiver, leading to still frames and interruption in video display at the sender. When multimedia is transmitted over an available bit rate (ABR) service of ATM networks, depending on the level of network congestion, the amount of network bandwidth available to the multimedia source varies over time. In case of congestion, the bit rate of the source can be scaled down to help the network to recover from congestion. This approach does not suit the delay sensitive applications because of the delay introduced by the feedback congestion control.

Flow control can be implemented by the source sending information about its buffer level to the destination, as shown in Figure 2.7 [51]. Because the round-trip time (RTT) between the sender and destination is small and varies over a small range, it is relatively easy to match the source's transmit speed to the dynamic variation of the receiver buffer level. The large control latency resulting from a high value of RTT in a distributed multimedia system makes it difficult to implement the scheme based on direct feedback between the destination and source. Hop-by-hop flow control can be used to solve the above problem. In the hop-by-hop scheme, each node acts like an active local information generator. Let $\lambda_i(t)$ denote the outgoing rate for node i at time t, $X_{i+1}(t)$ the buffer occupancy at node $i+1$ at time t, and t the time interval for sampling of the buffer occupancy. The sending rate at node i is periodically computed based on the local source capacity and the buffer occupancy of the downstream switching node using $\lambda_i(t) = F(X_{i+1}(t-\tau), \lambda_i(t-\tau))$, this scheme can overcome the long response time between the source and receiver of a multimedia system. However, since each node needs to calculate the sending rate periodically, it requires a large amount of computing power at the nodes of a network with a large number of nodes and connections, eventually giving rise to scalability issues [51].

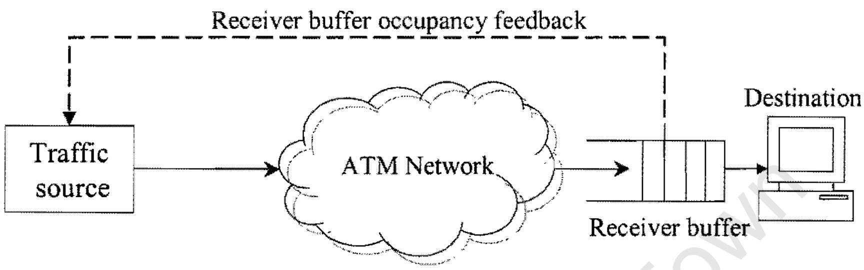


Figure 2.7: Flow control traffic model

2.7.2 Rate and Credit Based Control Schemes

Flow specification and traffic shaping alone do not help to control the rate of the traffic during congestion periods. Rate based closed loop schemes are considered as reactive approach of congestion control for ATM networks. They use a feedback mechanism to inform the source and/or destination so that an action can be taken for decongestion. Rate control and flow control are not the same. Rate control is concerned with the entire subnetwork's load, whereas flow control relates to the point-to-point traffic between a given source and a given destination. A source can get a "slow down" message either because the destination cannot handle it (flow control) or the network cannot handle it (rate control). Often the terms flow control and congestion control are used interchangeably.

In all feedback schemes, the hope is that the feedback will make the sources take appropriate actions for reducing the congestion. ABR traffic is the only type of traffic that can react to rate based control in current ATM networks. ABR congestion control is based on the idea that the source has an Actual Cell Rate (ACR) that falls between MCR and PCR. When congestion occurs, ACR is reduced; when congestion is removed, ACR is increased.

Rate-based control can be implemented either as single-bit-feedback (binary) or explicit-feedback. The binary feedback can only tell the source whether it should increase the rate or decrease it. This results in several round trips for the binary feedback to settle to the optimal operation. The explicit feedback can set the rate to the optimal operating point within few round trips by directly specifying the request rate.

ATM uses Resource Management (RM) cells as feedback packets. The RM cell contains a desired Explicit Rate (ER) field, a Current Cell Rate (CCR) field, a Congestion Indication bit (CI), as well as other fields. When RM cells are sent only to decrease the rate but nothing is sent to increase the rate, the method is called negative polarity feedback. The drawback of this method is that when RM cells get lost, the source will continue to increase its rate. Positive polarity feedback is when RM cells are sent only to increase the rate of the source. In bipolar feedback, RM cells are sent either to increase or to decrease the rate.

The basic idea behind the credit-based schemes is that the network nodes require credits in order to be able to send information. Nodes have one queue for each source maintaining a per-VC queue. The debate between rate-based flow control and credit-based control has now ended and the rate-based method has been chosen by the ATM Forum as a standard [32], however some switch vendors continue to use the credit based method. Credit-based schemes are different from rate-based implementations from the following perspectives:

- Per-VC Queuing: The credit-based method requires network nods to keep a separate queue for each VC. This applies even to inactive VCs. Per-VC queuing makes the network node complexity proportional to the number of VCs.
- Zero Cell Loss: The credit-based method can guarantee zero cell loss under ideal conditions. Even under extreme overloads, the queue lengths cannot grow beyond the credits granted. The rate-based method cannot guarantee zero cell loss.
- Ramp-up Time: The static credit-based method allows VCs to ramp up to the full rate very fast. Any free capacity can be used immediately. For rate-based schemes

and the adaptive credit-based technique, ramp up time can take several round trip delays.

- Isolation and Misbehaving Users: In per-VC method, misbehaving users cannot disrupt the operation of well-behaving users. In the adaptive scheme, a misbehaving user can get a higher share of buffers by increasing its rate. Note that isolation is attained by per-VC queuing not by credits. Thus, a rate-based network node can also achieve isolation by implementing per-VC queuing.
- Buffer Requirements: In the credit-based method, buffer requirement is proportional to link delay, while in the rate-based method; the total buffer requirement is proportional to the end-to-end delay.
- Delay Estimation: Setting the congestion control parameters in the credit-based method require knowledge of link trip delay. At least, the link length and speed must be known. This is not required for rate-based method.
- Network Node Design Flexibility: The credit-based method dictates each network node to use per-VC queuing with round-robin service. The explicit rate schemes provide more flexibility, so that some network nodes can optimise by minimizing their queue length, some can optimise their throughput, while others can optimise their profits.
- Network node vs. End-System Complexity: Credit-based approaches introduce complexity in the network nodes, but simplifies end-system's job. The source network interface cards are much simpler since they do not have to schedule each and every cell. As long as credits are available, the cells can be sent at the peak rate. On the other hand, all networks interface cards have to have schedulers for their CBR and VBR traffic. The same mechanism may be used for ABR without introducing much complexity.

One of the better-known credit based schemes is the Flow Controlled Virtual Circuit (FCVC) method, where each network node maintains a separate queue for each channel. The network node monitors queue lengths of each channel and determines the number of cells (credits) the source can transmit on that channel. The source can transmit only as many cells as allowed by the credit. The disadvantage is that if credits are lost, the source will not know it, and each channel needs to reserve the entire round trip worth of buffers even though many channels share the link. In

FCVC, the credit amount is computed by the network node (not by the source). Another method, called the pessimistic method, requires an explicit permit from the destination LAN gateway for each transmitted cell. There is no cells dropping, but the delay may be high.

2.7.3 Rate-Scalable Multimedia Transmission

It is important to have continuous transmission of multimedia traffic even during congestion in the network. Rate-scalable transmission refers to scaling the bit rate of multimedia traffic during periods of network congestion with a view to maintaining continuous transmission. However, this is achieved, in most cases, at a graceful degradation of video quality at the sender. There are several ways to achieve rate-scalable transmission of compressed video stream. These may include adaptive encoding, switching among multiple pre-encoded versions, and multilayer (hierarchical) encoding. Considering adaptive encoding, the encoder re-quantizes data on-the-fly based on network feedback [78][79][80][81]. However, since encoding is CPU-intensive, sources are unlikely to perform that for a large number of receivers. An alternative option is for the source to retain (keep) several versions of each video stream with different resolution. Thus depending on the network's available bandwidth the source switches between the lower and higher resolution quality streams.

Hierarchical or multilayer encoding, is adopted in this research and will be explained in more details in Chapter 4 & 5; it is a family of signal representation techniques, and generally refers to an approach in which the video compression source scales its output compression rate by partitioning the video stream into sub-streams or layers, each layer representing a portion of the signal. The greater the number of layers received at the end stations, the better is the quality of the reconstructed signal [197]. In this approach, as more bandwidth becomes available, more layers of the encoded stream are delivered. If the available bandwidth decreases, the source should be able to drop some of the layers. The multilayer encoding usually have the decoding constraint that a particular enhancement layer can only be decoded if all the lower quality layers have been received. There are several advantages of multilayer encoding, including that of less storage requirements at the source and provision of an opportunity for selective prioritisation of the important information. The design of an effective multi-layered video transmission system basically entails the design of an efficient drop and adds layered mechanism that can maximize the perceptual quality of the received video stream.

2.8 Buffer Management

Buffer management controls the way buffer space is allocated between different calls using the network and how the arriving cells are allowed to enter a buffer. Buffer management policies fall into two categories: cell blocking schemes [83] and cell discarding schemes [86]. Cell blocking schemes are allowed to block arriving cells in favour of others, which may arrive from a different traffic class. On the other hand, cell-discarding schemes are allowed to eject a cell from one class to make room for an arriving cell of a different class. Such schemes are sometimes called delayed resolution policies, because unlike cell blocking schemes, they can accept all cells with a "wait and see" attitude. If the space used by those cells is needed at later point in time, they can simply be discarded. Discarding schemes outperform blocking schemes because they never need to block a cell unnecessarily.

2.9 Buffer Scheduling

While buffer management schemes control how cells access a buffer, scheduling schemes control how cells are served and, therefore, removed from a buffer. Scheduling problems can become quite complicated. Of course, the simplest policy is FIFO, which serves cells in the order in which they arrive. Introducing priorities allows the network to account for the different performance requirements of different applications. Another simple scheme, called static priority, always serves cells from a higher priority class if there are any available in the buffer. Siram [87] presents an interesting scheme for scheduling packets from voice and data traffic based on a pair of parameters, (T_1, T_2), which control the maximum amount of time for which each of the traffic classes can be continuously served. Hyman et al. [88] introduces a more complicated scheduling algorithm, MARS, which guarantees the QoS requirements for the tree traffic classes introduced by Lazar et al.[89]. They show that the number of calls which can be carried through a multiplexer is substantially higher with MARS than with simple static priority. Mitrou and Pendarakis [90] investigated a joint buffer management/scheduling scheme in which buffer space and service bandwidth are partitioned between two traffic classes subject to their respective cell loss and delay constraints. Note that for all above-mentioned schemes, it is assumed that the network supports multiple traffic classes. ATM networks currently have very little support for different traffic classes, with only a single bit reserved in the cell header for priority.

2.10 Conclusions

In this chapter we have looked at several components of ATM traffic and congestion control issues. We have reviewed various mechanisms for traffic control such as call admission control, traffic policing, traffic shaping, rate and credit based schemes, and buffer management, etc. Each traffic control mechanism has distinct features and functionality many of them use similar techniques, such as feedback.

This chapter has considered also the traffic control issue from the inter-operation point of view, and captured the interdependencies among different traffic control aspects to ensure a proper network operation.

Chapter 3

Video Traffic Modelling and Analysis

The size of ATM– based multimedia networks, combined with the complexity and diversity of their projected traffic mix (voice, video, data, etc.), promise to render the task of network design and management a hard one. Any effort to develop control and management algorithms for multimedia networks has to be preceded by thorough study of the traffic that these networks will carry. This is particularly true of VBR video traffic, which has been shown to be significantly more difficult to characterize than traditional voice and data traffic. The term VBR refers to the fact that the bit rate of compressed video is not constant, but rather a random process. The VBR video traffic also imposes a stringent real-time constraint on the network, especially in the case of interactive services. Since these traffic streams have a complex structure, their effect on network performance may be much more complex than that predicted by simple, analytically tractable traffic models. Thus, a careful performance analysis effort is required for accurate modelling of multimedia networks, mainly via simulation experiments. Effective broadband network simulations must rely critically on high-fidelity traffic modelling.

This chapter was motivated by the need to evaluate network performance, design admission control, dynamically allocate bandwidth, and specify customer traffic. Traffic models are employed in two fundamental ways: as part of an analytical model, or to drive a network simulation program. Several issues associated with traffic modelling are discussed including the impact of video traffic parameters and the bursty nature of the VBR video, on the network performance.

3.1 Video Traffic Modelling in ATM

In order to address the multimedia-based ATM network performance issues properly, an accurate video traffic model is needed, in particular to evaluate queuing behaviour. Developing of models that are accurate, yet as simple as possible is necessary for simulating complete network settings.

Traditionally, telephony network traffic has been modelled by Poisson processes. The arrivals of telephony customers are essentially independent, and therefore, these kinds of models work well. Moreover, Poisson processes are analytically tractable, and many results about network performance, such as blocking and delay probability, can be obtained.

The bulk of previous work on the local and wide area network traffic shows that Internet and ATM traffic are bursty and strongly dependent [97][99][101]. These characteristics have profound impacts on network performance because they affect the queuing behaviours of networks [105]. In general, the dependence will degrade the network performance because it often results in buffer overflow [104]. Empirical studies and statistical analysis on network traffic gathered from high-speed networks provide evidence of the prevalence of self-similar patterns [6][97][103]. Obviously, it is inappropriate to use traditional traffic models to characterise ATM traffic because traditional models focus on a very limited range of time scale while the traffic exhibits correlations over a wide range of time scale [101].

The recent development of coding standards for digital video, such as H.261, H262, H. 263, MPEG-2, and MPEG-4, has made it feasible to transmit video data over communication networks. Video traffic will be dominating part of the network traffic, and, therefore, it is necessary to model video traffic. Due to the fact that the contents of a video stream are strongly dependent in nature, video traffic itself possesses burstiness and long-range dependence [6], and thus new methods to model network traffic, especially methods to model video traffic, are necessary.

There are two approaches to transport the video over communication networks as shown in Figure 3.1 [120]. One approach is to control the quantisation steps of the encoder to produce a CBR output. The drawback of this approach is that the quality of video is variable, especially in cases with many scene changes and high complex motion. An alternative approach is VBR transmission. VBR does not attempt to control the encoder's output, but instead produces a variable bit rate so that the video quality can be constant. This approach makes the bandwidth allocation very difficult. To increase network utilisation or statistical multiplexing gain, good models for VBR video are needed. This research considers the video that is transmitted over a VBR service.

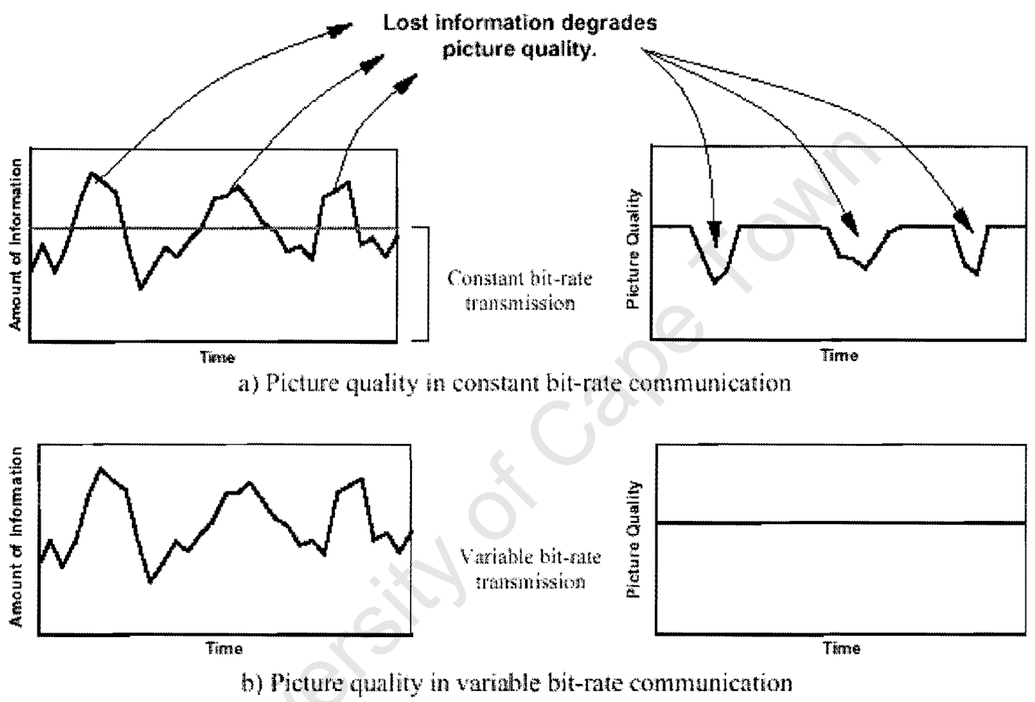


Figure 3.3-1: Video quality in CBR and VBR transmission

To model VBR video traffic accurately, autocorrelations among data should be taken into consideration. A considerable amount of work on video modelling has been done, which can be classified into statistical and deterministic models. The statistical models include:

- Renewal process models
- Markov models
- Transform expand sample (TES) model

- Markov modulated process model
- Histogram-based model
- Auto-regressive models
- Fluid model
- $M/G/\infty$ input process model
- Long range dependent model or self-similar model

Long range models can be further categorized into two classes:

- Short range dependent models (SRD),
- Long range dependent models (LRD).

These models are used to capture two statistical factors: marginal distribution (first-order statistics) and autocorrelation function (second-order statistics) of traffic data. LRD models can capture long range dependence, while SRD models can capture short range dependence. The impact of traffic dependence on queuing performance measures, such as queue length, waiting time, and cell loss rate, can be very dramatic. It is common belief that traffic dependence will degrade queuing performance.

To describe network traffic accurately, the models should not only capture the first-order statistics, but also the second-order statistics as well. Almost all modern traffic models take into consideration the traffic dependence to some extent. The difference between SRD and LRD models are the extent to which the dependence is considered.

Because of the statistical nature of video traffic, deterministic models are not popular as statistical models. A few deterministic models have been reported [102]. In the following sections we report some of the statistical traffic models.

3.2 Renewal Models

Historically, queuing systems have been analysed under renewal traffic models in which the inter-arrival times is identical independent distribution (i.i.d). A well-know example of renewal models is the Poisson process, in which the interarrival times are exponentially distributed. Renewal models include the phase-type renewal process [104][105], in which the interarrival times are derived from a continuous-time Markov process with discrete state space $\{0,1,...,M\}$. State 0 is absorbing, while all other states are transient. Starting with some probability distribution, an interval time is taken as the time to reach absorption. Subsequent interarrival times are obtained similarly by restarting the chain with the same initial distribution.

The advantage of using renewal models comes from their simplicity and analytical tractability. However, given the burstiness and the inherent correlations in ATM traffic, renewal models significantly underestimate the queuing performance, which is greatly affected by traffic correlations.

Traditional traffic can be described by a random point process. A random point process $\varphi = \{t_n : n \geq 1\}$ is a sequence of random points t_n at which an event occurred, where,

$$0 < t_1 < t_2 < ..... \tag{3.1}$$

with $t_n \to \infty$ as $n \to \infty$.

The point process has two equivalent processes: The counting process and the interarrival process. The counting process $\{N(t) : t \geq 0\}$ for φ is the number of points that fall in the interval $(0,t]$. Let $T_n = t_n - t_{n-1}$, then the process $\varphi = \{T_n : n > 0\}$ is the interarrival process. Furthermore, if φ is i.i.d, then this random point process is called a renewal process. Since renewal processes are analytically tractable, they have been traditionally used as traffic models. Poisson process is a special case of the renewal process. Due to the fact that Poisson processes have some elegant properties, they have been used widely in telephone industry. A Poisson process is an independent incremental process. Its interarrival process is exponentially distributed, thus making it memoryless, and greatly simplifying the analysis of a queuing system.

It is believed that traffic burstiness can be explained to a large extent by two factors: the shape of marginal distribution and autocorrelation [104]. Strong positive autocorrelations are strong sources of burstiness. The autocorrelation of renewal processes, however, vanishes for all non-zero lags, and therefore they fail to model modern network traffic.

The phase-type renewal process is an important special renewal process [104]. Its associated arrival process can be modelled as the time spent for continuous Markov process $\{C(t): 0 < t < \infty\}$, whose state space is $\{0,1,....,m\}$ and has an absorbing state, to go to absorption. To get a sample of the process A_n, start the Markov process C with the initial distribution π, the elapsed time is the value of the sample A_n. All the samples are obtained from the same initial distribution π. This kind of model is analytically tractable.

3.3 Markov-Modulated Poisson Process (MMPP)

The Markov modulated process can be identified by the modulated processes used. The most well known one is Markov Modulated Poisson Process. The MMPP is a doubly stochastic process where the arrival rate of a Poisson process is defined by the state of a Markov chain. A two state MMPP is shown in Figure 3.2.

The MMPP is a Poisson process whose arrival rate is a random variable that is modulated by the state of a continuous-time Markov chain. It is a correlated process that has a tractable queuing analysis and can be used to capture the randomness at different scales, and therefore, is versatile to capture traffic characteristics. A generator matrix for the Markov chain and an associated arrival rate matrix for the Markov chain can characterize an MMPP. Let $M(t)$ be a continuous Markov process with state space $\{0,1,2,...,M\}$. If the probability law of a process is determined by the state of the current state of $M(t)$ completely, then the process is called a Markov Modulated Process.

The MMPP introduces some correlations between successive inter-arrival times, and therefore can capture the bursty nature of video traffic up to some extent. With different number of states of the modulating Markov chain, the MMPP can be used to model different traffic sources [106][107]. The simplest case, however, is one that uses MMPP to model ON/OFF traffic sources. In this case the arrival during the OFF period is zero, while the arrival process during the ON period is the arrival rate of traffic. This model is sometime denoted as Interrupted Poisson Process (IPP).

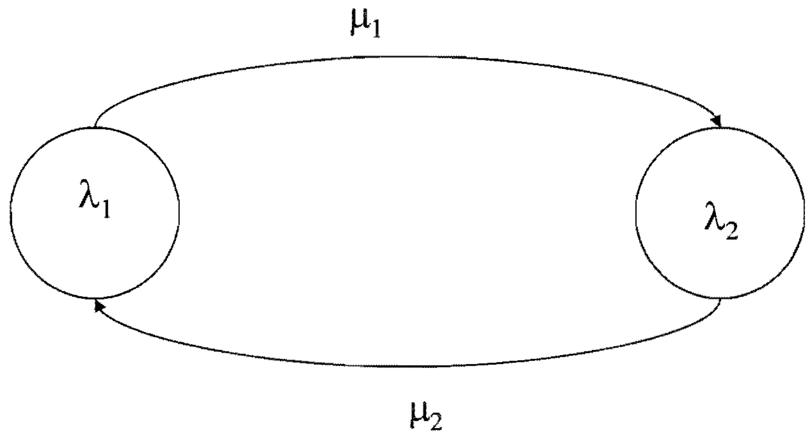


Figure 3.3-2: Two state MMPP

3.4 Fluid Models

The most common approaches employed are based on fluid and autoregressive models. In the fluid approach, a video stream is viewed as a stream of fluid that is characterized by a flow rate. The notion of discrete arrivals is lost as packets are assumed to be infinitesimally small (see Figure 3.3). The fluid approach has been found particularly appropriate to model the traffic in ATM networks for a number of reasons [108]. First, this model is able to capture the burstiness of ATM traffic. Second, the traffic granularity, caused by small-size cells that are transmitted at very high speeds, makes the impact of individual cells insignificant. This gives a justification for the separation of cell-level and burst-level time scales, which is the underlying theme in the fluid approximation. Third, the computational complexity of fluid analysis is independent of the buffer size, making the fluid modelling particularly useful for systems with large buffers. Fourth, the fluid models have a trustable queuing analysis, where the queuing results are often obtained numerically.

Fluid models were originally developed for data and voice sources [105]. The compressed voice and bursty data streams exhibit an ON/OFF characteristic when transmitted over a constant bit rate services. Therefore, the ON and OFF periods of the fluid flow are modulated by some stochastic process. During ON periods, the fluid arrives at peak bit rate. However, ON and OFF periods are i.i.d, often exponentially distributed duration. The attractiveness of the exponential distribution is that it gives rise to a superposition rule that, in fact, applies to all Markov-modulated models. An important property of the exponential distribution is that the minimum of several independent and exponentially distributed random variables is another exponentially distributed random variable.

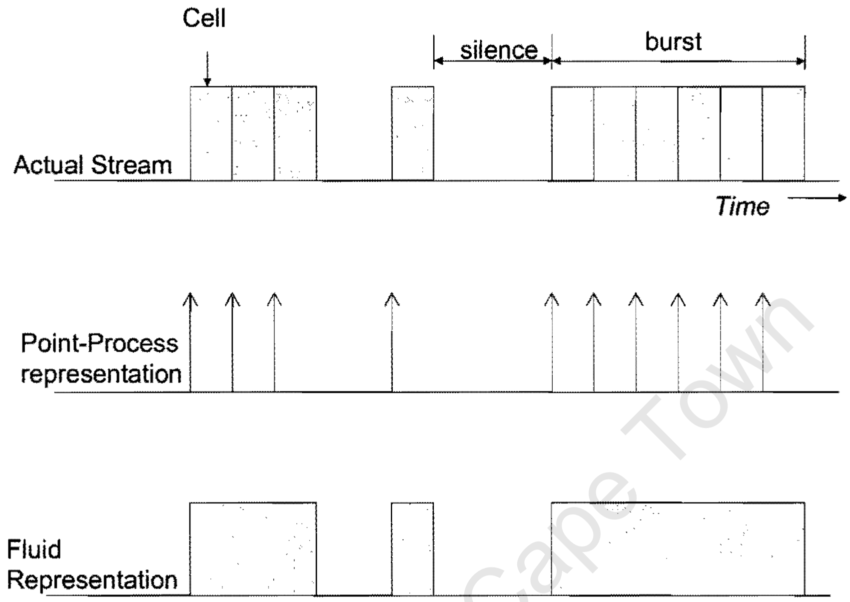


Figure 3.3: Fluid representation of an ON/OFF traffic source

3.5 Markov Modulated Autoregressive Models

In the autoregressive (AR) models, the video bit rate is modelled as a weighted sum of a finite number of previous bit rates. *Marglaris et al.* [109] modelled a video source as a first-order autoregressive process AR(1) with marginal probability density function Gaussian and exponential autocorrelation function. The main advantage of this model is its simplicity. A more sophisticated Autoregressive Moving Average (ARMA) process followed by memoryless non-linear filter has also been used [110]. This method matches the mean, variance, and autocorrelation function but not the marginal probability density function of the empirical video sequence. *Frater et al.* [110] employs the DAR (Discrete Autoregressive) scheme to model full-motion movies, including "*Star Wars*". The DAR model was also applied in *Heyman et al.* [9] to a variety of video conference records and was shown to work well.

Suppose Y_n is the number of bits of the nth frame. The state of the Markov chain at time n is $X_n(X_n \in \{0,1,...N-1\})$. The number of bits at time $n+1$ is given by:

$$Y_{n+1} = \begin{cases} a(i)Y_n + G(\mu(i),\sigma(i)^2), & \text{if } X_{n+1} = X_n = i \\ G(\eta(i),v(i)), & \text{if } X_{n+1} \neq X_n; X_{n+1} = i \end{cases} \tag{3.2}$$

where $a(i)$ is the autocorrelation at lag 1 and state i. G is a Gaussian random variable, where μ and σ are mean and variance of the Gaussian process, respectively. These parameters can be obtained as follows:

$$a(i) = 1 - \frac{D^2(i)}{2v(i)} \tag{3.3}$$

$$\mu(i) = \frac{\eta(i)D^2(i)}{2v(i)} \tag{3.4}$$

$$\sigma(i)^2 = D^2(i)\left(1 - \frac{D^2(i)}{4v(i)}\right) \tag{3.5}$$

$$D^2(i) = E\left[(Y_{n+1} - Y_n)^2 \middle| X_{n+1} = X_n = i\right] \tag{3.6}$$

where, $D, v,$ and η are estimated from empirical data, and using the above formulas.

From the above formulas we can see that the traffic rates at the same level are modelled by an AR process, while the number of bits for the first frame after a level change is the sample of a Gaussian process.

One potential drawback of autoregressive models is that while they aim to fit the empirical autocorrelation function, they lack a systematic way of fitting the empirical marginal distribution. However, the shape of the marginal bit rate distribution as well as that of the empirical autocorrelation function can cause traffic burstiness. Consequently, it is important to capture faithfully both first-order and second-order statistics of the empirical data.

3.6 Transform Expanded Sample Models

Transform expanded sample (TES) techniques belong to a class of non-liner regression processes. The main characteristic of the TES model is that it can accurately capture both first-order and second-order statistics of an empirical bit rate record; more specifically, the model approximates well the marginal distribution and the autocorrelation of an empirical data simultaneously [111][112]. For a given set of parameters, the autocorrelation function of a TES model is given in closed form. Therefore, by systematically searching in the space of parameters and numerically computing the resulting autocorrelation function of the model, one is able to approximately match the autocorrelation function of a given VBR sequence. A TES random process is generated form the so-called background random process by two transformations. Let $\{U_n : n = 0,1,....\}$ be the background random process with F_B being the uniform distribution, then the model is generated by the following formula:

$$X_n = F^{-1}(F_B(U_n)) \tag{3.7}$$

where F is the desired distribution function. F^{-1} is the inverse of F. U_n covering the range $[0,1]$ is produced recursively, which can be represented by a walk around a circle with unit length. The main problem of the TES modelling is to find the adequate distribution for the innovations and parameter for the transformation. It is necessary to make a good choice, because the distribution and the parameter determine the autocorrelation function of the generated sequence. The desired distribution is often expressed in the form of a histogram. If the distribution of U_n is uniform, the formula to generate the model data becomes:

$$X_n = F^{-1}(U_n) \tag{3.8}$$

There are two different TES models: TES$^+$ and TES$^-$, differing by the background process adopted. The background process used in generating TES$^+$ models is given by

$$U_n^+ = \begin{cases} U_0, & n = 0 \\ < U_{n-1}^+ + V_n >, & n > 0 \end{cases} \tag{3.9}$$

where $< x >$ is the fractional part of x. U_0 is distributed uniformly on $[0,1)$ and $V_n : n = 1,2,....,$ is a sequence of i.i.d random variables, with marginal distribution F_v and is called the innovation sequence. The background process used in generating TES$^-$ model is given by:

$$U_n^+ = \begin{cases} U_n^+, & n \text{ is even} \\ 1 - U_n^+, & n \text{ is odd} \end{cases} \tag{3.10}$$

These background processes are Markovian with uniform distribution. V_n used here should be independent of U_0.

In general, $\{V_n\}$ is obtained from distribution F_v, which is typically restricted to step functions in order to simplify the parameter search. The simplest method to get $\{V_n\}$ is given by:

$$V_n L + (R - L)Z_n \qquad -0.5 \le L < 0.5 \tag{3.11}$$

where Z_n are i.i.d and uniformly distributed on $[0,1)$. L and R can be replaced by α and ϕ as follows:

$$\alpha = R - L \tag{3.12}$$

and

$$\phi = \frac{R + L}{R - L} \tag{3.13}$$

which are more convenient for use because α controls the magnitude of the autocorrelation function and ϕ controls the oscillations.

A "smoothing" operation or stitching transformation may be applied to U_n before the inverse transformation F^{-1} is applied, so that the traffic model seems more homogeneous. The stitching function has the following form:

$$S_\xi(U_n) = \begin{cases} U_n / \xi, & 0 \le U_n \le \xi \\ (1 - U_n)/(1 - \xi), & \xi \le U_n < 1 \end{cases} \tag{3.14}$$

and thus, the formula for TES model is given by:

$$X_n = F^{-1}(S_\xi(U_n)) \tag{3.15}$$

All the TES background processes are Markovian and uniformly distributed on $[0,1)$ regardless of the innovation methods used, and the inversion method can ensure that we can always transform the background process to the desired distribution, that is, the TES models always have the distribution and autocorrelation functions.

Through appropriate selection of ξ and F_v, some autocorrelation functions can be fitted very well.

3.7 Long-Range Dependent Models

A common aspect of all models presented so far is that the interarrival times are either uncorrelated or are correlated with exponentially decaying ACF (e.g. Markovian models). Such ACFs are summable, that is, $\sum_{k=0}^{\infty} \rho_k < \infty$. The power spectrums are bounded at low frequency. Recently, a number of studies supported by extensive statistic analysis indicate the presence of persistent correlation in various types of network traffic, including LAN, WAN, and VBR video traffic [6][97]. This kind of phenomenon is well described by long range dependence process (LRD) [6] [103][113].

A LRD process has a ACF that is not summable, i.e., $\sum_{k=0}^{\infty} \rho_k < \infty$. Its power spectrum at low frequency is unbounded and approaches infinity as the frequency approaches zero [114]. It has been argued that Markov-like models cannot adequately capture the correlations persistence in network traffic. Instead, new models that exhibit the LRD behaviour should be used to characterise network traffic and capture its correlations at multiple time scales.

3.7.1 Fractional ARIMA Model

Long-range dependence is display by the FARIMA process, which is an extension of the conventional Autoregressive Integrated Moving Average (ARIMA) process defined by a different operator. Let Δ be the differencing operator, then:

$$\Delta X_k = (X_k - X_{k-1}) \tag{3.17}$$

The operation can be iterated as follows:

$$\Delta^2 X_k = (X_k - X_{k-1})(X_{k-1} - x_{k-2}) \tag{3.18}$$

and

$$\Delta^n X_k = \sum_{i=0}^{n} \binom{n}{i} (-1)^i X_{k-i}, \quad n = 1,2,3,.... \tag{3.19}$$

where n is an integer. X_k is the so-called ARIMA process. If the difference equation above is generalized to the non-integer case, then a fractional ARIMA process is obtained. The generation can be done by Gamma function in the following way:

$$\Delta^d X_k = \sum_{i=0}^{\infty} \binom{d}{i} (-1)^i X_{k-i}, \quad -1/2 < d < 1/2 \tag{3.20}$$

where $\binom{d}{i}$ is generalised factorial function.

$$\binom{d}{i} (-1)^i = \frac{\Gamma(-d+i)}{\Gamma(-d)\Gamma(i+1)} \tag{3.21}$$

where $\Gamma(x) \stackrel{\Delta}{=} \int_0^\infty t^{x-1} e^{-t} dt$ is the Gamma function.

It can be verified that the ACF of FARIMA process has the following form:

$$\rho_k = \frac{\Gamma(d+1)}{\Gamma(d) k^{2d-1}} \tag{3.22}$$

Thus, for $0 < d < 0.5$, the model exhibits LRD.

3.7.2 Fractional Gaussian Noise Model (FGN)

FGN is exactly a second-order self-similar process, obtained from stationary increments of a Fractional Brownian Motion (FBM). Fraction Brownian Motion, $B_H(t)$, is a Gaussian process with Hurst parameter $H \in (0,1)$. For the discrete-time case, the ACF of the discrete FGN is given by:

$$\rho_k = \frac{1}{2}\left(|k+1|^{2H} - 2|k|^{2H} + |k-1|^{2H}\right) \tag{3.23}$$

If $0.5 < H < 1$, then, as $k \to \infty$, $\rho_k \to H(2H-1)k^{2H-2}$, and the FGN exhibits LRD. H is the only parameter required for this model.

3.8 Modelling a VBR MPEG Stream

The main objective of this section is to find a suitable and simple model to capture the statistical behaviour of a VBR MPEG sequence. The model is used to generate a synthetic workload representing VBR MPEG traffic. This section presents two Markovian-based models, namely the Histogram and the Detailed Marko Chain models (DMC). The Markov chain process has been used because its parameters can be found easily and it can be easily analysed. This could be helpful to find the most appropriate model. The Markov chain method can be used to model different layers of an MPEG sequence (scene, Group of Pictures (GOP), frame, or slide). It is very difficult to find a model that covers all three types [19]. Therefore, we have to decide on which layer is to be used. A higher layer will add more complexity to the model but it will also improve long-range dependencies behaviour. The GOP layer can be used to these models without modelling *frame-by-frame* correlation and the only correlation used is *GOP-by-GOP* (*frame-by-frame* correlation is used at the traffic generation process). In addition, an experimental result showed that *frame-by-frame* correlation has no influence on cell loss results [19]. Therefore, in some cases it could be enough to use only one level of correlation.

For the histogram model, 0-order Markov chain method has been used and n-order Markov chain for the DMC. Both models have a finite number of states and are used to generate a GOP size process. For both models, the range of GOP sizes of the empirical MPEG sequence are divided into several quantisation intervals $\{q_i : i = 0,1,2,3,...,M-1\}$ where M is the number of quantisation intervals and the number of the intervals depends on the formulated model. Each interval is related to a state of the Markov chain. Therefore, the number of states is equal to the number of GOP intervals. For each state, there is a mean value μ_{q_i} of the GOP interval associated with it. The mean value of interval i, μ_{q_i}, represent the size of the quantisation interval q_i. In the Markov chain context, a transition matrix controls the transition from a state to another. With each state transition (entrance from the current state into the next state), a GOP size is generated according to the mean value of the next state.

These models are used to approximate the statistical behaviour of the MPEG sequence. Both models are well known. But with some modification and scene changes consideration, it is possible to be tuned for some type of VBR MPEG sequences to improve the actual traffic approximation. Another improvement can be added to these models when they are used with the generation process to perform multiple levels of correlations (GOP-by-GOP and frame-by-frame). Also a scene change-based model is presented later in this chapter. The scene-based model will also be used to approximate the behaviour of the MPEG sequence.

For the development of video traffic models, prior-knowledge of the MPEG coding technique and the statistical analysis of the frame size sequence, which can be obtain from measurement, is required.

Figure 3.4 and Figure 3.5 depict representative portions of the original video traces[115]. It is observed that the VBR traces of both frame sizes, $\{F_n\}$, and slice sizes, $\{S_m\}$, consist of segments of variable length which exhibit high short-term autocorrelations, as suggested by the fact the successive frame and slice sizes tend to lie close to each other. The boundaries of these segments are characterised by burst large magnitude changes in the local average of frame or slice sizes. Moreover, as one would expect, the segment boundaries coincide in the time sequence of both frames and slices.

Intuitively, one would expect to relate visually discernible segments in the empirical video sequence to a physical phenomenon in the underlying movie. Specifically, we wish to establish that these segments do correspond to distinct scenes of the movie, i.e., portions of the movie without significant shifts of view; such shifts result from burst camera zooming, panning, or cutting to a new visual context. By observing the actual movie frames and the corresponding bit rate sequence in lock step, it was possible to ascertain that scene changes do indeed coincide with bursty frame or slice size.

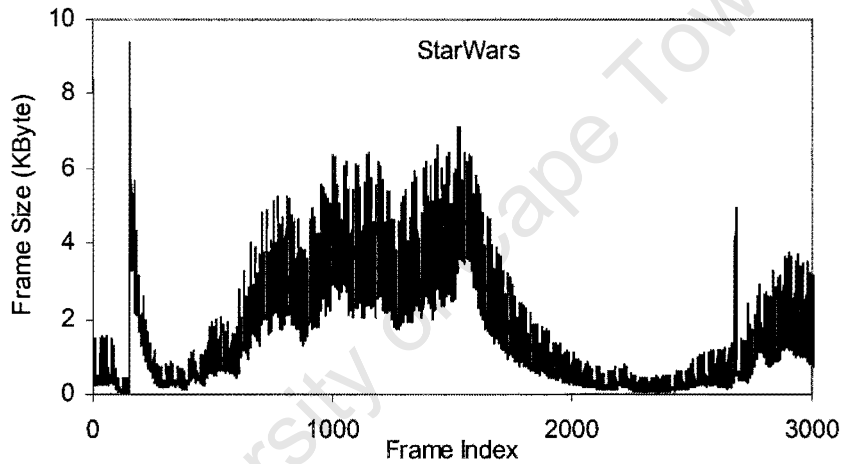


Figure 3.4: Frame size sequence (MPEG-4)

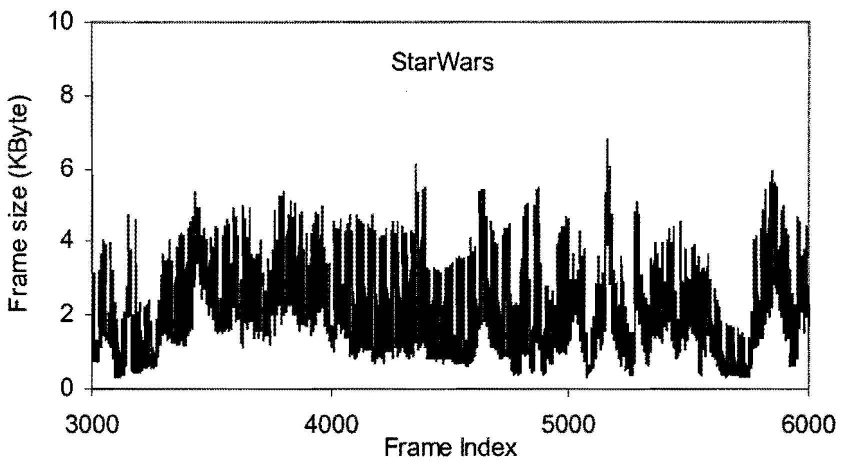


Figure 3.5: Frame size sequence (MPEG-4)

3.8.1 Histogram Model

The histogram-based model can be described by a simple Markov chain with a finite number of states (M) which is equal to the number of the quantisation intervals. The number of quantisation intervals is based on the selected number of the histogram bins. It is possible to improve the distribution feature of the model by increasing the number of quantisation intervals (smaller size of bin interval), but this will lead to an increase in the number of states. Each state i is associated with mean value μ_{q_i} of the *ith* interval (bin). Thus, with every transition from i to $i+1$ state, the μ_{q_i} value of next state is generated. We estimate and define the transition matrix of size $1 \times M$ as follows[119]:

$$P_{ij} = \frac{Number\ of\ GoPs\ within\ \text{int}erval\ i}{Total\ number\ of\ GoPs} = \frac{n_i}{N} \tag{3.24}$$

where, $N = \sum_{i=0}^{M-1} n_i$ and $n_i = number\ of\ GOPs\ in\ q_i$

The transition from state to state is statistically independent. The Histogram model can be used to determine the empirical GOP size distribution; however it doesn't model the GOP correlation because the GOP samples are generated according to the histogram bins, which are independent from each other.

3.8.2 Detailed Marko Chain Model (DMC)

The DMC model differs from the Histogram model in two main ways: the number of quantisation intervals (number of states) and the estimation of transition matrix. For long range dependency sequence, the 0-order Markov chain is not adequate. But, with some effort it is possible to obtain a Markov chain model with a high coefficient of autocorrelation, even for large lags. This could be done by increasing the number of states to a reasonable number. However, increasing the number of states increases the model complexity. Thus, the number of states M can be found as follows:

$$Size_q = \frac{Max_{GoP} - Min_{GoP}}{k} \tag{3.25}$$

where Max_{GOP} and Min_{GOP} are the minimum and maximum value of GOP size and k is a selected number of states. The transition matrix for the random process can be approximated by:

$$P_{ij} = \frac{Number\ of\ transation\ from\ i\ to\ j}{Number\ of\ transation\ out\ of\ j} \tag{3.26}$$

Therefore, $N_i = \sum_{i=0}^{M-1} n_{ij}$ and $\sum_{j=0}^{M-1} P_{ij} = 1 \qquad for \quad i = 0,1,2,...,M-1$

where, n_{ij} is the number of transition from state i to state j, N_i is the total number of transitions out from state i.

3.8.3 Scene Length Distribution

It is observed from Figure 3.2 and Figure 3.3 that I frames exhibit different VBR dynamics at different time scales. At a time scale of the order of few seconds, the bit rate fluctuates in small amounts about some mean level, which itself varies drastically at a larger time scale. The fluctuations in the mean levels at the larger time scale are often attributed to "scene" changes [116][117]. A scene, in the visual sense, is loosely defined as a portion of the movie without sudden changes in view, but with some panning and zooming [105]. Incorporating a "scenic" component in a traffic model gives the VBR dynamics a physical interpretation and often leads to better performance predictions. In [118] a heuristic approach to determine scene boundaries was proposed using only the sizes of I frames. The heuristic is based on the premise that "significant" changes in the sizes of consecutive I frames are strong indications of scene changes. We use this heuristic to obtain the sequence of scene lengths that are computed to an accuracy of a GOP period (i.e., 30 seconds). This has minor impact on the goodness of the fit since a scene lasts on average for several seconds.

3.9 Conclusions

In order to achieve the goal of this research to develop a dynamic bandwidth management procedure for bursty VBR traffic like video, traffic modelling has to be addressed first. Traffic modelling is essential to characterise the behaviour of the traffic that the network will carry. It be should be sufficiently simple yet accurate enough to capture the distribution and correlation function of the source bit stream. To achieve this goal, we have presented in this chapter models that can match the behaviour of VBR traffic including MPEG encoded video. It been explained in this chapter that video traffic has short range dependence (SRD) and long range dependence (LRD). Traffic dependence has a drastic effect on cell loss ratio and other network performance. SRD models include traditional traffic models, such as Markov processes and regression models. These models were discussed in section 3.2 and 3.3. However to capture LRD and SRD, a Markov-modulated self-similar process model has been presented to capture video traffic dependencies characteristic. Also the introduction of TES process allows for modelling of wider range of stochastic processes than previously possible.

Chapter 4

Transmission of Compressed Video over ATM Networks

ATM networks promise to provide the means to transport diverse traffic streams. These streams vary in their traffic characteristics and performance requirements. The largest portion of bandwidth in ATM networks is expected to be consumed by video applications. This is because of introduction of many new video/multimedia services. However, few video streams transmitted in their uncompressed digital format can saturate the huge link capacity provided by optical fibres. To increase the number of video streams that can be simultaneously carried over the link, compression schemes are employed in order to reduce the amount of bits contained in video frames. The size of a compressed video frame varies depending on the scene activity and the type of compression involved. Hence, the output of a video compressor is a VBR stream.

Several standards organizations have been working intensely on the enhanced compression techniques, which are suitable for broadband communication. The focus of this chapter is on a particular compression technique, which has gained considerable attention, namely the Motion Picture Experts Group (MPEG) [121]. MPEG is a standard jointly developed by the International Organization of Standards (ISO) and the International Electro-technical Commission (IEC). The ISO has branched into MPEG-1, MPEG-2, and MPEG-4, with cooperation from The ITU-T. The MPEG algorithm utilises the Discrete Cosine Transform (DCT) and motion entropy coding to obtain extremely high compression ratios.

Since this research is centred on bandwidth allocation and QoS provisioning under the influence of ATM traffic burstiness and video is generally sent over an ATM network in compressed format, digital video compression needs to be examined. Because the traffic source considered in this research is a video in MPEG format, a broad understanding on the MPEG compression standard is also required. Lastly, as will be shown in subsequent chapters, it is possible for video quality to be affected by different bandwidth allocation schemes. These schemes have a varying ability to improve network performance and support the QoS required by different video sources. Therefore, issues related to the transportation of digital video over an ATM network also need to be reviewed.

4.1 MPEG Video Compression

The MPEG standard is a widely used format for coding digital video and associated audio information. It makes use of the temporal and spatial redundancies found in video frames to achieve a high degree of compression (ranging from 30 to 1 to as high as 100 to 1). A number of MPEG standards are known as MPEG-1 (ISO11172), MPEG-2 (ISO13818), and MPEG-4 (ISO14496-2). MPEG-2 is an extension to MPEG-1 in terms of providing a wide range of resolutions, bit rates, and encoding options [122][146][147]. MPEG-4 natural video standard consists of a collection of tools that support wide video applications from digital television, streaming video to mobile multimedia. The MPEG-4 provides tools for shape coding, motion estimation and compensation, texture coding, error resilience, sprite coding, and scalability. The MPEG standards have three key parts:

- 'Systems' – addresses the synchronization of video and audio information, initial and continuous management of coded data buffers to prevent over or underflow, absolute time identification, etc;
- 'Video' – addresses video coding;
- 'Audio' – addresses the compression of digital audio information.

The hierarchical structure of MPEG bit stream is illustrated in Figure 4.1 [159]. MPEG video sequence starts with a sequence header that describes the basic parameters of the coded sequence such as the dimensions, resolution and frame rate of the sequence. At the Group of Picture (GOP) level, coded frames are grouped together with the header containing time and reconstruction information. Within each GOP, are a number of frames or pictures, each of which is composed of a series of slices. The slice header can be used by the decoder to re-synchronize with the coded bit stream when an error occurs. In this case, the decoding process skips to the beginning of the next slice and continues decoding. Each slice contains macroblocks that are made up of luminance components and the spatially corresponding chrominance components. In the macroblock layer, blocks of DCT coefficients are used to represent these luminance and chrominance components. Knowledge of the MPEG bit stream structure demonstrates that information fields in the MPEG bit stream carries a different level of importance. It also enhances the analysis of the network performance evaluation.

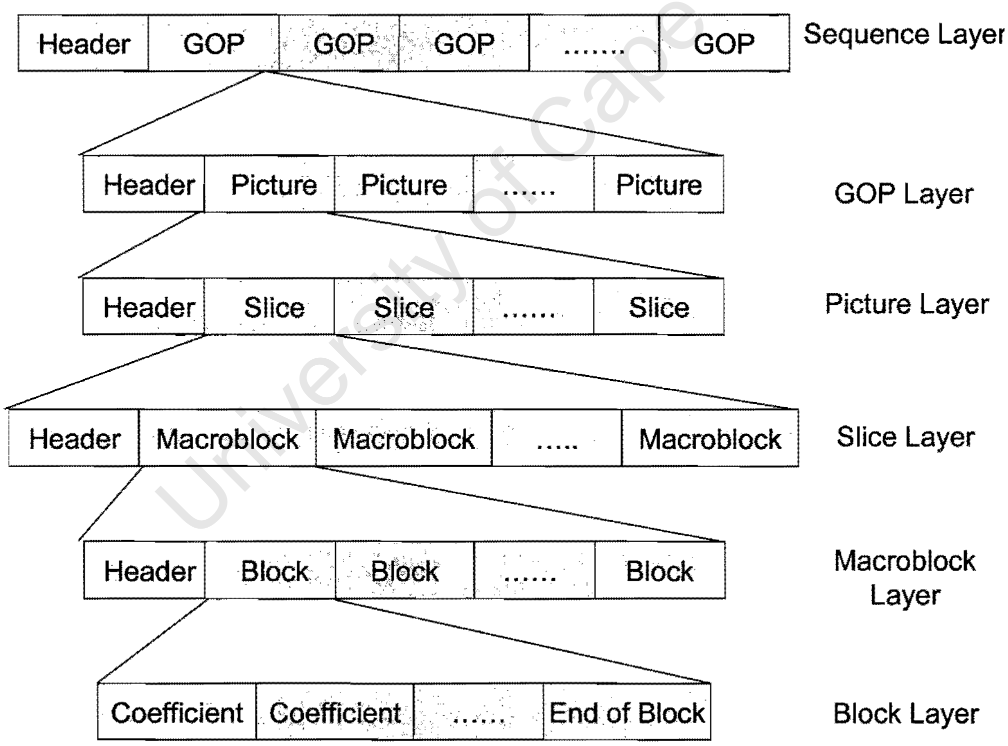


Figure 4.1: Structure of MPEG stream

Three types of picture frames are defined in MPEG standard [121][122][123]: I frame, P frame, and B frame. A mixture of these three types of frames are present in a GOP. The structure and size of each GOP is not specified in the standard and can be chosen to suit the application. Figure 4.2 shows an example of a GOP structure as well as the prediction dependencies among the different types of frames. Each type of frame uses different coding methods described as follows:

- An Intra-coded picture (or I-frame) is coded using information only from itself and only makes use of intraframe coding techniques;
- A Predictive-coded picture or (P-frame) is a picture which is coded using motion compensated prediction from a past reference frame (i.e. a previous I or P frame in the sequence);
- A Bi-directionally predictive-coded picture (or B-frame) is interframe coded using interpolated motion prediction between the previous I or P frame and the next I or P frame in the sequence.

Generally, B-Frames can achieve a higher degree of compression than P-Frames because of their bi-directional prediction nature, resulting in a smaller frame size. P-frames are in turn smaller than I-frames, as I-frames only makes use of intraframe compression. As a result, the generated bit rate is not constant and MPEG videos are generally bursty and considered to constitute variable bit rate traffic in ATM networks. Typically, MPEG-1 encoders support the coding of video and associated audio into a single data stream of bit rates around 1.5Mbps with 30 frames per second at a resolution of 352 x 240 pixels. The video quality is comparable to that produced in videocassette recorders. Amongst the MPEG encoding parameters, changing the interframe to intraframe ratio and the quantisation scale can lead to a significant change in the characteristics of the resulting video sequence [124]. MPEG-2 builds on the MPEG-1 video standard as a compatible extension in terms of providing a wide range of resolutions, bit rates, and encoding options.

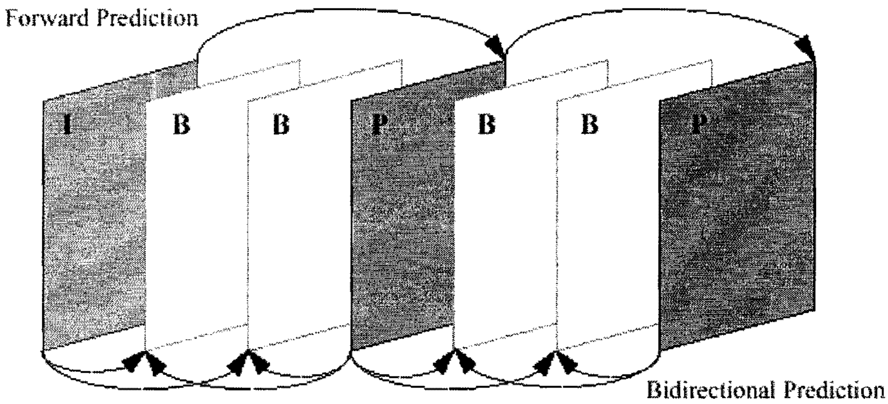


Figure 4.2: MPEG Frame Level Structure

4.2 MPEG-2 over ATM

MPEG-2 transmission has been researched by many groups [125][126] [127] [128] [129][130][131]. MPEG-2 is adopted by ITU-T for video transmission over AAL 5. Another group investigating MPEG-2 over ATM networks is the Digital Audio Visual Council (DAVC). MPEG-2 calls digitised video and audio data streams elementary bit streams. Elementary bit streams are formed into variable-length packet elementary streams (PES) as illustrated in Figure 4.3. There are two methods for constructing a single channel representing multiple applications: the program stream (PS) and transport stream (TS) methods.

In the PS method, various elementary bit streams are multiplexed by transmitting the bits for the complete packets in sequence resulting in a sequence of variable length packets in the channel. Program streams were designed for relatively error-free media such as CD-ROMs. MPEG-1 is a program stream-based system.

In the TS method, PES packets including the PES headers from various elementary bit streams are carried as a payload within fixed-length transport packets. Each transport packet is accompanied by a transport header that includes bit stream identification information. Each PES packet would then occupy a variable number of packets and data from various elementary bit streams are interleaved with each other at the transport packet layer. New PES packets always start a new transport packet layer, and stuffing bytes are used to fill packets that are only partially filled with

PES data. The MPEG-2 transport stream packet structure consists of 188 bytes comprised of 4 header bytes and 184 payload bytes.

The specific adaptation layer chosen for MPEG-2 is AAL-5. AAL-5 consists of a video-audio Service Specific Convergence Sublayer (SSCS) and the Common Part Convergence Sublayer (CPCS). The Convergence Sublayer (CS) could be design to support CBR as well as VBR MPEG-2 traffic, and more generally to support other non-MPEG video-audio services that do not have all the functionally of the MPEG-2 systems layer. Therefore one disadvantage of including a CS is that the additional functionality it provides would be redundant with functionality provided at the MPEG-2 systems sublayer. Considering that ATM cells may have up to 4 bytes in the 48 byte payload field, three techniques are suggested for mapping MPEG TS (188 bytes) into the ATM transport packet:

- The simplest method is the null AAL structure: The transport packet is partitioned into 48 byte payloads that can be included in the user data field.
- Use single byte of AAL in payload: The TS is partitioned into 47 byte payload resulting in an integer multiple of the ATM payload. TS of 188 bytes will fit into four ATM cells and the FEC code may be included in the 1 byte AAL.
- Dual AAL byte structure: TS is to be partitioned to 46 byte length.

The id of the transport header can be discarded since it can be reconstructed from the ATM header.

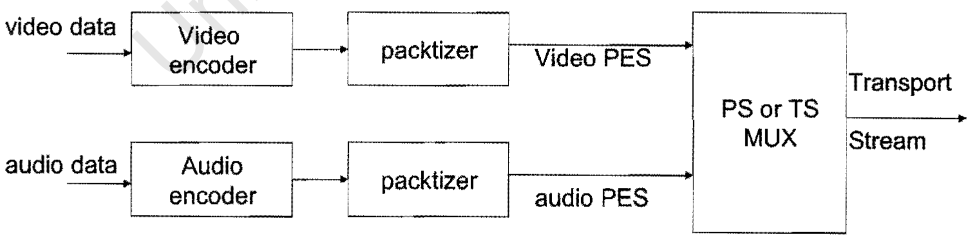


Figure 4.3: MPEG-2 system multiplexing

4.2.1 MPEG Transport over AAL1

According to the ITU-T Recommendation H.222.1 [132], the Program Stream and the Transport Stream may use the service provided by AAL type 1 at the AAL Service Access Point (SAP). Figure 4.4 shows the mapping of a 188-byte TS packet into exactly four ATM cells using AAL 1. AAL 1 is designed for real-time applications and therefore, it is suitable for supporting video streaming over ATM. Figure 4.5 shows the SAR-PDU format for AAL 1 [132]. The other advantageous features provided by the CS of AAL type 1 to transport video signals for interactive and distributive services include the following:

- Handling of possible rate mismatch between the sender and receiver end-systems – this is handled by the Synchronous Residual Time Stamp (SRTS) or the adaptive clock recovery method and prevents buffer to overflow or underflow during the delivery of video streams;
- Handling of lost and misinserted cells – the sequence count values are used to detect and locate lost and misinserted cells at the receiver. While detected misinserted cells are discarded, it may be necessary to insert appropriate dummy SAR-PDU payloads (maintaining the bit count integrity) or set the error indication bit in the TS packets (activating error concealment in the decoder) to compensate for lost cells. The Sequence Counter is also protected against bit errors;

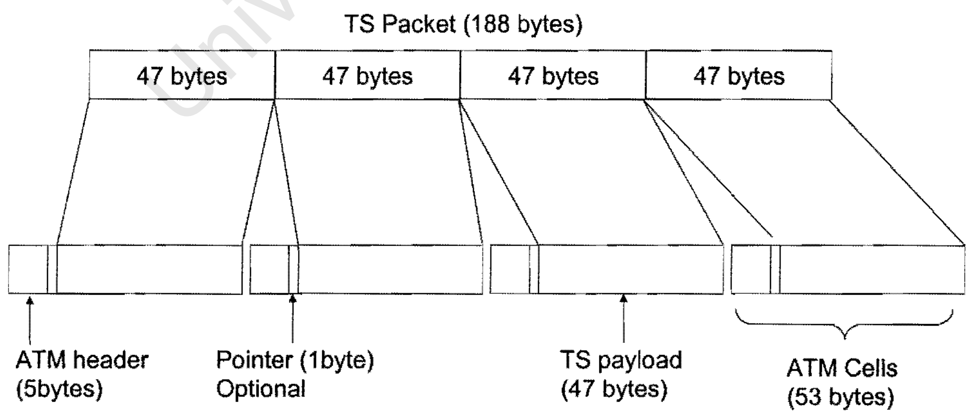


Figure 4.4: TS packet to AAL 1 cell mapping

- Handling of cell delay variation – a buffer is used to support this function. In the event of buffer underflow or overflow it may be necessary for the CS to maintain bit count integrity by inserting or dropping an appropriate number of bits;
- Correction of bit errors and lost cells – this is an optional function provided by the AAL 1 Convergence Sublayer. For correcting bit errors, the Forward Error Correction (FEC) technique using Reed-Solomon (128, 124) codes, which are able to correct up to 2 errored octets, is used. For the correction of bit errors and cell losses with delay restrictions, a method that combines FEC (Reed-Solomon (94, 88) codes) with octet interleaving of data (using a 16-cell interleaver) is used. This method can correct one cell loss occurrence in the group of 16 cells, or three errored octets in a row of 94 octets but has an overhead of around 6% [133].

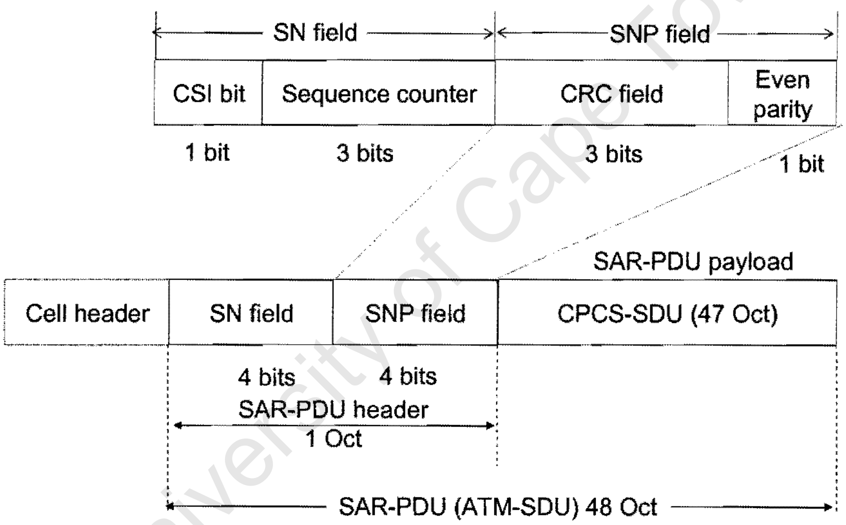


Figure 4.5: SAR-PDU structure for the AAL 1

On the other hand, the transportation of MPEG streams over AAL 1 has the following disadvantages:

- AAL 1 is designed to support CBR applications. However, MPEG bit streams are generally considered as VBR in nature. As a result, special encoding technique is required to generate CBR MPEG streams;
- In providing some of the functions to support video signals listed above, the use of the SN and SNP fields (1 byte out of 48 in the ATM cell payload) in AAL 1 results in higher transmission and processing overheads compared to AAL 5;

- Support for AAL 1 in end-user equipment is not very widespread at this point in time yet.

4.2.2 MPEG Transport over AAL 5

The Video-on-Demand specification approved by the ATM Forum [158] as well as the ITU-T Recommendation H.222.1 [132], defines the mapping of MPEG-2 Transport Stream packets into AAL 5 with a NULL Service Specific Convergence Sublayer (SSCS). One to N TS packets are mapped into an AAL 5 SDU and the value of N depends on the type of virtual circuit used. For Switched Virtual Circuits (SVCs), the value of N is established during the ATM user-to-network connection setup phase as described in ATM Signalling 4.0 of the ATM Forum [32] and ITU-T Recommendation Q.2931 [92]. For Permanent Virtual Circuits (PVCs), the default value of N is 2. Figure 4.6 shows the mapping of two TS packets into an AAL 5 CPCS-PDU which eventually breaks up into 8 ATM cells by the SAR sublayer. The functions offered by AAL 5 and its advantages to carry MPEG streams include the following:

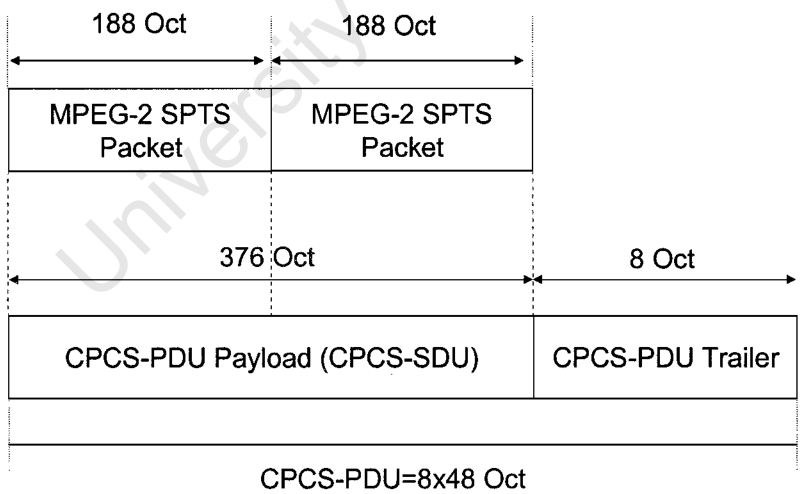


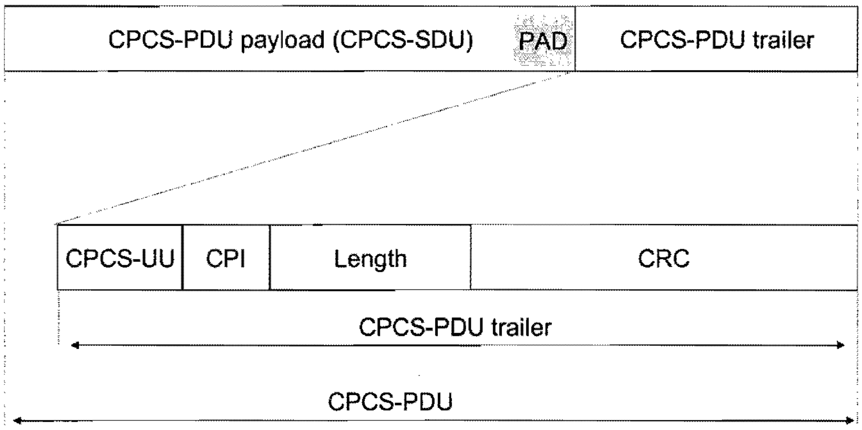
Figure 4.6: Format of AAL 5 PDU containing 2 TS packets

- AAL 5 is designed to be a simple and efficient. It carries relatively low transmission and processing overhead. There is only 8 octets transmission overhead per AAL 5 PDU and no per cell SAR protocol processing;

- Because a Service Specific Convergence Sublayer (SSCS) has been defined in AAL 5 to support ATM signalling protocols, it has become widely available to all ATM switches and end-stations that implement SVCs. For video application running on end-stations that possess signalling capabilities, AAL 5 is already implemented and ready to be utilised;
- The adoption of a NULL CS requires no additional network functionality to be defined;
- The size of the CPCS-SDU can be up to 65535 octets (64 kilobytes - 1), which allows for a high degree of flexibility in the encapsulation of MPEG encoded bit stream;
- End of SAR-SDU indication – the SAR sublayer utilises the ATM-User-to-ATM-User indication (AUU) parameter in the PT field of the ATM header (Figure 4.7) to indicate whether the payload of the current ATM cell contains the end of a SAR-SDU (AUU = 1) or not (AUU = 0);

Figure 4.7: SAR-PDU format for the AAL 5

- Detection of lost or misinserted cells – the 8 octet CPCS-PDU trailer contains a Length field and a Cyclic Redundancy Checker (CRC) field (Figure 4.8). The Length field is used to encode the length of the CPCS-PDU payload field and enables the receiver to detect cell loss or cell misinsertion. As in the case for AAL 1, error concealment in the decoder at the receiver can be activated by the detection of a loss cell;
- Detection of bit errors – the CRC field is used to detect bit errors in the CPCS-PDU. It contains the value of a CRC-32 calculation that is performed over the entire contents of the CPCS-PDU, including the CPCS-payload, the PAD field, and the first four octets of the CPCS-PDU trailer.



PAD	Padding	(0...47OCT)
CPCS-UU	CPCS User-to-User Indication	(1 OCT)
CPI Common	Part Indicator	(1 OCT)
Length	Length of CPCS-SDU	(2OCT)
CRC	Cyclic Redundancy Check	(4 OCT)

Figure 4.8: CPCS-PDU Format for the AAL 5

- Error handling – corrupted CPCS-SDU can either be discarded or optionally delivered to the SSCS. On the other hand, the transportation of video signals over AAL 5 has the following drawbacks:
- There is no mechanism built into AAL 5 for timing recovery; AAL 5 does not provide native support for FEC; Although the procedures for the delivery of corrupted CPCS-SDU are defined in the ITU-T Recommendations I.363.5, they have not been widely implemented in end-user equipment;
- While the Length Field of AAL 5 is only capable of detecting the presence of a cell count error, the Sequence Counter in AAL 1 can also locate the lost or misinserted cell.

The comparisons between the use of AAL 1 and AAL 5 to carry MPEG video streams over ATM networks can be highlighted by the fact that more error handling functions in AAL 1 results in higher transmission and processing overheads. On the other hand, a lower overhead in AAL 5 implies less error handling ability. The other critical difference between these two AAL types is that AAL 5 is already widely implemented in end-user equipment and ATM switches while AAL 1 is still catching up. This is why the Video-on-Demand application adopted AAL 5 services. Other possible schemes to pack MPEG information into AAL-SDU are examined in [134] and [135].

4.3 An Overview of MPEG-4

MPEG-4 is targeted at video conferencing applications with low bit rate requirements [136][137]. Content-based coding and scalability, along with improved efficiency through coding of concurrent streams, are the main properties of MPEG-4. Bit rates targeted for MPEG-4 standard are 5 to 64Kbps for mobile applications and up to 4 Mbps for TV/film applications [137].

The key innovation in MPEG-4 is the introduction of objects as the smallest accessible units compared to the traditional frame-based approach. These objects can be auditive or visual, static or dynamic, natural or synthetic. Thus, MPEG-4 can offer new interactivities for end users of multimedia streaming applications. For instance, besides VCR functionalities that are usually provided by frame-based video, object-based video can allow users to interact with video contents (video objects) dynamically, such as object moving, object zooming/out, object adding/deleting, and object quality enhancing/degrading. However, the interactivities may bring out the new network traffic control issues: how to adapt the bit-rate of each object in the same scene and how to assign different transmission priorities to different objects according to user's dynamic interactions. These issues will be discussed in the next chapter.

The ability to identify and selectively decode and reconstruct video content is referred to as content-based scalability. This feature provides the most elementary mechanism for interactivity and manipulation of contents of images or video in the compressed domain without the need for further segmentation or transcoding at the receiver. To enable content based interactive functionality, MPEG-4 video Verification Model (VM) introduces the concept of Video Object Planes (VOPs). It is assumed that each frame of an input video sequence is segmented into a number of arbitrarily shaped image regions, where each of the regions may possibly cover particular image or video content of interest, i.e. describing physical objects or content within scenes. The input to be coded can be a VOP image region of arbitrary shape, and the location of the region can vary from frame to frame. Successive VOPs belonging to the same physical object in a scene are referred to as Video Objects (VOs) a sequence of VOPs of possibly arbitrary shape and position. The shape, motion, and texture information of the VOPs belonging to the same VO is encoded and transmitted or coded into a separate Video Object Layer (VOL). In addition, relevant information needed to identify each of the VOLs and how the various VOLs are composed at the receiver to reconstruct the entire original sequence is also included in the bitstream. This allows the separate decoding of each VOP and the required flexible manipulation of the video sequence and scalable coding.

The definition of the VOPs may be done in multiple ways, such as by automatic or semi automatic segmentation, by hand segmentation, or they may always have been available if the scene was created as a composition of martial (VOPs) taken from different sources. The MPEG-4 content-based method can be seen as a logical extension of the conventional MPEG-1 and MPEG-2 coding method toward image input sequences of arbitrary shape.

The MPEG-4 standard will provide the tools to achieve error resilient object-based streams, either in terms of bit errors or cell loss in relevant environments, such as mobile networks with severe error conditions, ATM networks, or storage media. The error protection will make sure that some objects receive different protection than others, and that some parts of an object bitstream receive different protection than other parts (e.g. header or shape information receives better protection). Error resilience should consider concealment, fault tolerance, graceful degradation and graceful recovery, also in an object-based way. It will be possible to switch off error protection if there is no need for it. MPEG-4 system and video will provide capability for data prioritisation, error detection (corrupt data, insertion, deletion), and error concealment.

MPEG-4 video will provide the ability to withstand random errors and produce usable video (as defined within the context of profiles) with a Bit Error Ratio (BER) up to 10^{-4} [162]. MPEG-4 systems will provide, for errors that do not cause loss of the channel, a recovery time within one round trip delay. The additional processing time is acceptable and does not cause a problem. MPEG-4 systems will provide the possibility to do reliable downloading to ensure the integrity of identified data, e.g. scene composition data.

In terms of the ISO seven-layer communications model, no specific transport mechanism is defined in MPEG-4. Existing transport formats and their multiplex formats suffice, including the MPEG-2 transport stream, ATM and RTP on the Internet. A separate transport channel could be setup for each data stream, but there can be many of these for a single MPEG-4 scene and as a result the process could be unwieldy and waste bits. To remedy matters, a small tool in MPEG-4, FlexMux was design to act as an intermediate step to any suitable form of transport. In addition, another interface defined in MPEG-4 lets the application ask for connections with a quality of service in terms of parameters like bandwidth, error rate, or delay.

4.4 Multilayer Video Encoding

The basic idea of the multilayer (Hierarchical) encoding technique is, to split the video signal into components of varying importance [138][139]. The aggregation of these components reconstructs the original data, but subsets of the data can also provide various degrees of approximation to the original signal as illustrated in Figure 4.9 and Figure 4.10. Signal subsets are coded separately, and therefore, they are decoded separately. By careful design, the first components in the hierarchy can be a good approximation to the overall signal, providing a good first impression of the information without requiring all the components to be received.

In addition, future digital video broadcasting, video storage, video on demand, and video conferencing have different bandwidth requirements. To fulfil the different requirements using one common bitstream in a wide range of video services in heterogeneous environments, video coding techniques are needed that can simultaneously support a variety of bit rates tailored to individual services. One approach is to represent video information in a multilayer compression format, which enables a receiver to select a part of the generated bitstream and decodes it with the given available resources.

Multilayer encoding with a variable spatial resolution [140], has received less attention than non-layered encoding (sequential) with a fixed resolution, despite its adoption in some popular coding standards [141][142][143].

Many different proposals have been made for multilayer encoding of video (and continuous media in general). *Karlsson, et al.* [138] present a general approach to video transmission over packet-switching networks, emphasizing the role of multilayer encoding. One of the most basic and conceptually simple methods for hierarchical or layered coding is bit-plane separation [140]. This method applies to images with representations that use multiple bits per pixel. It encodes first separately subsets of the image that contain the Most Significant Bits (MSBs) of each pixel, which have the most important image information. Then, it progresses down through the bit-layers, towards the least significant bits, which provide the high-resolution sub-signal. Thus each bit-plane (or each bit-plane subset) can be separately encoded, progressively transmitted, and decoded independently from the others. Interestingly, with bit-plane separation, the most (visually) important components (i.e. those produced by the MSBs), are also the more highly compressible ones.

More elaborate multilayer encoding is based on sub-band coding, where spectral decomposition is carried out on the luminance (intensity) portion of the video signal. This splits the frequency spectrum of the luminance temporally, vertically, and horizontally, to give 11 separate sub-bands. Then, the 13 sub-bands (i.e., the 11 luminance bands plus 2 chrominance components) are coded individually. In [144] the authors have considered a sub-band coding scheme with 11 bands, of which, however, only 4 were determined to be important and were included in the measurements. The study revealed that source correlation could generate severe problems for naive statistical multiplexing schemes, emphasizing the importance of hierarchical coding.

Two-layer coding was considered in [139]. It was presented with low frequency; subsignal usually referred to as the base layer, and the second, higher resolution subsignal termed the enhancement layer. It was reported there that the video was of acceptable quality even with 10% packet loss at the enhancement layer.

Some of advantages of multilayer encoding are:

- Progressive presentation.
- Reduced bandwidth transmission: High quality images can be transmitted faster over low-bandwidth channels, at lower resolution.
- Resolution refinement: This is the ability to produce still images at higher resolution than when viewed in motion [145].

If the various frames in a video traffic are compressed on multilayer or hierarchical format, then only the acceptable resolution components can be decoded at the receiver, while the higher resolution components can be cached to be used only for still frame display. This reduces the decoding time for the common case.

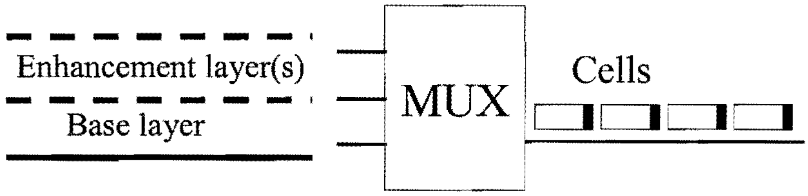


Figure 4.9: Multiplexing of layered data onto a single stream

4.4.1 Scalability Models

Common video compression techniques like MPEG-2 and MPEG-4 already allow scalability and, thereby, the adaptation of the quality of the video to the available bandwidth [160]. The idea behind these scalable schemes is to encode video signals not only into one stream but, instead, into a number of output streams as shown in Figure 4.10.

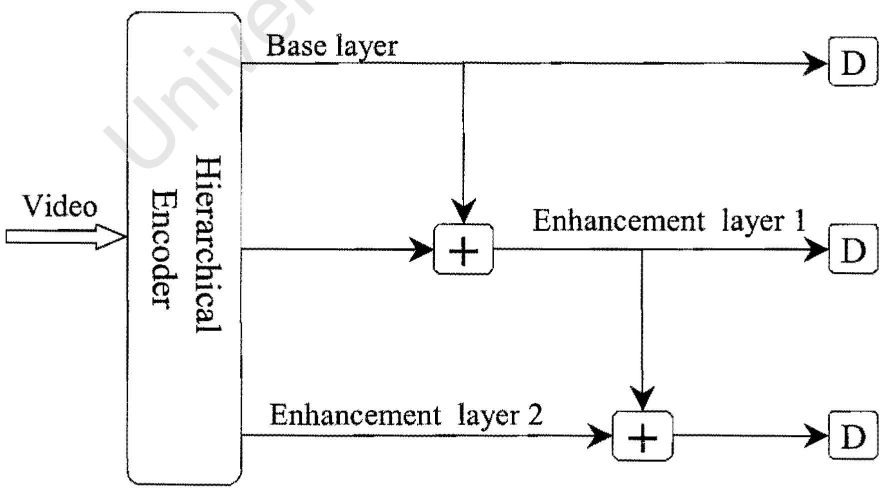


Figure 4.10: Multilayer video encoder

In this work, we focus on SNR scalability modelling, in which the base layer consists of a coarsely quantized version of the video, and the enhancement layers contain the refinement information. The SNR scalability relies on the DCT approach. However spatial scalability employs spatial pyramid encoding. In the temporal scalability approach frames are distributed between base and enhancement layers [82].

Currently, MPEG-4 supports frame-based temporal, object-based temporal, and frame-based spatial scalability. In general, two enhancement types can be discriminated: (1) the enhancement layer increases the resolution of a particular object or region of the base layer; (2) the enhancement layer increases the resolution of the entire base layer.

4.4.1.1 Temporal Scaling

Temporal scaling can be accomplished easily if each frame is compressed independently without motion compensation. In this case, the frames can be freely distributed over different layers. However, MPEG compression schemes take motion compensation into account between subsequent encoded frames. So, if we distribute them among a number of layers without taking the frame dependency into account, a receiver, which eventually receives some of these layers, will not be able to decode the video. There are two possible ways to scale motion compensated video. The first approach is to encode the video independently on each layer. This approach results in transmitting groups of pictures (GOPs) on each layer. Between such groups dependencies are avoided. The second approach takes the structure of the GOPs into account, as shown in Figure 4.11 where, a possible scaling method with three layers is described. This work adopts the second scaling approach, where all independent coded frames (I-frames) have to be transmitted in the base layer. On the second layer, the predictive-code frames (P-frames) are transmitted and the highest layer transports the bidirectionally predictive-coded frames (B-frames) [160].

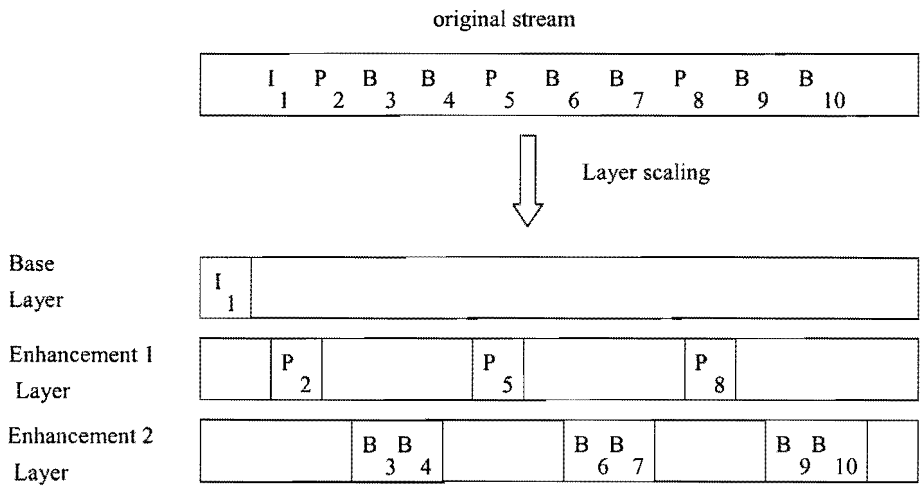


Figure 4.11: SNR temporal scaling

4.4.1.2 MPEG-2 Based on SNR Scalability

There are three major kinds of multilayer in case of video: spatial resolution, temporal resolution, and data-rate (SNR) scalabilities [146]. Spatial resolution scalability is functionality to decode images at different resolutions. Temporal scalability means that the refresh rate of the frames can be adjusted, and data-rate (SNR) scalability implies that any target data rate can be achieved from a single compressed bitstream according to the user requirements based on the available network bandwidth or system capability. The later scalability approach has been adopted in this work as shown in Chapters 5 and 6.

MPEG-2 offers a significant improvement over MPEG-1 by allowing hierarchical multilayered encoding. Its new extension is called Scalability and permits the reconstruction of a video stream from two or three sub-streams (or layers). MPEG-2 also standardizes a variety of algorithms for multilayer video, including SNR scalability. The base layer encoding process is identical to that for a non-layered encoder. The quantized DCT coefficients from the base layer (after being dequantized) are subtracted from the input DCT block. The resulting quantisation error from each block is next requantized more finely and encoded to form the enhancement-layer bitstream.

The construction of a base layer and one or two enhancement layers, requires that the MPEG-2 supports scalability. MPEG-2 has defined four different scalability modes, namely Data Partitioning, SNR Scalability, Spatial Scalability, and Temporal Scalability. In addition, the last three modes may be combined to form the Hybrid Scalability.

In this research, the video encoding is done using the SNR scalability, as in [148]. The SNR scalability mode, however, offers the best tradeoff between robustness against cell losses and design complexity.
In the next two subsections, the base and enhancement layer are described.

The Base Layer

ATM can support guaranteed CBR connections with extremely low cell loss ratios. Hence, we have chosen to encode the base layer using a closed loop rate control. In this type of coding a fixed quantity of bits are allocated to each GOP, which in turn is apportioned to the successive GOP pictures, and within pictures, to the pictures macroblocks. The number of generated bits is used in the feedback loop to act on the quantisation parameter used for each macroblock.

The Enhancement Layer

In order to preserve a constant video quality, we have chosen to code the enhancement layer in open loop coding. In this type of coding, there is no control on the bit rate produced by the encoder. Thus a fixed quantisation parameter is used for the whole video sequence, leaving the output stream fluctuate at its nature, and produce a VBR traffic.

4.4.1.3 MPEG-4 Content Based Scalability

The aim of scalable coding of video is to support receivers with different bandwidth or display capabilities or display requests to allow video database browsing and multiresolution playback of video content in multimedia environments. Another important purpose of scalable coding is to provide layered video bit stream, which is agreeable to prioritised transmission. The techniques used in the VM allow the content based access or transmission of arbitrary shaped VOPs at various temporal or spatial resolutions in contrast to the frame-based scalability methods introduced for MPEG-2. Receivers either not capable or willing to reconstruct the full resolution arbitrarily shaped VOPs can decode subsets of the layered bit stream to display the arbitrarily shaped VOPs content/objects at lower spatial or temporal resolution or with lower quality.

MPEG-4 will provide the tools and syntactic elements to achieve scalability with a fine granularity in terms of content, texture-SNR, shape-SNR, spatial and temporal scalability. These types of scalability will result in a very flexible content-based scaling of the video information. MPEG-4 video will support spatial/temporal texture scalability by allowing objects in a scene to be coded with a base layer and up to 4 enhancement layers. In the context of profiles, more specific requirements regarding the number of texture and shape scalable layers will be specified. MPEG-4 video will support joint coding of at least 4 views of a video scene concurrently. For any stereoscopic video, MPEG-4 will perform at least as well in exploiting redundancy as the MPEG-2 multi view profile.

Initially MPEG-4 targeted 32 levels of priority classes for Audio Video (AV) Objects. The current version 2 of MPEG-4 provides for decoding of still images, spatial scalability with up to 11 levels of granularity and also quality scalability up to the bit level. For video sequences, an initial maximum of 3 levels of granularity will be supported, but work is ongoing to raise this level to 9.

4.4.2 Impact of Cell Losses on Multilayer Encoding

In ATM, cell losses may have a very annoying impact on video quality. Therefore, the limitations on the Cell Loss Ratio (CLR) should be considered in order to prevent quality degradation. However, the target CLR for video is often 10^{-9} [149]. Therefore, the network will be operated uneconomically if it turns out to be overly precautions. There are overall few results reported on the perceptual aspects of video transfers and there are consequently few guidelines to follow, especially concerning acceptable loss levels [26].

In multilayer encoding schemes, tests have shown that by coding the video into multilayer, and by allowing cell losses to occur only in the enhancement layers, the overall coding could tolerate higher cell loss ratios than non-layered coding while preserving a consistent video quality as we have proved on Chapter 6.

Another technique to reduce the effect of ATM network impairments on video quality is to encode the video information differently so that the encoded bit stream is more robust to errors. Besides improving video quality, this technique must ensure that the encoded bit stream is compatible with standard video decoders. Two such techniques proposed in the literature are scalable, or layered, coding and temporal localization. Layered video coding techniques encode video into two or more layers; each contains different component(s) of the original video information. The video sequence is encoded at a low quality to form the base layer. The difference between the base quality and the original quality of the source is encoded as one or more enhancement layers. While decoding the base layer results in video of low quality, the decoding of the base layer with the enhancement layers result in video of improved quality. The base and the enhancement layers are sent over multiple Virtual connections (VCs) with different QoS requirements. Since the base layer is more significant to the decoding process than the enhancement layers, it is handled with a higher priority and a stricter QoS compared to the enhancement layers during transmission. As a result, the base layer is affected by network impairments to a lesser extent and maintains basic video quality when the enhancement layers are affected by network impairments. Figure 4.12 shows the functional blocks of the layered coding technique. Since temporal propagation of video distortions does not proceed beyond an error free I-frame in a MPEG sequence, reducing the size of the group of picture, i.e. reducing the number of frames between consecutive I-frames, should reduce the extent of such error propagation. Temporal localization techniques reduce the temporal spacing of I-frames and hence increase the proportion of I-frames in the coded sequence. Although the extent of temporal propagation of video distortions is reduced, this method has the disadvantage of decreasing the coding efficiency.

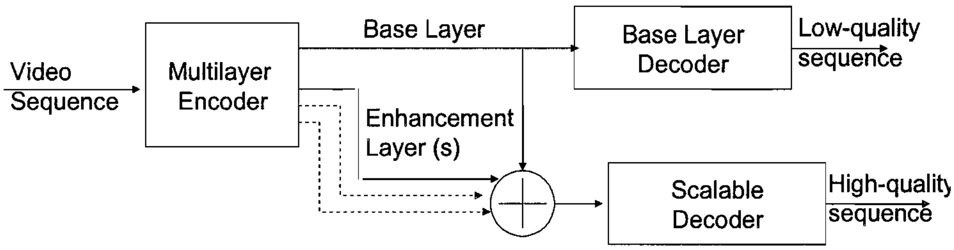


Figure 4.12: Block diagram of scalable or layered coding system

4.4.3 Multilayer Encoding for Multicasting

It is expected that there will be many types of multimedia terminals and that possibly many formats for representation of each medium type will exist [150]. This has been the case for traditional text-based computer communications (where the compatibility problems are considerably simpler than for multimedia). The presentation layer in the OSI architecture addresses the issues of data representation compatibility problems. When incompatible transmitters and receivers communicate, translation is necessary. This service can be provided in one of three places: (1) at the transmitter, (2) within the network, or (3) at the receiver.

If the third solution is possible, there is no real compatibility problem. The second approach is the typical solution (e.g., traditional protocol converters). The first solution is similar to the third in the point-to-point case, but is rather inefficient in the multipoint case. It requires the sender to translate its data format once for every incompatible receiver. This translation consumes sender resources that are usually highly contended. Furthermore, it requires the network to transport a higher volume of information because the sharing of links is not possible. In essence, severe compatibility problems limit the effectiveness of multicast solutions and the communication model reverts to a series of unicasts.

Facilitating translation at the receiver through synthesis of the signal from separate components can be achieved through multilayer encoding. In addition, network routing algorithms may take into account the different needs and capabilities of the destinations by, for example, forwarding only usable components to select destinations [151]. Furthermore, there is an interesting interaction between hierarchical coding and multicasting. Multilayer encoding enables destinations to adjust the quality of each signal they receive, independently and without the source actually being aware of this adjustment. This is a very important property considering the feedback control problems of multipoint communication, and can also be used to effectively address many compatibility problems.

Thus, multilayer encoding can help diminish compatibility problems for continuous media and also address other problems such as real-time delivery, error control, and network congestion control, as well as to minimize communications costs.

4.5 Video Traffic and ATM Connections

In ATM networks, one or several connections can be established to carry video traffic and control traffic for single direction video delivery. Three scenarios are listed and explained below:

- Single Connection: In this case, only one ATM connection is set up between the sender and the receiver and video information is serialized and transported over this VC together with control information;
- Dual Connections: In this scenario, two ATM connections are required. One connection is used for video traffic and the other is used to carry control information between the sender and the receiver. These two connections will obviously have very different traffic characteristics and are therefore setup with different QoS attributes;
- Multiple Connections: Another alternative is to use multiple connections for the different information components (video, audio, control etc).

In this research work, the multilayer video encoding constructs MPEG bit streams into a base layer (which carries important, base resolution information) and an enhancement layers (which carry quality and resolution enhancement information). Video traffic belongs to these layers has a different priority and carries different QoS attributes. During network congestion, ATM cells belonging to the enhancement layer with lower priority are discarded before information belonging to the base layer with higher priority is affected. This decreases the probability of any interruption in the delivery of video streams to a large extent. The synchronization between the various components can be embedded in each connection so that they can be combined at the receiver.

The use of one or more ATM connections to support video traffic is implementation-specific, i.e. determined by the architecture of the video application and the video encoding scheme employed. Since the aim of the underlying ATM network is to provide the services and resources to satisfy the connection requests from applications, it plays no part in deciding the number of connections used to carry video traffic. Chapter 5, proposes a dynamic bandwidth allocation based on the later scenario, which is establishes multiple connections to carry multilayer compressed video.

4.6 Modelling of Multilayer Encoded Video

Here, we have adopted the single-source model shown in Figure 4.13 for VBR video to model the enhancement layer of a multilayer video encoding (e.g. we consider a two-layer video encoder). Our primary goal is to explore the relative efficiency of transporting video streams on multilayer video format. Early results [152][153][154] indicated that while multilayer video could withstand higher cell-loss rates than non-layered video, its overhead bit rate was large enough that it still had worse multiplexing performance. However, the recent multilayer compression algorithms adopted in this research are more efficient and, hence, our results from the simulation experiments reported on Chapter 6 indicated, that a multilayer video encoding has provided a better throughput, lower cell-loss, and higher multiplexing gains than non-layered video.

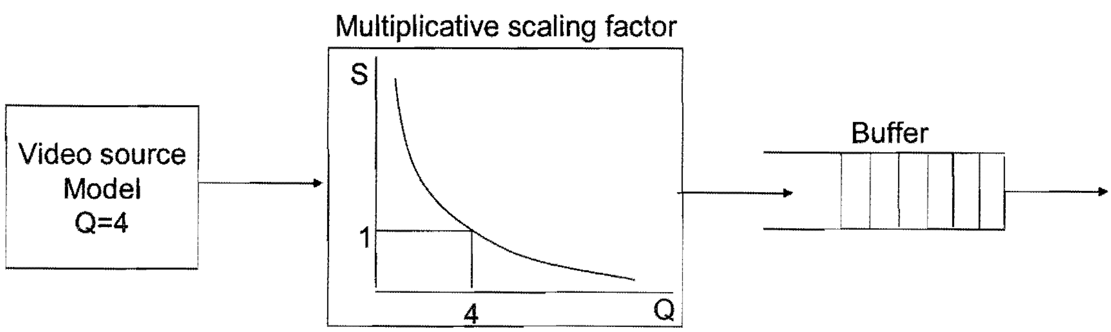


Figure 4.13: Rate control for one-layer video encoder

Studying the characteristics of multilayered video, as opposed to non-layer video, vastly increases the number of free parameters. For example, each different base-layer bit rate produces different characteristics in the enhancement layers bit rate. To study this effect without a model, it would be necessary to encode the video for each different base layer bit rate of interest. Furthermore, the enhancement layers bit-rate characteristics also depend on the particular rate-control algorithm used to enforce a constant rate in the base layer. If the impacts of different rate-control algorithms are to be studied, traffic measurements must again be performed for each different algorithm at each different base layer bit rate of interest. Clearly, the amount of data gathering quickly becomes prohibitive if actual traces are to be used. The multilayer video encoding model based on traffic measurements gathered during a single set of encoding using different base layer quantizer step sizes is presented [155]. Both quantizers are fixed throughout encoding, but the enhancement layer quantizers are the same for each encoding so that each encoding has essentially equivalent video quality. The model is used to characterize the enhancement layers traffic generated for a specific base layer bit rate and rate-control algorithm. The explicitly of the solution allows different base layer bit rates and rate-control algorithms to be analysed without gathering additional data. *Rodriguez–Dagnino et al.* used a similar approach in [156], where they proposed a set of fundamental parameters that characterize a bit-rate process from which the effect of different encoding schemes can be determined.

In [155], a model for one-layer video that incorporates feedback for buffer control, as shown in Figure 4.13 was proposed. This model enables the characterization of a video source compressed for a constant-rate channel. The source model generates the output bit rate of a constant quality VBR encoder with a quantisation parameter fixed at $Q = 4$, while a multiplicative scaling factor accounts for variations in bit rate caused by using a different quantisation parameter to ensure no buffer overflow or underflow. The model in Figure 4.13 was shown to accurately capture the video quality when CBR transport is used [157]. This type of model can also be used to shape the one-layer bit rate to meet a given UPC constraint.

The above modelling concept could be extended, as shown in Figure 4.14, to obtain a model for multilayer video when the base layer is transported with CBR, as we have proposed on Chapter 5. Base layer and enhancement layers data measured using a constant quantizer step size of 8 in the base layer, and in the enhancement layers, are used as reference input for generating the multilayer video.

Since the base layer is to be transported at a CBR, the quantisation parameter actually used when encoding the base layer would have to be chosen such that the base-layer buffer neither overflows nor underflows. By changing the actual Q_b to be different than 8, the base layer bit rate varies, as does the enhancement layer bit rate. These variations are well characterized by a multiplicative scaling factor plus an offset, $R_Q = A_Q R_8 + B_Q$. The parameters A_Q and B_Q for these scaling functions are derived from measurements of base and enhancement layer signals encoded at a range of base layer quantisation step sizes varying from 4 to 30. A pair wise regression analysis on these measurements provides the scaling parameters that map the bit rate corresponding to a reference quantisation step size to the approximate bit rate for any other quantisation parameter. Separate scaling functions are used for the I and P frames. As depicted in Figure 4.14, when Q_b increases, less data are coded in the base layer and more is left to be coded in the enhancement layer.

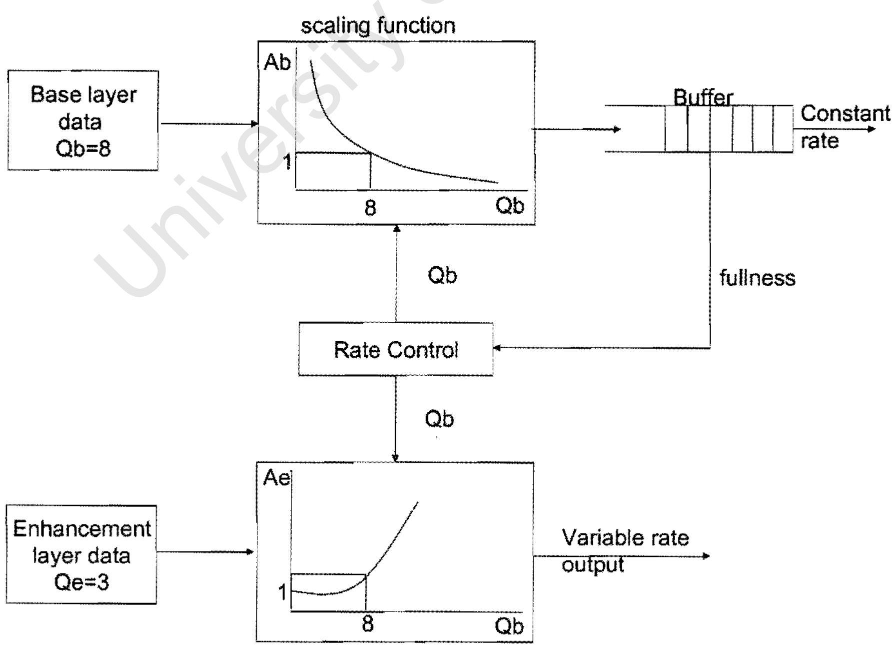


Figure 4.14: Effect of base layer rate control on enhancement layer bit rate

The rate-control algorithm used for buffer control is as follows. The quantisation step (q-step) is adjusted based solely on the buffer fullness. When the buffer is empty, the minimum q-step of 4 is used, while when the buffer is full, the maximum q-step of 30 is used. Between these end points, the chosen q-step is an exponential function of the buffer fullness. (We also considered a linear function of the buffer fullness and found no significant statistical difference in the results.) For each value of the base layer CBR rate chosen, the rate-control algorithm generates a different sequence for the VBR enhancement layer. This data is then fit using the described source. The model-generated data is then used to compare the tradeoffs in one-layer and multilayer encoding.

4.7 Conclusions

This chapter has described the MPEG compression standards for encoding video and appropriate transmission scheme over ATM networks. We have given a path for the design and implementation of the emulated network by presenting the essential concepts on MPEG video compression (MPEG-2 & MPEG-4) standards, and multilayer compression techniques. However, MPEG compressed video is very sensitive to data loss; not only may corruption caused by the loss of an ATM cell be quite severe, but the use of inter-frame predictive coding may cause the corruption to persist or even spread over a period of several frames. In order to lessen the impact of cell loss on compressed video, the proposal is to use multilayer compression techniques for splitting the coded bit stream into a number of layers of differing importance to the decoded video quality, and transmitting these layers at different ATM connection with different QoS requirements. The high priority layer is the base layer, which is sent over a CBR connection and protected against cell loss, while a VBR connection is established for each enhancement layer in order to maintain a consistent video quality. The robustness against cell loss feature of multilayer compression techniques has received a considerable attention due to the present heterogeneity nature of ATM networks. Because of their capabilities to handle layered video streams MPEG-2 and especially MPEG-4 are promising frameworks for ATM and Internet-based multimedia networks. However, our proposed bandwidth allocation model in this research has adopted the above multilayer video compression technique as a traffic source to develop a more flexible bandwidth management scheme to control the burstiness of compressed video. The great improvements gained from the use of such multilayer layered coding and transmission models raise the question of how much the improvement might be gained on the network utilisation and performance. This question will be addressed in the subsequent chapters.

Chapter 5

Renegotiation-Based Dynamic Bandwidth Allocation

Future integrated broadband networks are expected to support applications with diverse traffic characteristics and performance requirements. There are three generic types of traffic for broadband ATM networks: constant bit rate (CBR), variable bit rate (VBR), and best-effort traffic. Among these, real-time VBR traffic poses unique challenges on its bandwidth management and allocation [161][162][163]. Whereas bandwidth allocation schemes are easy to implement for CBR traffic, and there are numerous bandwidth allocation approaches based on feedback and re-transmission for best-effort traffic, there is however no consensus on the strategy for real-time VBR traffic, in particular, compressed video. This is mainly due to two conflicting design goals: good quality of service (QoS) and high network utilisation [196].

The compressed MPEG video sequences are generally highly bursty in nature, i.e. these may generate various amounts of data during varying time intervals, where the amount of aggregate incoming traffic is greater than the outgoing link speed and cells have to be buffered. If the situation persists, cells will be dropped due to buffer overflows, which will in turn cause a degradation of application's QoS. Thus, in order to guarantee the transmission of compressed video with consistent visual quality, the maximum bandwidth required for transmitting encoded data has to be allocated.

Smoothing [164][165] is one approach that has been utilized in an effort to reduce the burstiness of video traffic, and thus efficiently enhancing network utilisation. However, since the smoothing schemes do not take into consideration the scenery changes that are caused by very different statistical and delay sensitivity of the video traffic, the problem still remains. In this chapter we present our contribution to solve the problems caused by bursty traffics on network utilisation by proposing a more flexible renegotiation-based dynamic bandwidth allocation protocol, instead of allocating a static effective bandwidth at call level only. Our renegotiation-based dynamic bandwidth allocation protocol is capable of rapidly adapting to changes in network conditions, providing high quality video service and good network resources utilisation. The protocol renegotiates the allocated bandwidth during call lifetime using traffic predictors. The predictors are used to forecast the required bandwidth to accommodate the incoming bursts, given that the source is capable to accept graceful degradation of QoS when the predicted bandwidth cannot be provided.

Our proposal is evaluated with MPEG-4 video stream. The video stream is found to be highly bursty and very difficult to be characterized accurately. The video source is compressed on multilayer (hierarchical) format. Therefore, in the next section, we look at compressed video transmission issues.

5.1 Compressed Video Transmission

There are several ways of adjusting the quality of compressed video stream. These may include adaptive encoding, switching among multiple pre-encoded versions, and multilayer (hierarchical) encoding. Considering adaptive encoding the encoder re-quantises data *on-the-fly* based on network feedback [78][79][80][81]. Since encoding is CPU-intensive, sources are unlikely to perform that for a large number of receivers. An alternative option is for the source to retain several versions of each video stream with a different resolution. Thus depending on the network's available bandwidth, the source would switch between the lower and higher resolution quality streams as required. This option requires large buffers at traffic source.

Hierarchical or multilayer encoding is also another option that is part of a family of signal representation techniques. Generally, it refers to an approach in which the video compression source scales its output compression rate by partitioning the video stream into sub-streams or layers, each layer representing a portion of the signal, as was discussed in section 4.4. The greater the number of layers received at the end stations, the better is the quality of the reconstructed signal [82]. In this approach as more bandwidth becomes available, more layers of the encoded stream are delivered. If the available bandwidth decreases, the source would then drop some of the layers. The multilayer encoding usually have the decoding constraint that a particular enhancement layer can only be decoded if all the lower quality layers have been received. There are several advantages of multilayer encoding, including that of less storage requirements at the source, and provision of an opportunity for selective prioritisation of the important information. The design of an effective multilayered video transmission system basically entails the design of an efficient drop and adds a layered mechanism that can maximize the perceptual quality of the received video stream.

5.2 Renegotiation of the Allocated Bandwidth

The renegotiation-based dynamic bandwidth allocation protocol proposed in this chapter offers the possibility of implementing such a protocol, on an end-to-end basis, both at the application (QoS profile) and at the network (traffic contract) levels. During a session when, a change occurs in the state of the application, in accordance with the hierarchical model, an indication is made by the application to the bandwidth renegotiation unit in Figure 5.4, which in turn will ask the network to change the state of the connection. Such a dynamic mechanism offers to the application the guarantee of acceding to the requested level of quality (service guarantee), while letting flexible applications reach unused resources (higher utilisation rate).

Our approach uses two kinds of bandwidth allocation strategies: permanent allocation and released-on-demand allocation. Our approach is compared to the traditional static bandwidth allocation (i.e. the bandwidth is allocated statistically at call set up phase only) protocols, which uses permanent allocation only. Figure 5.1 illustrates the differences between our proposal protocol and the traditional static protocols. The main descriptions of the allocation strategies are as follows. In the permanent bandwidth allocation strategy, a static amount of bandwidth is deterministically or statistically allocated to a traffic source, where the bandwidth is permanently allocated at call setup phase. In our protocol, the permanent bandwidth allocation strategy is used to allocate a deterministic amount of bandwidth to the base layer of the multilayer compressed video with CBR service.

In the released-on-demand allocation strategy, the bandwidth is allocated adaptively to the enhancement layers for flexible applications with higher QoS parameters than the base layer.

The renegotiation-based dynamic bandwidth allocation approach improves the bandwidth utilisation, and consequently a higher number of connections could be accepted, which normally be rejected in static allocation mode.

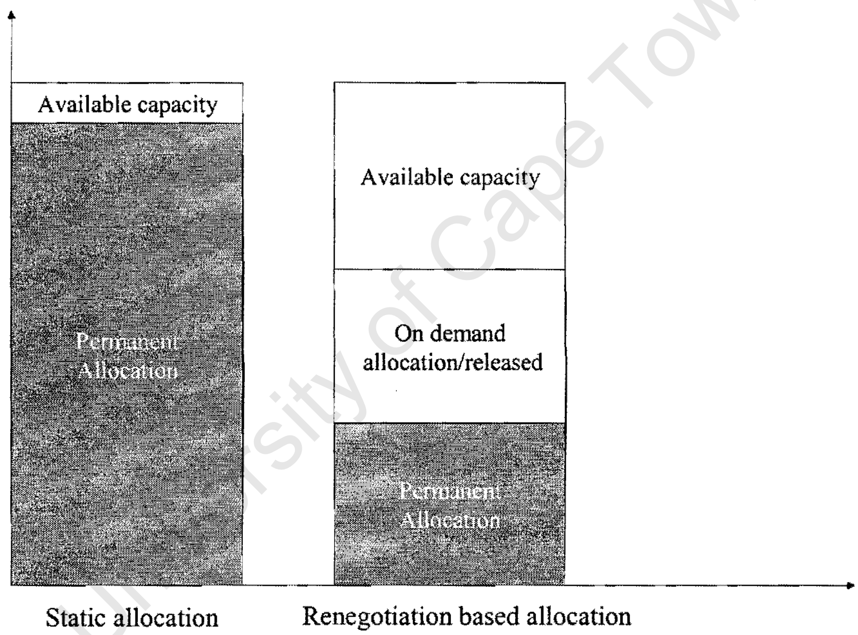


Figure 5.1: Traditional static and dynamic bandwidth allocation schemes

The idea of dynamic allocation of bandwidth has been proposed in the literature and demonstrated using different approaches [78][83][84][2]. *Vickers et. al* have proposed a new ATM service called quasi-VBR for video. This service is based on Enhanced Proportional Rate Control Algorithm (EPRCA) for ABR service [83]. The quasi-VBR video service could be provided through a combination of proportional rate control, a modified T_1/T_2 service and adaptive quantisation step encoding. However, this approach requires that the ATM standards be modified. The service is also based on closed loop congestion control policy, which is irrelevant for real-time applications. *Seckin* and *Gloshani* approaches the adaptability on the allocated bandwidth by proposing a new multilevel priority approach combined with buffer management [84]. The new traffic management model is able to handle more than two levels of prioritisation for video transmission. The proposed algorithm requires that a new AAL layer protocol has to be defined and moves the switching to AAL level. In this research the dynamic allocation of bandwidth is based on renegotiation of traffic source QoS parameters [57]. The proposed approach allows the traffic source to change its traffic descriptor dynamically according to the network bandwidth availability.

The renegotiation-based dynamic bandwidth allocation scheme establishes several connections to transfer multilayer encoded video stream, as shown in Figure 5.2. Each connection has different QoS parameters and traffic definitions. Generally, the base layer, since it carries more important information than the enhancement layers, is transmitted separately on a more reliable connection, such as CBR. Each receiver must receive a base layer quality on a certain virtual connection (VC) as CBR service connection and sufficient number of enhanced layers in other VCs as VBR service connection(s) depending on the network available bandwidth [57][170].

The renegotiation-based dynamic bandwidth allocation protocol uses a new multilevel traffic descriptor to map the multilayer video stream into ATM virtual connections. Next section describes the proposed traffic descriptor.

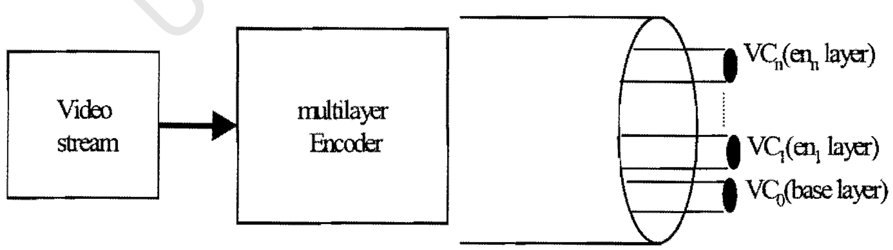


Figure 5.2: Multilayer video Transmitter

5.3 Multilevel Traffic Descriptor

The multilevel traffic profile/descriptor model is also proposed as shown in Figure 5.3 [57]. The multilevel traffic descriptor is based on the alternative traffic descriptor model proposed by ITU-T series of recommendations Q2963.3 [93] which specifies the procedure of the modification of ATM traffic parameters during the connection lifetime. This model not only enhances the connection acceptation probability, but also enables the network to dynamically modify the connection's characteristics, based on the global state of the network after the establishment phase is over (for flexible applications only). To setup a video connection using multilevel QoS profile model, the user/application must first specify its QoS specification on multilevel format, as shown in Figure 5.3. This can be done through a user selection interface or by an application programmer. The multilevel QoS profile is used for negotiations and renegotiation with the network to select the required QoS, as will be discussed later in this chapter.

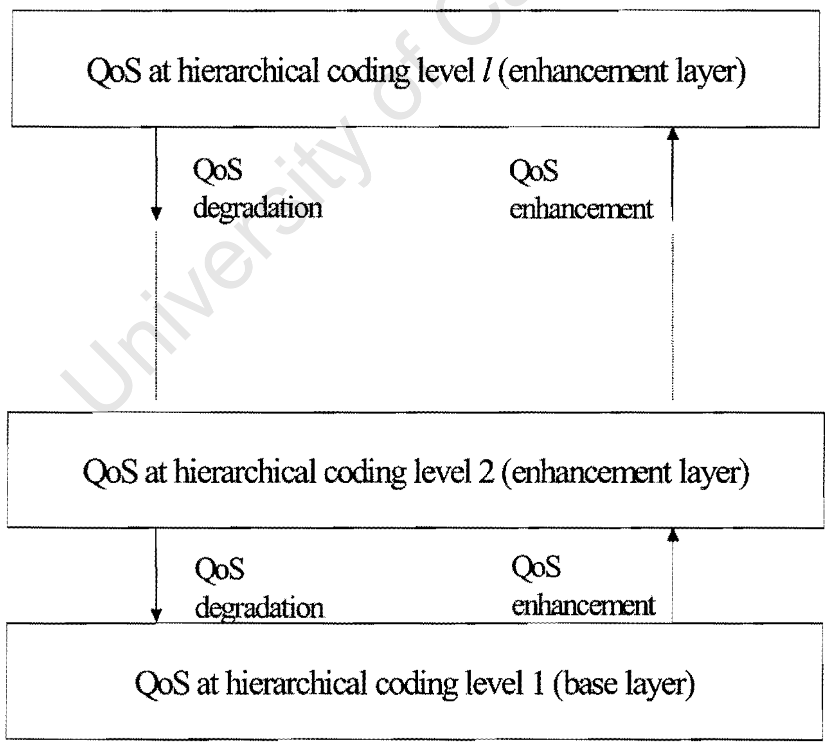


Figure 5.3: l-level traffic descriptor model

5.4 Protocol Description

The framework of our proposed protocol is illustrated on Figure 5.4, which we were earlier presented [57]. The framework shows the different components required for protocol implementation. The renegotiation-based bandwidth allocation protocol's procedure is presented as a flow chart in Figure 5.5.

At call setup, a CBR connection is established for the base layer and a VBR connection for each enhancement layer of the multilayer compressed video. The bandwidth is allocated dynamically for VBR enhancement layers connections depending on bandwidth availability. The lower enhancement layer is generally accorded higher priority than the higher layers when it comes to bandwidth allocation. A new enhancement layer can be added as soon as the available bandwidth exceeds the consumption rate of the existing enhancement layers plus that of the new enhancement layer. However, this would be problematic, as without prior knowledge of future available bandwidth we cannot decide how many enhancement layers could be sent or dropped during the renegotiation process between the network and the video source. To overcome this problem the on-line predictors are used to forecast the enhancement layers bandwidth requirement, as shown in Figure 5.4.

The bandwidth renegotiation unit in Figure 5.4 makes its decision (based on the network available bandwidth) whether it is able to grant the predicted bandwidth or not. If the network has sufficient bandwidth, then the requested resources will be granted, otherwise a graceful degradation on source's QoS parameters should be considered at that time. This is done by adopting a multilevel traffic descriptor discussed in section 5.3. In this case the traffic contract has to be modified with degraded QoS parameters during the congestion period [57][170]. This bandwidth constraint for adding a new layer is still not sufficiently conservative, as it may result in several layers being added and dropped with each cycle of renegotiation. Such rapid changes in quality would be disconcerting for the viewer. One way to prevent rapid changes in quality is to add buffering condition, such that adding a new enhancement layer does not endanger existing enhancement layers. Thus, the sender may add a new layer when:

1. The instantaneous available bandwidth is greater than the required service rate of base layer and the existing enhancement layers plus the new enhancement layer,

$$R > R_b + C_{new} + \sum_{i=1}^{n_a} C_i \tag{5.1}$$

2. There is sufficient total buffering at the receiver to survive an immediate backoff and continue playing all the existing enhancement layers plus the new enhancement layer.

$$B_b + \sum_{i=1}^{n_a} buf_i \geq \frac{\left(R_b + \sum_{i=1}^{n_a} C_i - \frac{R}{2}\right)^2}{2S} \tag{5.2}$$

where

R is the current transmission rate,

R_b is the transmission rate for base layer,

n_a is the number of currently active enhancement layers,

buf_i is the amount of buffered data for enhancement layer i,

B_b is the amount of buffer space allocated to base layer connection,

S is the rate of linear increase in service rate, and

C_i is the required service rate for enhancement layer i.

The above constraints are the minimum requirements for adding a new enhancement layer. If these constraints are held, a new layer can be kept for a reasonable period of time during the renegotiation periods.

If the network available bandwidth or amount of buffering at destination is less than the required bandwidth to transport all enhancement layers a backoff occurs, the correct action is to immediately drop the highest enhancement layer. This is will reduce the required bandwidth to transport the multilayer video stream $(\sum_{i=1}^{n_a} C_i)$ and hence reduces the required bandwidth or buffering at destination for recovery. If the network available bandwidth is still insufficient, the server should iteratively drop the highest enhancement layer until the amount of bandwidth is sufficient.

$$While\left(R_b + \sum_{i=1}^{n_a} C_i > R + \sqrt{2S(B_b + \sum_{i=1}^{n_a} buf_i)} \right) \tag{5.2}$$

$$Do \quad n_a = n_a - 1$$

We should remember that base layer is sent over a CBR connection, where a static amount of bandwidth always is guaranteed to be available.

The protocol uses burst predictors to forecast the required bandwidth for enhancement layers. The next section discusses the different burst predictors and their characteristics.

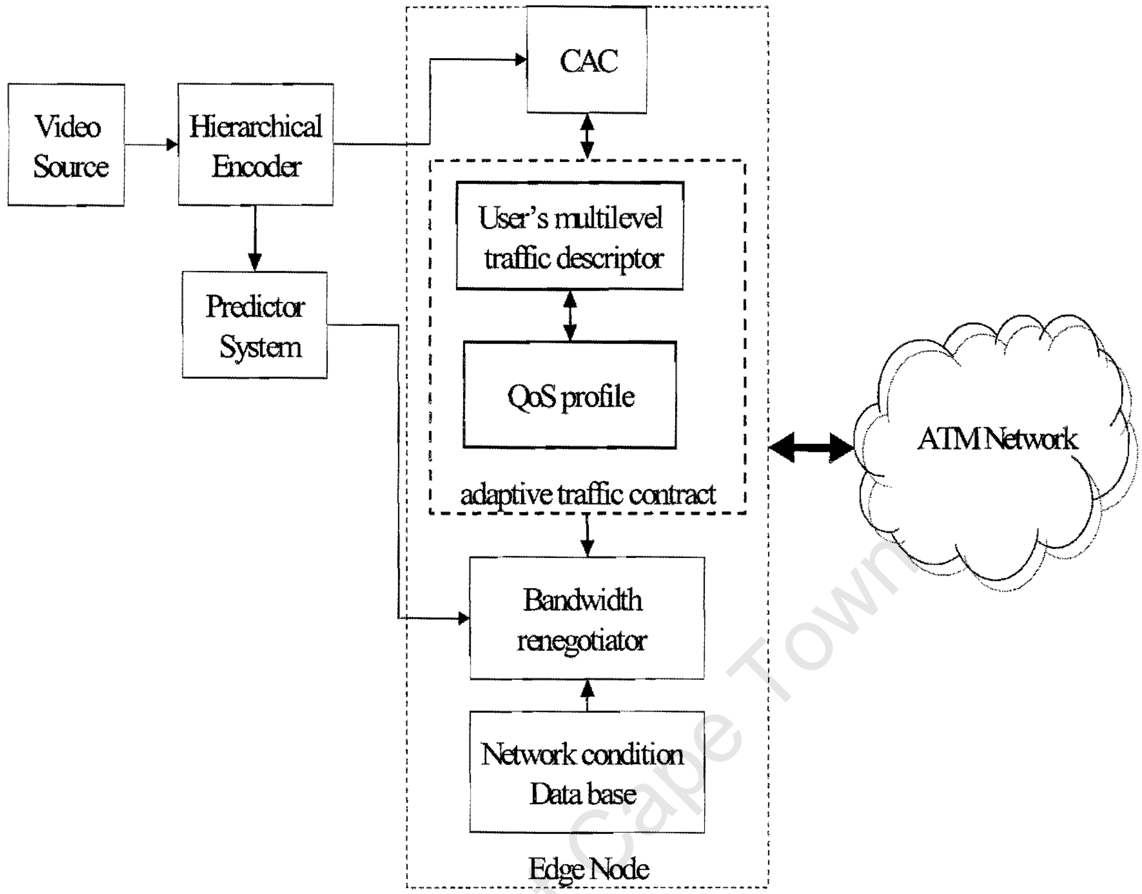


Figure 5.4: Protocol's framework

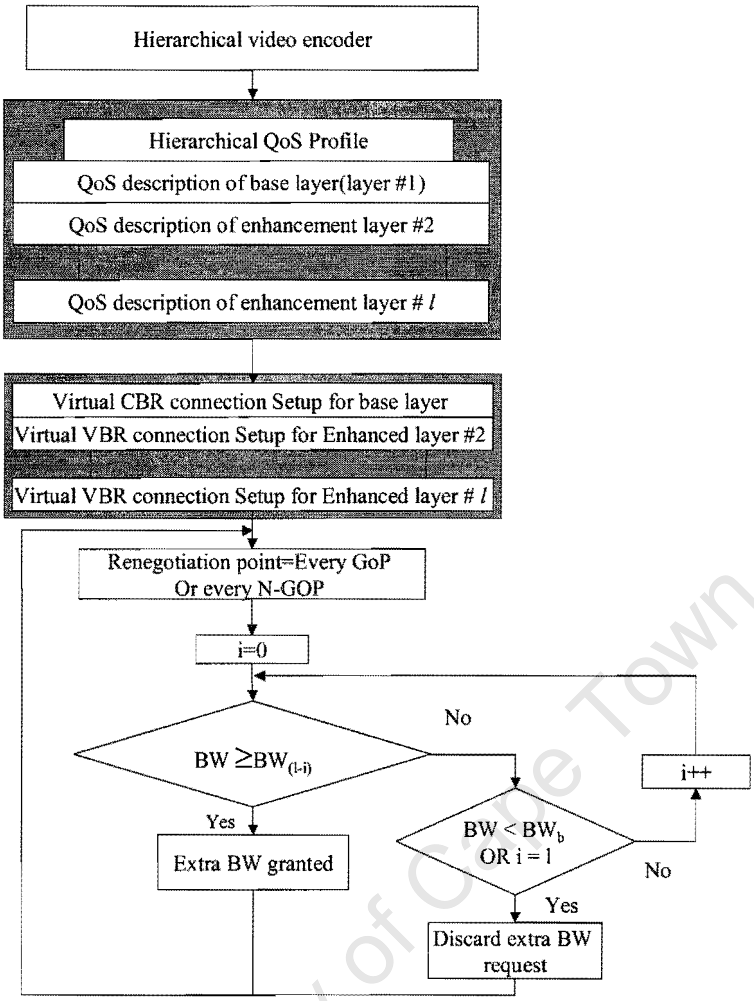


Figure 5.5: Renegotiation-based bandwidth allocation protocol

5.5 Burst Prediction

The protocol's framework in Figure 5.4 shows the predictor unit that is used to forecast the future required bandwidth for enhancement layers. For non-real time applications, burst prediction is not required and bandwidth allocation may be implemented more easily and efficiently because the frame size is known before transmission. However, for real-time application, at the time of renegotiation, the required bandwidth for enhancement layers have to be predicted first in order to effectively implement the protocol. The contribution of our work is instead of allocating a static amount of bandwidth at call setup phase only, we propose, that the bandwidth is allocated dynamically during call lifetime using on-line predictors. This is given that the source must accept graceful degradation on QoS when the predicted bandwidth cannot be provided.

The better prediction system is one that allows a relatively longer lead-time to predict any incoming burst's bandwidth requirement. If the required bandwidth is not provided on time, the burst will be considered useless and will be discarded.

In this section three prediction schemes are discussed as on-line predictors for forecasting the burst's bandwidth. The first scheme is based on the least mean square method (LMS) error linear predictor [171]. The LMS is an adaptive approach, which does not require prior knowledge of the video statistics, and does not assume stationary. Therefore, it can be used efficiently as on-line predictor for real-time applications. The second scheme is based on the Recursive Least Square (RLS) method. It is also a linear predictor like LMS, but its adaptive approach has the advantage of fast convergence and robustness over the LMS adaptive approach [166]. The third scheme takes the Artificial Neural Network (ANN) approach. The ANN approach is used to predict the burst's bandwidth requirement for real-time video.

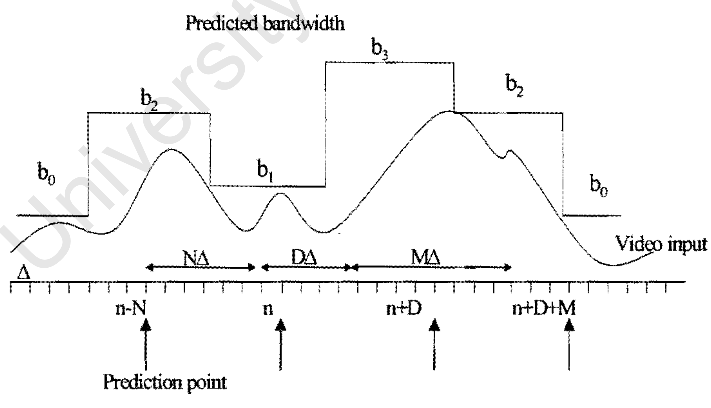


Figure 5.6: Burst bandwidth prediction in a video signal

The burst's bandwidth requirement can be renegotiated with the network intermittently using on-line prediction by filtered input video signal $x_L(t)$ [166]. The renegotiation interval cannot be too short since it is limited by the protocol processing time at each node. Let the video signal be sampled at time unit Δ and denoted by $x_L(n)$ sampled at the n-th Δ unit as shown in Figure 5.6. The bandwidth is renegotiated at interval of $M\Delta$, which is equivalent, for example, to GOP duration. There will be $D\Delta$ lead time for computation of the prediction protocol and the processing time for renegotiation-based dynamic bandwidth allocation protocol. That is, if the prediction starts at the n-th unit, the bandwidth will be renegotiated at the $(n+D)$-th unit. Consequently, the next bandwidth renegotiation will occur at the $(n+D+M)$-th unit. The prediction of the input rate at each consecutive unit of renegotiation interval, denoted by $\hat{x}_L(n+D+l)$ at $l = 0,1,...,M$, is made on the basis of $(N+1)$ consecutive observations collected at the n-th group of pictures. In other words, the predictions $\{\hat{x}_L(n+D+0), \hat{x}_L(n+D+1), ... \hat{x}_L(n+D+M)\}$ are made on the basis of $\{x_L(n-N), x_L(n-N+1), ..., x_L(n)\}$.

In our protocol, Predictors are using RM cells within a sender equipment loop to provide the renegotiation unit in Figure 5.4 with predicted bandwidth information. We propose to use the 3 bit-PTI field on RM's cell header as an implicit request of the predicted bandwidth for every layer. The PTI field can offer $l = 2^3$ different bandwidth predicted levels $b_0, b_1, ..., b_{l-1}$ for each layer as shown in Figure 5.6. The RM cell is used to the trigger the renegotiation process. The bandwidth renegotiation interval time is controlled by how frequently the predictor sends the RM cells to the renegotiation control unit. Hence, in our protocol we suggest that the minimum allowed renegotiation interval is GOP transmission time. When the renegotiation process take place every GOP, only one predictor for each enhanced layer is required. Thus an RM cell will be sent with every group of pictures GOP or a couple of group of pictures (N-GOP).

Each of the previous predictors has its own strength and weakness in prediction design. In contrast to the LMS and RLS approaches, the ANN approach requires a training period with off-line computation time required and its solution may not be generally applied. However, the ANN approach requires a much less on-line computation time and no initial transit state for convergence than LMS and RLS approaches. On the other hand, the RLS and LMS are linear predictors, whereas the ANN approach is high-degree polynomial non-linear predictor. Furthermore, the

ANN approaches offers longer prediction lead times compared to that of the linear RLS and LMS prediction methods to achieve virtually the same performance.

5.6 Implementation Issues

A requirement for renegotiation-based dynamic bandwidth allocation is that the network allows renegotiations of the source traffic descriptor. The ITU-T Q.2963 defines an extension to ITU-TQ.2931 signalling, allowing renegotiation messages [92][93]. The ATM Forum has not defined renegotiation messages yet, but renegotiation is planned as a future work item for the next releases. Thus, it is essential to extend the current ATM signalling protocols and the native ATM Application Program Interface (API) to allow renegotiation to implement a dynamic bandwidth allocation schemes. The capability described in ITU-T Q2963 enables the applications to modify the ATM traffic descriptor with renegotiation for connections that have already been established, as shown in Figure 5.7 [93]. The recommendation specifies the procedure of the modification with negotiation of PCR, SCR, and MBS using either an alternative ATM traffic descriptor information element or a minimum acceptable ATM traffic descriptor information element. ATM traffic descriptor modification with renegotiation is applicable to all connection-oriented telecommunication services that are based on single point-to-point connections. New modifications are required to allow applications to modify their QoS parameters during a connection lifetime. Also, the current UNI and NNI signalling protocols have to be extended to incorporate new control message types for renegotiation requests and acknowledgements between applications and switches over transmission path of a connection.

Our proposed protocol seeks the extra resources on the transmission path that has been established during call setup phase. Thus, when the application requests a modification to the multilevel traffic descriptor, which has been discussed in section 5.3, a *MODIFY REQUEST* is sent to switches over the transmission path. At each switch along the path, the admission control protocol has to be modified to determine whether to accept changes to ongoing connections. If the requests for modification succeed at all switches over the path, an *ACKNOWLEDGEMENT* is sent back to the application. As the acknowledgement propagates back to the application along the path, the modified bandwidth allocation is installed at intermediate switches.

In the current native ATM API, the ATM data transfer and connection control operations do not take place concurrently [169]. For example, the control operation "*atm-connect*" must be first completed before the data transfer can begin. However, in ATM networks both control operation and data transfer in ATM are synchronous. Thus, such a situation doesn't suit renegotiation-based dynamic allocation schemes. To overcome this problem, the source must continue sending traffic that the receiver receives uninterrupted as well while the renegotiation process takes place between the application and the network. Thus, both ATM API and the signalling stack protocol must be modified to provide a functionality that separates data traffic transfer and control signalling messages and to make the distinction visible to the application. A simpler way of separating the data traffic from control signalling (renegotiation requests) traffic is to use two channels within the transmission path; one channel performs data transfer operations and the other handles renegotiation requests. Even with the separation of the data and signalling traffic, there are still other problems that remain. The ATM API library converts the signalling and data traffic into Unix streams messages that are passed down to ATM driver streams that do not differentiate between the signalling and data traffic. Thus, the interface has to provide a high priority to the signalling streams.

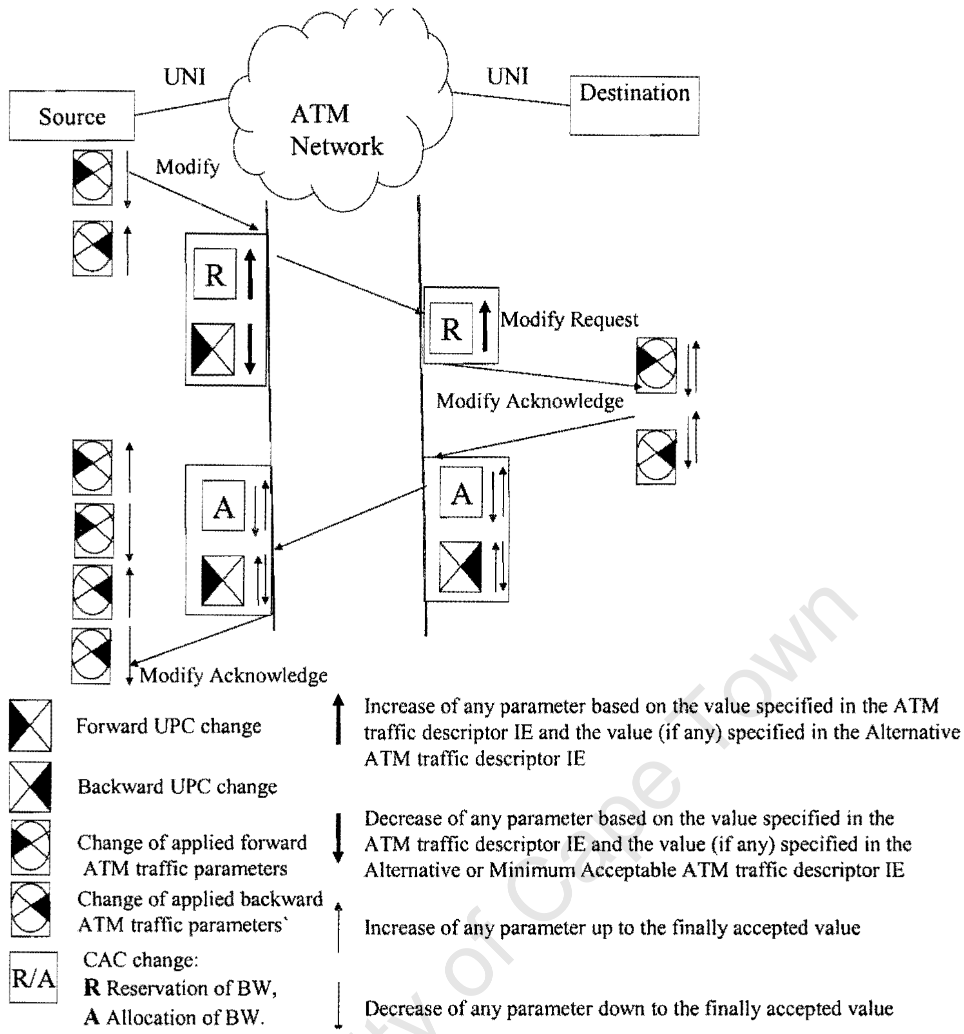


Figure 5.7: Connection modification procedures

5.7 Conclusions

In this chapter, we proposed a renegotiation-based bandwidth allocation protocol to transport a multilayered compressed video for broadband ATM networks. The protocol aims to find a solution to the problem of varying bandwidth constraints over band-limited networks, e.g. wireless and multicast applications. Since the VBR video is very bursty in nature, multilayer video encoders, like MPEG- 4, for example, may not alone be sufficient to achieve a high level of video quality and network utilisation because of network bandwidth availability and source bit rate often vary from time to time. In our model, we have proposed a multilevel traffic descriptor/profile for expressing user level QoS preferences. This offers a means to specify the main QoS parameters to be guarantee by the network. While keeping network level QoS parameters unchanged, by modifying quantitative parameters of the connection using traffic descriptors, we got a tradeoff between static bandwidth allocation with a hard QoS grantee but with low bandwidth utilisation and no reservation-based bandwidth allocation schemes with high bandwidth utilisation.

To improve bandwidth utilisation of such a network and to optimise the quality of the received video at the destination, the network must dynamically adjust the allocated bandwidth of the connection during its lifetime. Also in this model, when the network cannot provide extra resources, the applications will then gracefully degrade their QoS and, consequently, the bandwidth requirement based on the currently available bandwidth.

The renegotiation-based dynamic bandwidth allocation approach could be adapted for real-time applications because of the following reasons:

- It seeks the required extra bandwidth on the same transmission path, which has been already established during call set up phase.
- It uses on–line predictors to forecast the required bandwidth in advance.
- The renegotiation process during the call lifetime does not involve any routing or inquire any require any house-keeping records.

The aim of this approach is to find the required resources with minimum signalling overhead and processing time delay to accommodate bursts from real-time applications. This was accomplished by limiting the reallocation of bandwidth on the transmission path that has been established during call set up phase.
In the next chapter, the proposed protocol is evaluated using simulation experiments.

Chapter 6

Performance Evaluation of Renegotiation-Based Dynamic Bandwidth Allocation

This chapter evaluates the performance of the renegotiation-based dynamic bandwidth allocation protocol proposed in Chapter 5. We have conducted several simulation experiments using MPEG-4 traces to evaluate the performance of the proposed protocol. We start with description of MPEG-4 traces used in the simulation, then we proceed to present the performance results in terms of cell loss rate, throughput, resources utilisation, and signal to noise ratio (SNR). The impact of renegotiation on the network performance is also evaluated.

6.1 Simulation Model

The performance of the renegotiation-based dynamic bandwidth allocation protocol using multilayer video compression is evaluated and compared with traditional static bandwidth allocation scheme that uses non-layered encoded video where a statistical bandwidth is allocated permanently at call setup phase [197]. Bandwidth availability is assumed to be limited and fluctuating as a *saw-tooth* function as shown in Figure 6.1 [78]. This is because our protocol is proposed to work under situations of high bandwidth contention. When this is not the case both traditional static and dynamic protocols will perform the same.

The traces of actual video sequences "*Star Wars*" (see Figure 3.4 and Figure.3.5) are used with CIF sequences format and encoded with MPEG-4 at 10 frames per GOP [115]. The cell rates used in the simulation are calculated on the payload capacity of 48 bytes per ATM cell. We use the typical three-layer temporal-scalable MPEG 4 frame structure that was discussed in Chapter 4.

Two scenarios were studied through the simulation: firstly, the performance of traditional static bandwidth allocation scheme using non-layered encoded video, and secondly, the proposed renegotiation-based dynamic bandwidth allocation protocol using a multilayer encoded video [56][57].

The base layer bandwidth requirement is assumed to be available all the time. Thus, in our proposed model, the service rate is adjusted dynamically according to the available bandwidth. However, in the traditional static bandwidth allocation case the service rate is assumed as a VBR service rate.

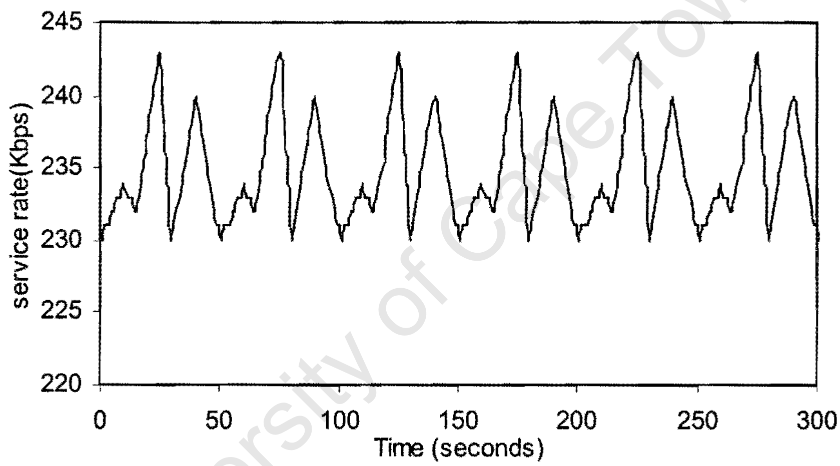


Figure 6.1: Service rate

Our model assumes the loss is occurred only from the unavailability of bandwidth. A GOP will be lost if the network available bandwidth is not sufficient to accommodates the whole GOP. The loss is calculated by the following equation:

$$Loss = \begin{cases} 0 & \mu \geq \lambda_{base} + \lambda_{en1} + \lambda_{en2} \\ \sum_{i=1}^{N} fr_i^{en2} & \lambda_{base} + \lambda_{en1} \leq \mu \leq \lambda_{en2} \\ \sum_{i=1}^{N} fr_i^{en2} + fr_i^{en1} & \lambda_{base} \leq \mu \leq \lambda_{en1} + \lambda_{en2} \end{cases} \tag{6.1}$$

where, fr_i^{en1}, and fr_i^{en2} are frame sizes for enhancement 1 and enhancement 2 layers respectively in bits, N is the number of frames in GOP, where $\lambda_{base}, \lambda_{en1}$, and λ_{en2} are the arrival rate of base, enhancement 1 and enhancement 2 layers respectively and μ is the available bandwidth.

Figure 6.2 shows the cell loss ratio for non-layered static bandwidth allocation and multilayer dynamic bandwidth allocation for *Star Wars* trace. The larger loss is due to both high correlation and heavy tail distribution of the input video stream. We have conducted the simulation under assumption of high contention and limited network bandwidth, to be able to compare our protocol to the traditional static bandwidth allocation schemes fairly. Therefore an excessive cell loss will occur unless a large buffer size is provided. Even if one can afford this large buffer size, the problem of delay remains for delay sensitive applications. Even our proposed protocol permanently allocates a subset of bandwidth as a CBR service for the base layer of multilayer video stream, still it has better performance in terms of cell loss than traditional static bandwidth allocation schemes that use non-layered video compression, as shown in Figure 6.2.

Figure 6.3 compares the average throughput of traditional static bandwidth allocation scheme uses non-layered encoded video with the renegotiation-based dynamic bandwidth allocation scheme.

The performance of our protocol can be examined also from the bandwidth utilisation point of view, as seen in Figure 6.4. The average bandwidth utilisation is calculated as the ratio between the instant required bandwidth to the reserved bandwidth.

$$U = \frac{1}{3}\left(\sum_i fr_i^{base} / B_{base} + \sum_i fr_i^{en1} / B_{en1} + \sum_i fr_i^{en2} / B_{en2}\right) \tag{6.2}$$

where, $B_{base}, B_{en1}, B_{en2}$ are the reserved bandwidth for base, enhancement 1 ,and enhancement 2 layer respectively.

The renegotiation-based dynamic bandwidth allocation protocol using multi-layer encoded video achieves higher bandwidth utilisation than traditional static bandwidth schemes. In our protocol, the bandwidth utilisation is mostly determined by the utilisation of the permanently allocated bandwidth to the base layer.

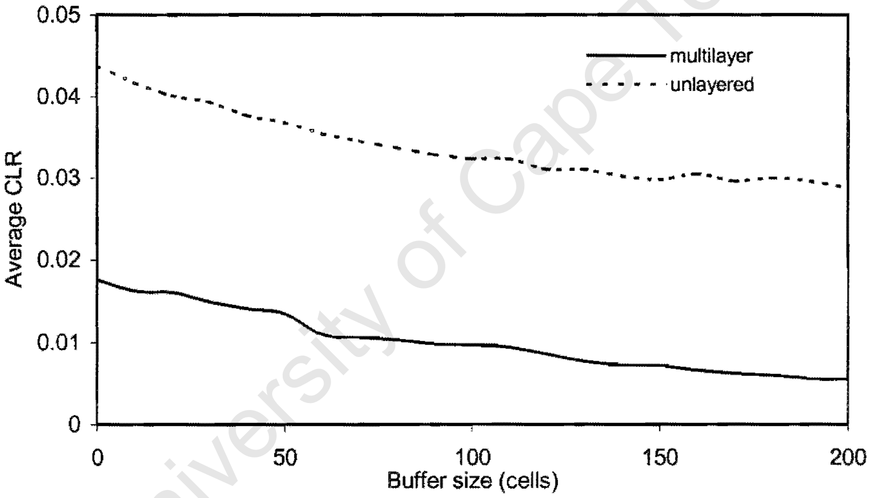


Figure 6.2: Average Cell Loss

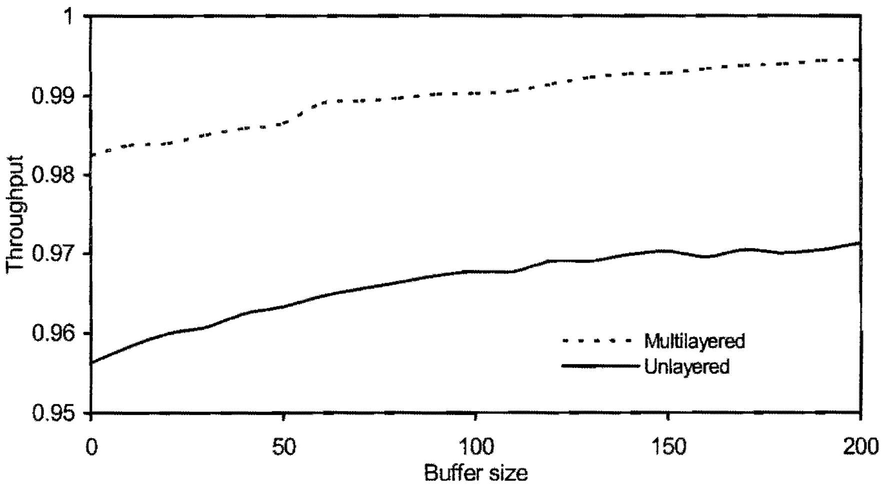


Figure 6.3: Average throughput

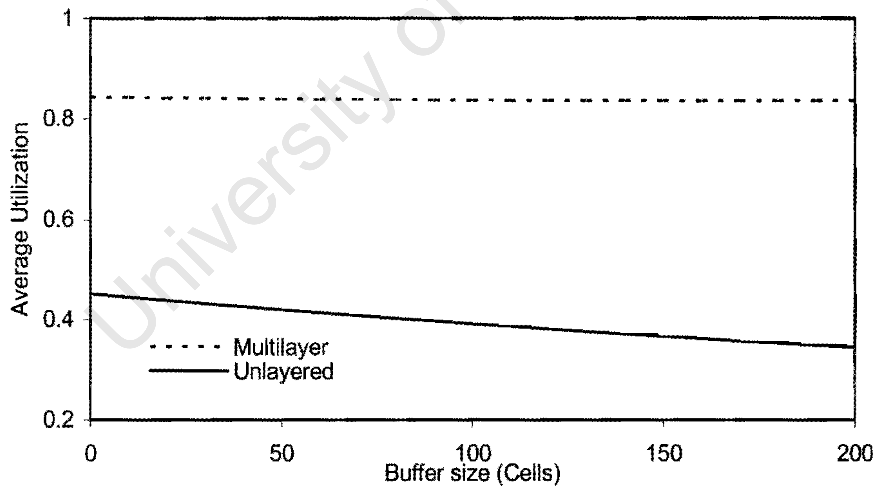


Figure 6.4: Bandwidth utilisation

6.2 Impact of Renegotiation Frequency

In the renegotiation-based dynamic bandwidth allocation protocol, the renegotiation for less bandwidth will always succeed, but renegotiation for more bandwidth might fail. In this case, an application should be prepared and able of reducing its requested QoS and bandwidth demand in order to be accommodated. This will effectively avoid blocking and congestion.

The on-line determination of bandwidth renegotiation points in VBR video generally falls into three categories: deterministic, traffic-based, and content-based. Deterministically setting the renegotiation points is the simplest method: bandwidth requests are made every frame/GOP or N frames/GOPs, where N is an empirically determined balance between requested overhead and correlation of frame or GOP bit rates. Traffic-based renegotiation, mentioned above, occurs when a stream violates a previously negotiated bandwidth request, or when utilisation drops below some level. In this work we consider the first category for renegotiation point determination, as the second approach is too computationally intensive to be used specially in real-time transmission [169].

There is a tradeoff between the renegotiation frequency and network utilisation. The more frequent renegotiations for the allocated bandwidth during the connection lifetime, the higher network utilisation, but at the expense of signalling overhead. One possible way to reduce signalling overhead is reducing the renegotiation frequency. Nevertheless renegotiation is only feasible in time scales of several hundred milliseconds to avoid excessive signalling overhead [172].

We use RM cells within a sender equipment loop to trigger the renegotiation process and to provide the renegotiation unit in Figure 5.4 with the predicted bandwidth. The bandwidth renegotiation interval time is controlled by how frequently the predictor sends the RM cells to the renegotiation control unit. In other words, if an RM cell is sent every frame, the renegotiation point interval becomes equivalent to the frame transmission time, which is too short and is not feasible for some applications [173]. However, instead of renegotiating on per-frame basis, we consider renegotiating the allocated bandwidth every group of pictures (GOP) or every couple of group of pictures (N-GOP) transmission time. Hence, in the case of renegotiation every GOP, only one predictor for each enhanced layer is required. Thus an RM cell will be sent every group of pictures GOP or N-GOP. The renegotiation process duration time has to be considered as well, as it cannot be neglected since it is limited by the protocol processing time at each node. It should be noticed also that during the renegotiation process, a switch controller does not need to re-compute routing, allocate a VCI, or inquire housekeeping records.

In this section, we present the results of simulation experiments conducted to evaluate the impact of renegotiation frequency on network resources utilisation, and to determine the upper bound on the sustainable frequency of renegotiation in ATM network. The renegotiation frequency is an important issue in adopting the renegotiation-based dynamic bandwidth allocation protocol, because it allows us to determine the time scales over which dynamic renegotiation of allocated bandwidth should take place irrespective of the time scales of the video stream. Thus, the optimum renegotiation frequency can be used to design a source traffic predictor.

Figure 6.5 and Figure 6.6 show the results of a simulation experiments conducted to determine the impact of the renegotiation frequency on bandwidth utilisation for renegotiation-based dynamic bandwidth allocation. Non-layered and multilayer video streams are used in this experiment. Our results show that with more frequent renegotiation, a higher bandwidth utilisation can be achieved. Also, Figure 6.5 and Figure 6.6 show that the impact of the renegotiation frequency on *Star Wars* traces is higher than the *News* traces. This is because the *Star Wars* movie is highly bursty compared to *News* traces and has dramatic variations in bandwidth needed over different time intervals. This results in low bandwidth utilisation unless the renegotiation process is triggered more often.

Figure 6.7 and Figure 6.8 present the impact of renegotiation frequency on cell loss ratio. They compare cell loss ratio of the renegotiation-based dynamic bandwidth allocation protocol when using non-layered encoded video and multilayer encoded video for *Star Wars* and *News* traces. The larger loss is due to both high correlation and heavy tail distribution of the input video stream. Therefore an excessive cell loss will occur unless a large buffer size is provided. Even if one can afford this large buffer size, the problem of delay remains for delay sensitive applications.

Thus, our results show that a more frequent renegotiation reduces the cell loss ratio significantly.

Figure 6.9 shows that, the renegotiation-based dynamic bandwidth allocation protocol using a multilayered encoded video stream can tolerate more loses in its higher enhancement layers than non-layered encoded video streams, thus, that will improve the overall network good-put significantly.

Table 6.1 and Figure 6.10 clearly show the effect of the renegotiation frequency on the received perceptual video quality in terms of Signal to Noise Ratio (SNR). Table 6.1 shows the percentage of the received video's GOPs received with full (i.e. enhancement 2, and 1 and base layers), enhancement 1 (i.e. enhancement 1 and base layers only) and base layer (basic quality) quality for multilayer encoding, also shows the percentage of video GOPs received with full quality and those which are blocked for non-layered encoded video. However, the multilayer video compression performs better in video quality delivery compared to non-layered video compression. This is because of the multilayer structure of the video compression, which allows video playback with receiving only the base layer data, and of course as more layers are received, a higher quality of video will be delivered.

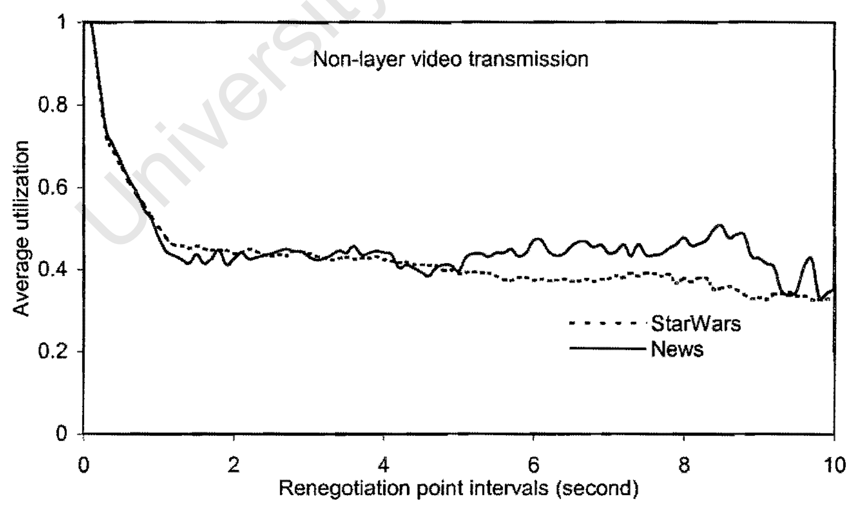


Figure 6.5: Bandwidth utilisation for non-layered video compression

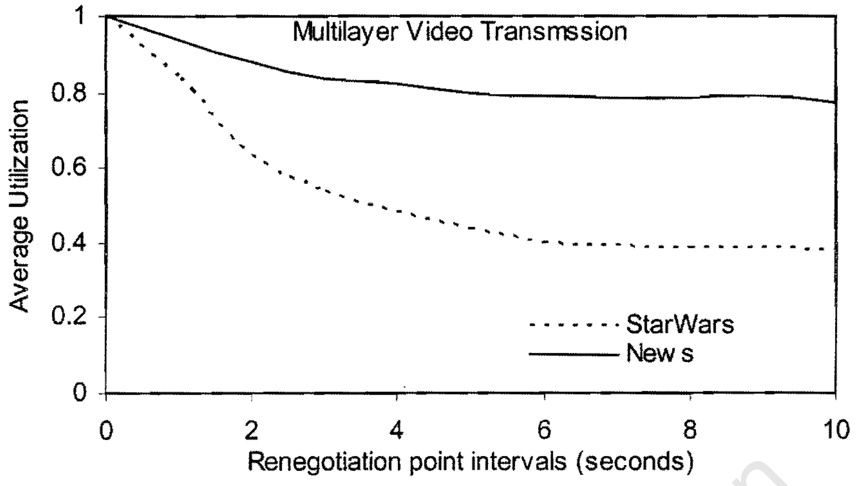


Figure 6.6: Bandwidth utilisation for multilayer video compression

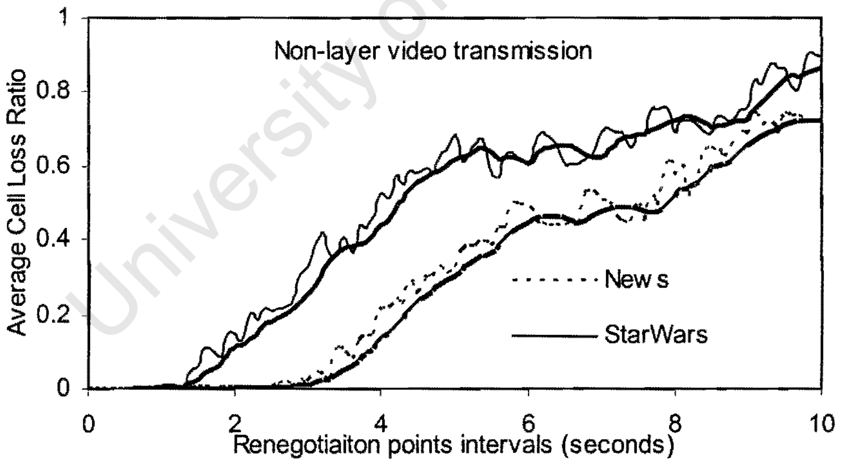


Figure 6.7: Average Cell Loss for non-layered video compression

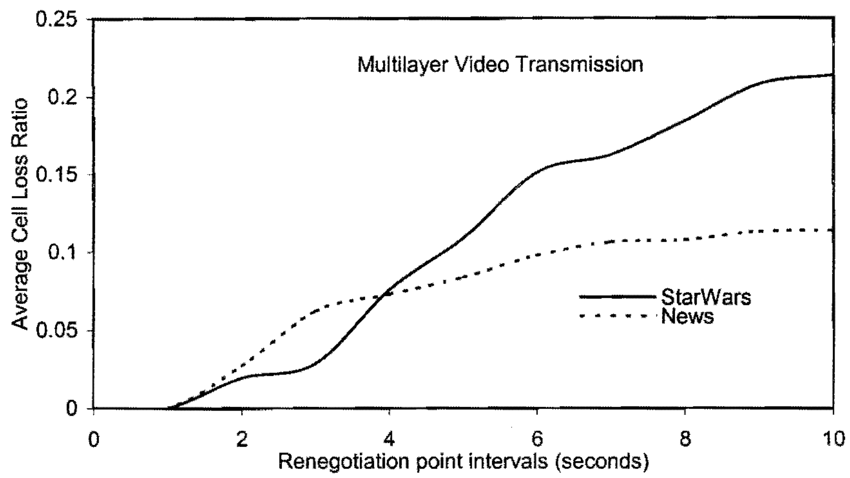


Figure 6.8: Average Cell Loss for multilayered video compression

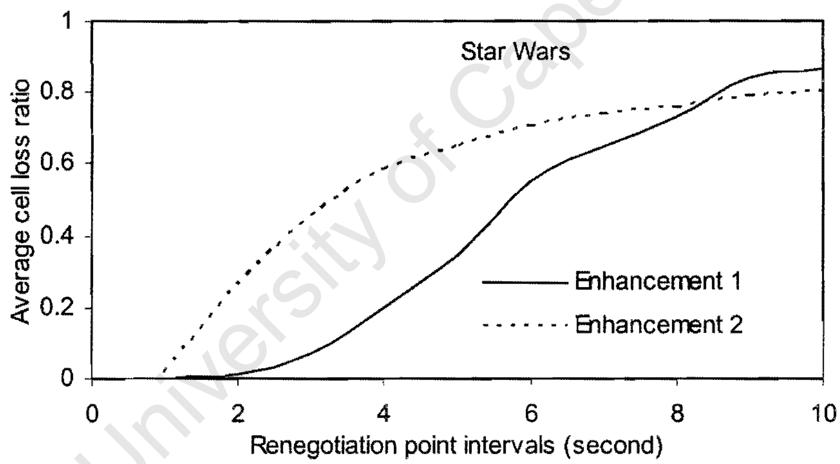


Figure 6.9: Average Cell Loss tolerance for multilayer video

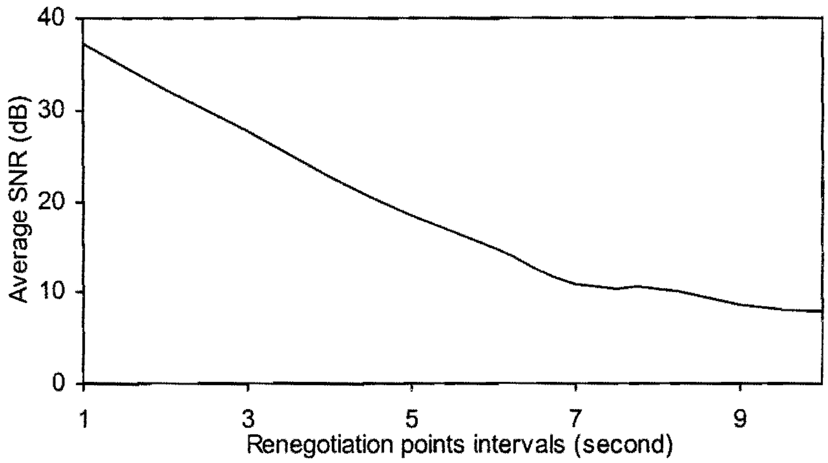


Figure 6.10: Effect of renegotiation intervals on perceptual video quality

Table 6.1: Impact of renegotiation frequency on perceptual video quality

Rengo. Intervals (Second)	Non-layered		Multilayer		
	%Poor quality(Blocked)	% Full quality	%Basic quality	%En1 quality	% Full quality
1	1.722	98.278	0	1.056	98.944
2	37.519	62.481	0.037	25.63	74.333
3	77.222	22.778	0.944	66.889	32.167
4	90.593	9.407	3.259	84.148	12.593
5	92.13	5.093	8.889	80.185	8.148
6	97.556	2.444	21.667	74.556	3.778
7	95.667	1.556	38.889	56.259	2.074
8	95.111	1.185	53.185	41.63	1.481
9	99.167	0.833	66.5	32.5	1
10	91.852	0.741	70.185	21.481	0.926

6.3 Conclusions

In this chapter we have evaluated the performance of the renegotiation-based dynamic bandwidth allocation protocol. According to our simulation results, we found that our proposed protocol achieves better performance results than the traditional static bandwidth allocation scheme (i.e. a statistical bandwidth amount is permanently allocated at call set up phase) using an non-layered video compression model that requires an accurate description of traffic parameters. This is under high contention and limited network resources.

The important feature of our protocol is that the adaptability in the allocation of bandwidth is achieved by allowing a flexible renegotiation of traffic parameters based on prediction, which does not require prior knowledge of the video correlation structure, nor does it assume it is stationary, so they can be used for real-time applications. Given a time-varying available bandwidth on the transmission path due to congestion control, the traffic source should be able to maximize the perceived quality of the delivered stream up to the level that the available network bandwidth will permit while preventing frequent changes in quality. This is the essence of QoS renegotiation. Significant improvement in bandwidth utilisation can be achieved using the renegotiation-based dynamic bandwidth allocation protocol with multilayer video compression, provided the renegotiation point intervals are selected carefully based on video traffic characteristics, such as burstiness and bandwidth required for the successive video scenes. In next chapter, we will propose another dynamic bandwidth allocation approach, which does not limit the re-allocation of bandwidth on the original path that was established during call setup phase, but instead seeks the required extra resources using multiple paths in parallel.

Chapter 7

Multipath Routing-Based Dynamic Bandwidth Allocation

In ATM, many applications such as distributed computing, multimedia, high-speed file transfer and image retrieval, exhibit high peak bit rate and long burst lengths. To support such applications, a certain number of cells are allocated on demand, which means that bursty traffic could be statistically multiplexed. This statistical multiplexing can lead to more efficient utilisation of network resources, but also can lead to severe congestion, unless a proper congestion control policy is enforced. The ATM network must be able to provide connections with QoS guarantees. Two key issues related to QoS guarantee must be addressed in order to support communications with QoS guarantees; QoS routing; which identifies paths that meet the QoS requirement and selects the one that leads to high overall resource efficiency, and bandwidth allocation; which allocates the bandwidth along the route.

When the traffic load is light, the network resources are readily available. The QoS routing is less important or critical in searching for a feasible alternative path/route. However, when the network load is heavy and at the same time bursty, efficient protocols for finding feasible routes are critical. Multipath routing can be used to increase the probability of accepting a connection under resource contention. There are two approaches for implementing multipath routing. The first one searches multipaths for a feasible route sequentially. ATM Forum, Private-Network-Network-Interface (PNNI) [195] uses a crankback based routing protocol to search for a feasible route sequentially. When the selected route does not have the required resources, the routing process is cranked back and re-searches on an alternative path. This routing protocol works well for network dynamics, but it is known that PNNI is a time consuming protocol. The multipath routing approach based on parallel searching was proposed to overcome the problem of searching for an alternative route to meet the QoS requirements [30] [192][193].

Although QoS routing and bandwidth allocation are two closely related network components [194], most existing traditional QoS routing schemes tend to decouple routing issues from the bandwidth reservation issues [193][198][199][200][201] [202]. These two issues were separated into two steps. First, route is selected, and is setup and the bandwidth is allocated along the route. Separating routing and bandwidth allocation simplifies the protocol design. However, for such bursty traffic, bandwidth availability may change rapidly and route information may become outdated. In such an environment, a route that was computed at the call setup phase (call level) may fail to accommodate the incoming bursts unless a peak bandwidth has been allocated. Combining the routing and bandwidth allocation was suggested as a solution to overcome this problem [30][192][193] [194][195][203]. In the next section we shall discuss different bandwidth allocation methodologies that are being used for high-speed networks.

In Chapter 5 the dynamic bandwidth allocation was accomplished by seeking the required extra transmission bandwidth on the same transmission path, which was established during call setup phase. The protocol was based on renegotiation of the traffic parameters during connection lifetime. There was no routing involved in the renegotiation process. Thus there would be a chance that one would fail to find the required extra bandwidth on the original path. In this chapter, we chose another approach to achieve our goal (i.e. the efficient utilisation of bandwidth). The protocol seeks the required bandwidth on alternative routes/paths using a multipath routing strategy. Therefore, we propose a multipath routing-based dynamic bandwidth allocation that combines bandwidth allocation and multipath routing. The multipath routing is invoked at burst level, when the original path cannot accommodate the incoming burst. Reservation cells are sent along multiple paths in parallel and reserve the bandwidth along the way to the destination. If more than one path is found, the best path is selected and the bandwidth that is reserved on the other paths is released. In our approach, the bandwidth is allocated and managed at different time scales on multilevels at call, burst, and cell levels, as shown in Figure 7.2, to obtain substantial bandwidth utilisation.

The proposed protocol reduces the probability of unsuccessfully finding an alternative route to accommodate the incoming burst (blocking probability) for short and longer bursts. Because the route is determined based on imprecise global information, and because a proper reservation process is not undertaken, it is possible that the determined route may actually not be able to provide the bandwidth to accommodate the incoming burst. To overcome this possible lack of bandwidth, a *route-map* is constructed. The search for extra bandwidth is limited to the alternative routes candidates in the *route-map* only.

7.1 Burst Level Bandwidth Allocation Methodologies

To solve the network bandwidth availability problem and the traffic burstiness problems is to apply a bandwidth allocation at call level, and when changes on bandwidth availability occur during the transmission phase (burst level). A Tenet RCAP protocol carries first this idea on their real time connections [60] with the Dynamic Connection Management (DMC). The Tenet protocol allows dynamic re-routing of their connections during its transmission phase. The DMC decides if a real-time connection should be rerouted to an alternative route, called the shadow channel.

Burst level bandwidth allocation methodologies in ATM networks can be classified as a reserved or non-reserved method. For non-reserved method, the network does not guarantee any amount of bandwidth, and instead, the bandwidth that is not used by the reserved applications is made available to non-reserved applications. A typical example of non-reserved method is best effort services (UBR).

The reserved bandwidth allocation method can be further classified into call level and burst level bandwidth allocation. At the call level, the bandwidth would be reserved according to peak cell rate, minimum throughput, or equivalent traffic bandwidth. However, due to traffic burstiness characteristic a burst level bandwidth management would be more desirable [62].

The burst level bandwidth allocation, is initiated at the time of burst arrival. The approach involves a fast reservation protocols in order to control traffic burstiness.

There are number of Fast Reservation Protocols (FRP) that have been proposed in the literature [22][44][174][175]. The aim of these protocols is to allocate the bandwidth efficiently for bursty traffic sources in ATM networks. Traditionally, in FRP schemes, the network allocates bandwidth equivalent to the peak cell rate for each burst. But in many applications, the required resources (bandwidth) for each burst could be negotiated between source and network as long as the burst transmission delay is kept within an acceptable range [44][174].

The FRP schemes can be classified into several groups, as shown in Table 7.1. From this table, many FRP variations are possible by selecting one component from each of these categories. The first category in Table 7.1 deals with the type of reservation. In the REQ/ACK reservation method, a burst can be transmitted only after the bandwidth on its route has been reserved unequivocally through the network. The efficiency of this method depends on the propagation delay-to-burst duration ratio. On the other hand, in the on-the-fly reservation method, a source can transmit a burst right after transmitting the reservation-request cell. In this case, if any of the traversed links are not available, the burst is blocked and the blocking node sends back a negative acknowledgement (NAK) cell. If the source receives the NAK cell, it stops transmitting the burst immediately. Consequently, the bandwidth used by the burst before receiving the NAK cell is wasted, which otherwise could been used by best effort services. In the multipath allocation scheme, multiple reservations are carried out in a parallel basis. The source broadcasts REQUEST cells over all routes candidates. The destination then selects one available path from them. Although a bandwidth de-allocation for unused paths is necessary, this scheme allows completion of the path selection in one round trip delay. The proposed protocol in this chapter falls under this FRP category.

The second category in Table 7.1 deals with the actions taken upon receipt of a NAK. In the random-backoff scheme, a blocked burst is retransmitted at later time so burst blocking means burst loss [67]. The no retry scheme is usually used with on-the-fly reservation for connectionless data service and real-time traffic, but it has problems when applied to error-sensitive real-time traffic. In the random-backoff scheme, since a source tries to request bandwidth again after a random backoff time, the possibility of unbounded burst delay could be a problem. Hence, to avoid this problem, it would be desirable that the network arbitrates and schedules burst re-transmissions, we call this the queued scheme. The problem in the queued scheme is that the complexity of the scheme increases as the number of nodes is increased.

The third category in Table 7.1 classifies the bandwidth allocation methodologies. The static bandwidth allocation schemes can be further classified into deterministic bandwidth allocation and statistical bandwidth allocation. Deterministic bandwidth allocation scheme supports applications requiring that no cells be discarded due to buffer overflows and that no cells violate their guaranteed *end-to-end* delay bounds. This scheme requires that the network assigns each connection the peak bandwidth for each burst if available, and otherwise rejects the reservation request. The primary component of a deterministic bandwidth allocation is the traffic specification profile, which provides the network with a worst-case description of a source's traffic characteristics. While a deterministic bandwidth allocation scheme has important advantages, i.e. in terms of the strength of the guarantee itself as well as the enforceability of the traffic specification, it can have a significant limitation in terms of the achievable utilisation of the network's resources. Statistical bandwidth allocation schemes achieve a statistical multiplexing gain by exploiting stochastic properties of individual traffic streams, as well as statistical independence among streams as shown in Figure 7.1 [23]. However, strong assumptions on the stochastic properties of traffic streams are inherently difficult for the network to enforce or police. The level of burstiness is dependent on various factors such as the compression scheme used and the level of motion between frames. For example, consider a Markovian source: in real time, it is impractical to determine whether a stream is following a certain transition matrix, i.e. is close enough to its implied marginal distribution, or has the required autocorrelation structure. Consequently, if a particular application does not conform to the chosen stochastic model, no guarantees can be made. Moreover, if admitted to the network, such a stream could adversely affect the performance of other applications if it is statistically multiplexed with them. Figure 7.1 also illustrates that the burstier (higher peak-to-average cell ratio) the traffic is, the higher bandwidth demanding becomes and the more bandwidth is left unused. Therefore, the effectiveness of the statistical multiplexing of VBR sources depends on the burstiness ratio of each individual source as well as the number of VBR sources. Table 7.2 shows an example for the level of burstiness for moving pictures for several video types [26].

Table 7.1: Burst level bandwidth management in ATM Networks

Reservation Scheme	REQ/ACK reservation
	On-the-fly reservation
	Multipath reservation
NAK Response	Random backoff
	Queued
	No retry
Bandwidth Allocation	Static bandwidth allocation (deterministic and statistical)
	Dynamic bandwidth

Table 7.2: Burstiness ratio for various types of video

Video Type	**Burstiness Ratio (Peak/Average bit rate)**
Studio-quality video	1.9
Broadcast-quality TV	2.7
Video conference	3.1
Video telephone	4.4

With the dynamic bandwidth allocation method, the network is able to adjust and re-allocate the bandwidth that has already been allocated during call setup phase to accommodate the incoming active bursts. Chapter 5 has proposed a solution to control the impact of traffic burstiness by proposing a renegotiation-based dynamic bandwidth allocation protocol [57] where renegotiation processes take place between the network and the traffic source. If the network has the sufficient required bandwidth to accommodate the incoming burst then the requested bandwidth will be allocated to the burst, otherwise a graceful degradation of QoS in the case of network congestion. However, in this chapter, the multipath routing is invoked when the traffic source intends to send a burst, and the required extra bandwidth to accommodate the burst is not available on the original path that was established during call setup phase. In contrast with traditional static bandwidth allocation schemes, where the bandwidth is permanently allocated at call level (see Figure 5.1), this chapter proposes that the bandwidth is allocated dynamically in multilevels.

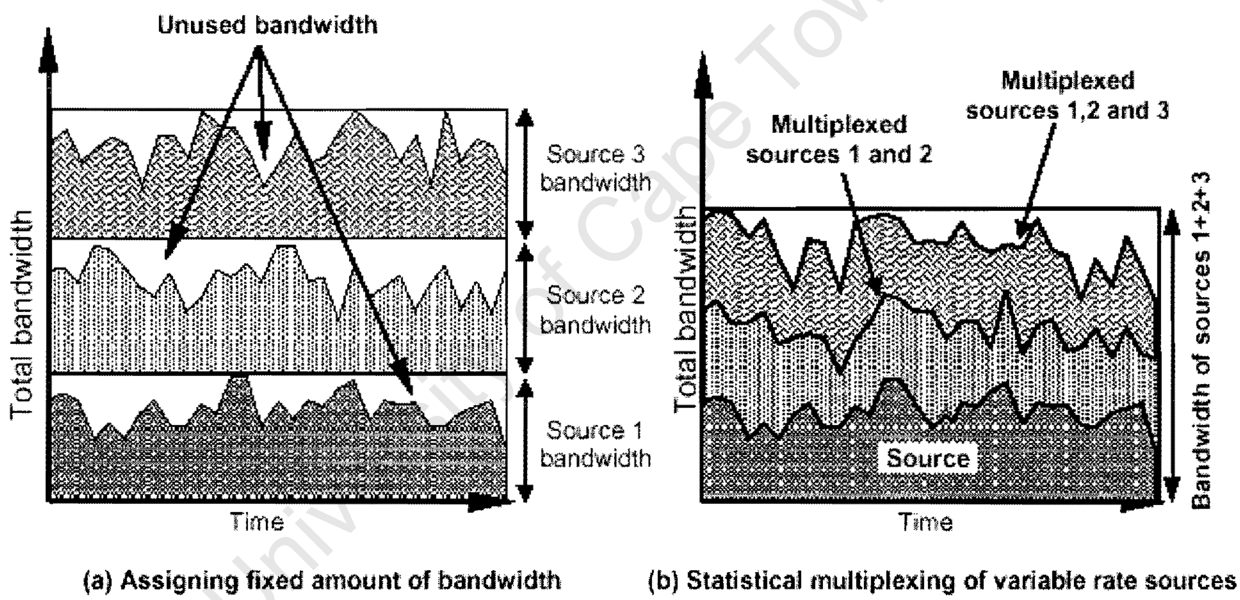


Figure 7.1: Statistical multiplexing gain

7.2 Multilevel Bandwidth Allocation and Control

The multilevel bandwidth allocation and control strategy is motivated by the fact that communication terminals often have different traffic states characterised at call, burst, and cell levels. Thus the bandwidth should be allocated and controlled at these levels in order to improve the network efficiency [28][30]. With this strategy, the bandwidth for any given ATM connection is allocated and controlled at three levels:

call, burst, and cell level, as shown in Figure 7.2. However, in a connection-oriented service, even when burst level bandwidth allocation is employed, call level bandwidth allocation is still required in advance to guarantee an acceptable burst delay and blocking probability. The cell control level bandwidth allocation policy should also be provided to ensure efficient utilisation of network resources and fairness. In the meantime it should provide isolation, i.e. the interaction between different sources types should be kept as small as possible in order to minimize interference.

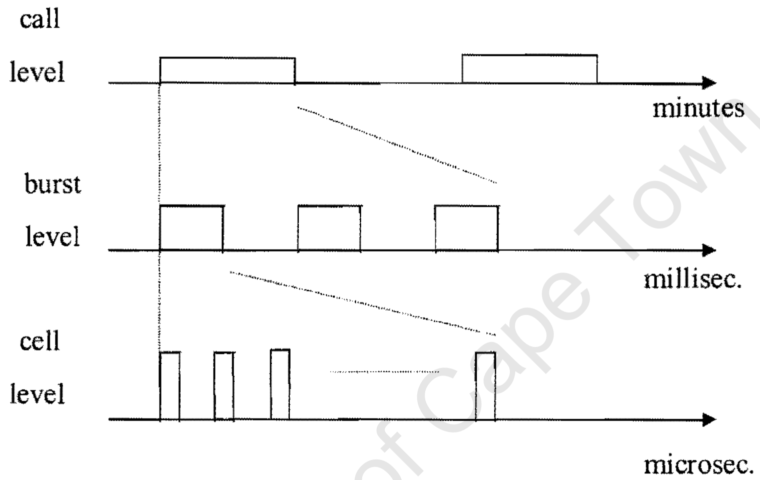


Figure 7.2: Multilevel bandwidth allocation

In this chapter, a multipath routing-based dynamic bandwidth allocation protocol is proposed. The protocol is based on multilevel bandwidth allocation strategy. Where a subset of network resources is allocated to the admitted calls at call level (call setup phase).CAC mechanism is used to control and manage the bandwidth allocation at call level. The CAC accepts connections with certain bandwidth and QoS requirements, and denies those connections, which might cause excessive burst blocking at burst level [56].

At the time of burst arrival, a subset of the network bandwidth is allocated at burst level to accommodate the incoming burst. Therefore, a Fast Reservation Multipath Protocol (FRMP) is applied as a burst level bandwidth allocation [184]. The FRMP basically searches for the burst's bandwidth requirement along the predetermined multiple routes that have been determined at call level (*route-map*) in a parallel basis. If the available bandwidth for any route is not sufficient to accommodate the peak bandwidth, the source request is denied and the burst will be discarded [56].

The burst occurrence is either determined at the source or at the network edge. When the burst occurrences is determined by the source, the source can use either predictors for real-time applications (see section 5.5), or compiling non-real time application (e.g. stored video) to determined in advance bursts occurrence times. When the network edge is responsible for determining bursts occurrence, a performance measurement-based algorithm with certain threshold at the edge determines when the source is bursting.

At the cell level, the network will allocate time slots to individual cells through an appropriate choice of scheduling mechanism. In subsequent sections, the operations needed to allocate the bandwidth according to these levels are described. In particular, we will show how traffic control at one level (e.g. call level) may reduce the blocking probability at a lower level (e.g. burst level). The operation of the proposed multipath routing-based dynamic bandwidth allocation protocol depicted according to call, burst, and cell levels is illustrated in Figure 7.4 and Figure 7.9 [28][30].

7.2.1 Bandwidth Allocation at Call Level

The aim of call level bandwidth allocation is to emulate circuit switching for fixed and high bit rate services and denies calls when facility overload may cause excessive burst blocking.

At call level (call setup phase), CAC is applied to setup a connection to carry the source's regular traffic without considering bursts bandwidth requirements. The CAC also, determines the possible alternative routes candidates between the source and the destination. These routes candidates are stored in translation table (*route-map*) of each participating ATM node.

Applications often have threshold requirements on the routes. A route with cost that is not within the threshold boundaries cannot be considered as route candidate in the *route-map*. This is to limit the search and reduce the signalling cost, such that it will include only routes whose cost is below a certain threshold. The threshold is designed such that eligible routes candidates are selected to provide the following requirement:

- Minimizing the blocking probability.
- Only the feasible route is considered with the minimal cost.
- The selected route is that with a cost close to the min-cost feasible route.

To this aim we define the *route-map*, as collection of routes candidates that link the source to destination, where the alternative routes candidates are searched. The source node based on a maintained topology map at call control level can construct the *route-map* locally. In this case, the protocol messages can carry the *route-map* structure. Another possibility is to periodically compute an incoming directed *route-map* for every destination in the network and to store it in *route-map* tables at the nodes. The use of these *route-map* tables is similar to the use of shortest path routing tables [176], only here the table can point to more than one link per destination. A third approach is to calculate the *route-map* in a distributed manner based on the local topology database that the nodes maintain.

After the call level bandwidth allocation and control phase is completed, each route candidate in the *route-map* should be associated with cost, available bandwidth, and transmission delays. While the cost of all the routes might be known before the protocol starts its burst level phase, the rapidly changing availability of bandwidth and the processing delay are unknown in advance. In addition, several connections might compete for bandwidth and even slight timing differences among the reservation messages can dictate which connections will reserve the bandwidth successfully and which will fail. Under the above constraints, we wish to find (and reserve) a path in the *route-map* between the source and the destination nodes that has a favourable cost and sufficient bandwidth to accommodate the incoming burst.

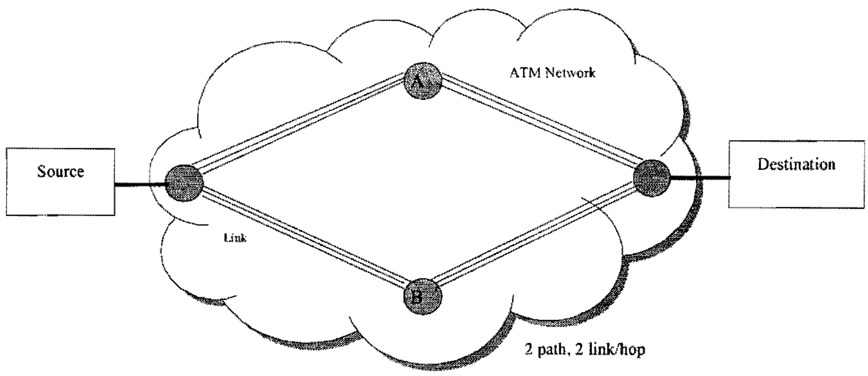


Figure 7.3: Multipath routes in ATM network

7.2.2 Bandwidth Allocation at Burst Level

After the call level bandwidth allocation phase of the multipath routing-based dynamic bandwidth allocation protocol is over, the burst level bandwidth allocation phase starts at the time of burst arrival. A Fast Multipath Routing Protocol (FMRP) is applied at burst level of the multipath routing-based dynamic bandwidth allocation protocol. The main objective of the FMRP protocol is to setup connections in the network at burst level to accommodate the incoming bursts with minimum blocking probability. At the time of a burst occurring, the protocol searches the *route-map* for extra bandwidth to accommodate the incoming burst on the *burst-by-burst* basis, when the required bandwidth can not be provided on the route that was established during call setup phase (call level). Reserving resources on multiple paths makes the routing faster and more resilient to the dynamic change of the network state and fault occurrence [185].

Two main properties that are important in the Fast Multipath Routing Protocol: the reservation of resources while the search is still progressing, and the extra bandwidth is searched in parallel along multiple routes. The first property saves searching time. The parallel search also increases the probability of finding an alternative route that has the required extra bandwidth to accommodate the incoming burst, as indicated by the results in section 7.5. In section 7.3, the FMRP procedure is explained in details.

7.2.3 Bandwidth Allocation at Cell Level

Once available route has been found to accommodate the incoming burst at burst level, individual cells belonging to a burst can be transferred with no cell loss due to the peak bandwidth reservation. Traffic enforcement has to be carried out at the network interface at cell level to monitor/enforce the users to maintain their peak bit rate by using, for example, the leaky bucket.

The cell level bandwidth allocation and control, is implemented by imposing a proper scheduling algorithm at this level. The performance of cell level bandwidth allocation depends on individual switches architecture and scheduling mechanism [177]. Most proposed ATM switches architectures in the literature [178][179] are port-oriented. They are based on the *first-in-first-out* (FIFO) principle: cells from various virtual connections leaving an output port are organized into FIFO queue and processed in the order in which they arrived. The functionality of traditional ATM switches is limited to routing cells from input ports to output ports. It is known that the operations of ATM networks is not just limited to the transport of cells from sources to their destinations; ATM networks also have to handle ATM traffic efficiently and reliably. An ATM switch is an essential element of an ATM network, and it should be designed in line with the operations of the network. We argue that ATM switches should be traffic-flow-oriented and that switches should handle each traffic flow separately to offer the following kinds of guarantees; a flow gets a specified throughput rate, the end-to-end delays of cells in a flow are bounded, and the delay jitter over a set of cells are bounded. New ATM switches are currently being developed to provide per-VC queuing [180].

7.3 Burst Level Fast Multipath Routing Protocol Description

This section describes the fast multipath routing protocol (FMRP). This protocol is a part of multilevel bandwidth allocation strategy, which was discussed in section 7.2. The FMRP is invoked in our protocol only if the bandwidth allocated at call level cannot accommodate the incoming burst. The protocol allocates the required bandwidth for each burst at burst level during a round-trip delay plus a burst transmission time to accommodate the incoming bursts. The protocol searches for the required extra bandwidth on multiple routes concurrently. Generally, searching from scratch for a route between two nodes in the entire network is inefficient in terms of signalling cost and processing time. Thus the protocol is based on a flooding-like algorithm [192][199][200] that attempts to reserve bandwidth along several possible routes on the *route-map* only. When a source sends a burst and the main connection does not have sufficient bandwidth to accommodate the burst, it searches for the best alternative route in the *route-map* that leads to the same destination and has a reasonable "cost".

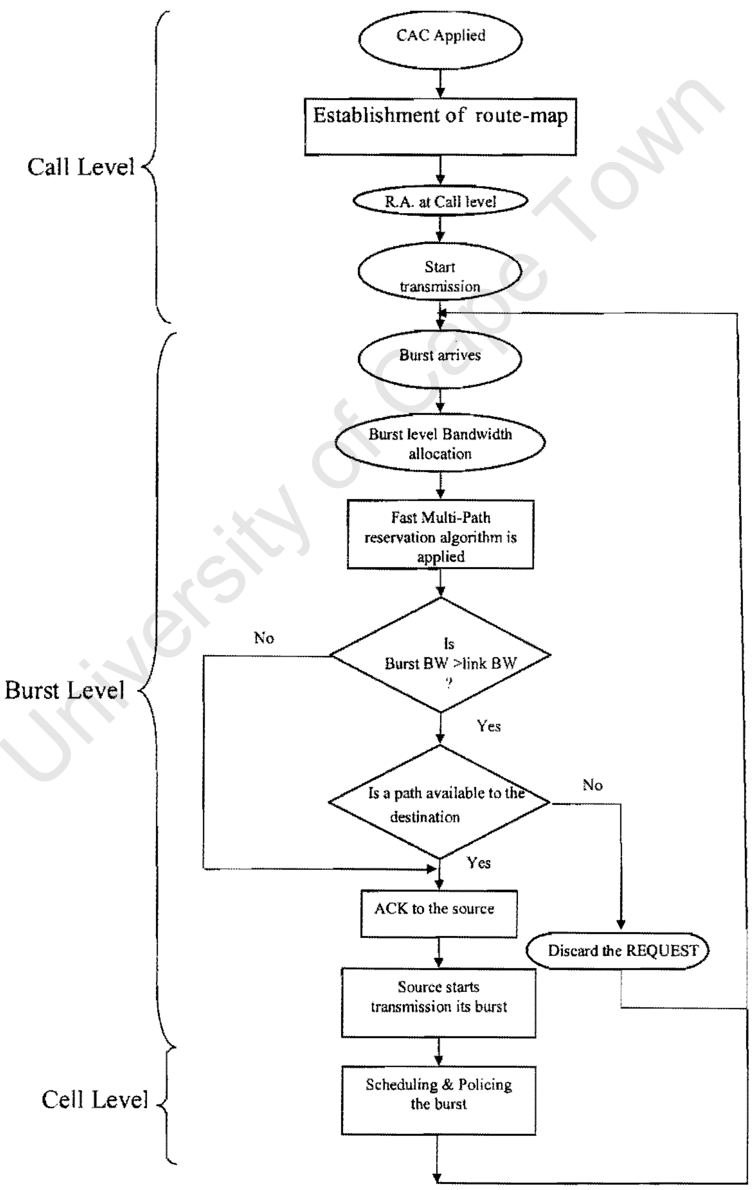


Figure 7.4: Multipath routing-based dynamic bandwidth allocation protocol

The protocol searches routes candidates in the *route-map* for feasible route. A route is called feasible if it has enough available bandwidth to accommodate the incoming burst to be considered as an alternative route. The following control cells are used in the multipath routing-based dynamic bandwidth allocation protocol.

- REQUEST cell: The information in this cell includes a specification of the bandwidth requested and QoS requirement for the burst and the total cost of the path from the source to the current node.
- REJECT cell: The information in this cell includes a negative feedback acknowledgment message back to the source. Any node along the path can send a REJECT cell if it has no outgoing link with the requested QoS and sufficient bandwidth. In this case, a REJECT cell is sent to inform the source with unavailability of bandwidth. Another scenario where a REJECT cell is sent is also when a node receives more than one REQUEST cell. In this case, all the REQUEST cells except the one with the minimal cost are answered with REJECT . A third situation for sending REJECT is when a node receives REJECT for every REQUEST it sent. When the source receives REJECT cells from its entire outgoing links, the protocol fails and the burst cannot be accommodated.
- ACCEPT cell: The information in this cell includes a positive feedback acknowledgment message back to the source.
- RELEASE cell: This cell will travel through the path and release all the resources allocated for the burst.

Before the source starts transmitting a burst, each node must maintain precise state information for all outgoing links. Since our protocol focuses on bandwidth, each node will maintain for each outgoing link the amount of used bandwidth, which is in use by other connections, and the amount of reserved bandwidth, which is the bandwidth that is reserved but is not currently used.

At the time of burst occurring, the source sends REQUEST cells in all the feasible outgoing routes candidates in the *route-map*. When an intermediate node receives a REQUST cell, it does the following: if the connection has been established for the burst, a REJECT cell will be sent since there is no meaning to reserve a bandwidth for burst already served. If the bandwidth has not been reserved for the burst (i.e. this is first REQUST the node receives), the node will check whether it has sufficient bandwidth to accommodates the incoming burst. The node will distribute REQUEST cells to all outgoing feasible routes candidates that may satisfy the QoS constraint and path-cost and marks the incoming link as master-link. This master-link is used as a control channel to carry the feedback information like positive or negative acknowledgment messages, ACCEPT or REJECT respectively, back to the source. If none of the outgoing links has sufficient bandwidth to accommodate the incoming burst, the node responds with REJECT cell. If the node has received a REQUEST cell, that is, the current REQUEST is not the only one, the node checks whether the current REQUEST can provide better QoS and path-cost for the burst. If the current REQUEST has better QoS and path-cost, the node would then forward the REQUEST, REJECT to old REQUEST, and changes the master-link marking for this node if the REQUEST cell arrives from a different in-coming link the node. Otherwise, the node will respond with REJECT cell.

The destination node answers with ACCEPT to the first REQUEST it receives, and with REJECT to all the other subsequent REQUEST cells. This ensures the fastest establishment of a connection to accommodate the incoming burst and it might come at the expense of the route optimality. The ACCEPT cell is immediately replied by every node to its master-link until it reaches the source, at which time it can start transmitting burst. A node that forwarded an ACCEPT cell will respond with REJECT to every subsequent REQUEST it receives to signal the termination of the route establishment process.

When the source receives the ACCEPT cell, it knows that a reserved route exists to the destination. Then starts sending its burst, and according to Figure 7.4, cell level bandwidth allocation phase starts doing its job as explained in section 7.2.3. However, the source does not know when all the nodes in the *route-map* terminate their part of the protocol, and more importantly, when the protocol terminates [181]. A node that is not part of the selected route can never learn that the search ended successfully. The problem with the inability to identify termination is that nodes cannot release the bandwidth allocated by the protocol. Note that this has nothing to do with the bandwidth required from the network, whose release is guaranteed. Thus, a termination mechanism is needed to terminate the protocol and release all bandwidth allocated for the burst. The mechanism is triggered by the source node when it receives ACCEPT or REJECT from all the eligible links. The source node releases the bandwidth allocated to the burst and forwards a RELEASE cells on all its outgoing links [182]. A node that receives RELEASE for the first time marks this link as the preferred link and sends RELEASE on its entire outgoing and incoming links. When the node receives RELEASE from its entire outgoing and incoming links, it sends one on the preferred link and releases all the variables of the protocol. When the source node receives RELEASE from its entire outgoing links, it knows that the protocol terminated and that all the nodes know this fact. Following a result by *Awerbuch et al.* [183], we can flood the RELEASE cells only along the links where REQUEST cells were sent.

Consider the *route-map* example illustrated in Figure 7.5 with 3 alternative routes candidates, where the numbers on the links are designate link delays, the costs of all links are equal to unity, and all routes are feasible to be considered as routes candidates. At the time of burst arrival at node A and there is no sufficient bandwidth on the original route that has been established during call level. An extra bandwidth is requested to accommodate the incoming burst destined for node F.

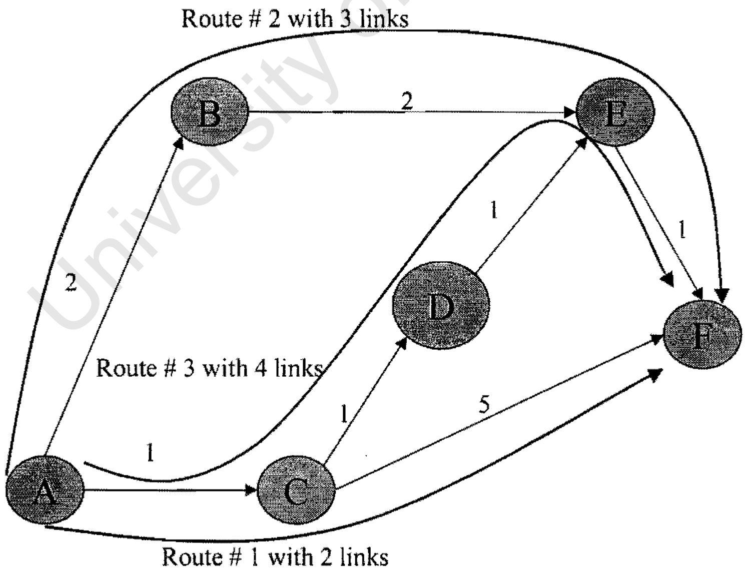


Figure 7.5: *Route-map* example

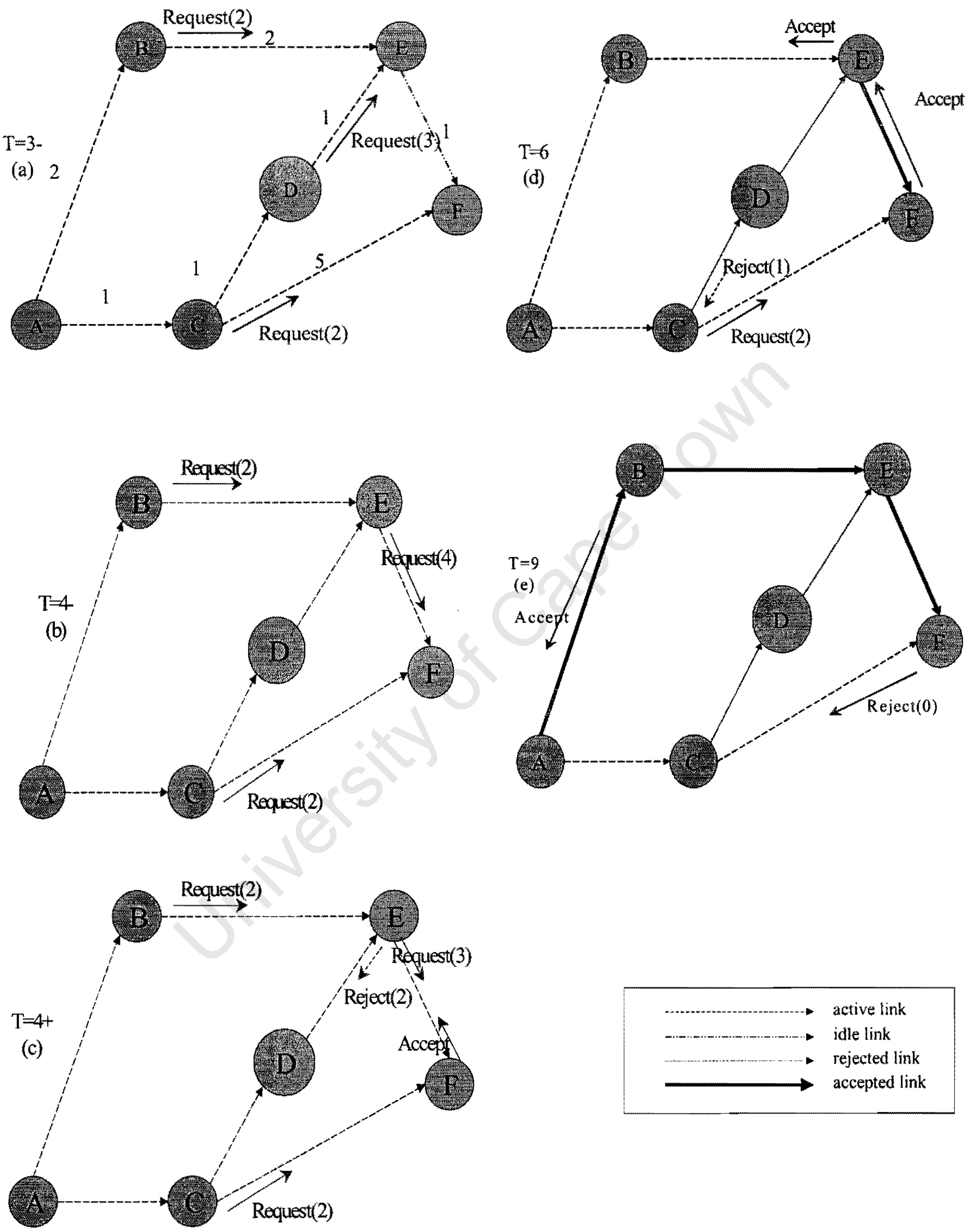


Figure 7.6: An example of Fast Multipath Routing Protocol execution

7.4 Network Model and Assumptions

In this analysis we have assumed that the network topology is modelled as undirected graph represented by $H(U,M)$, where U represent switches, routers and hosts and M represents unidirectional link connecting the nodes. Unidirectional connections are established on a *per-need basis* as a concatenation of one or more links. Although links are unidirectional for data traffic we assume that control traffic is bi-directional. This is the usual practice in ATM networks since the links are not physical entities but are themselves established logically atop a physical network. However, control channels are usually established together with the transmission link to enable its two end-points to converse. For example, in ATM, the links are unidirectional virtual paths (VPs) and each has a control virtual channel (VC) in each direction. The bandwidth for any given ATM connection is allocated and managed at three levels; calls level, burst level, and cell level [28][30].

At call level we assume that the predetermined paths are constructed and tabulated in *route-map* tables. Each link in the *route-map* is associated with a cost, available bandwidth, and processing/transmission delays. The network topology and the links costs are assumed to be almost static, i.e. they change slowly enough for the network control to handle the changes. This assumption fits well with the realistic reliability of the ATM network components. However, the availability of bandwidth is assumed to be dynamically fluctuating, as connections are brought up and torn down. It is impossible for the network control to follow the rapid changes on these link parameters, thus only time averages or estimates are available for nodes that are not in the link locale.

In this research we have assumed that each source-destination pair of nodes are connected by n disjoint routes, as shown in Figure 7.3, and each route can support a single burst at a time. The proposed multipath routing-based dynamic bandwidth allocation protocol seizes simultaneous bandwidth along source-destination paths. Therefore, the bandwidth is assumed to be cross-correlated and the degree of correlation is dependent on network topology, routing technique, connection types, and so on. It is known that as the number of nodes increases, the correlation between them decreases. Thus our assumption in this analysis is that the network nodes are weakly coupled. Therefore, the aggregate new burst arrivals from ON/OFF sources can be reasonably approximated by a Poisson process with intensity λ. We assume also that the burst has no pre-knowledge of routes bandwidth availability. Thus, the burst selects randomly r $(1 \le r \le n)$ of the routes and searchers for the required bandwidth on them, where n is the number of routes candidates in the *route-map* between the source and destination. The searching process is carried out in parallel on the *burst-by-burst basis*, as illustrated in Figure 7.8, instead of sequence searching, as in Figure 7.7. If more than one route is found, only a single route is used and the others are released.

The overall period of the route search and the burst duration is assumed to be exponentially distributed with mean $1/\mu$, as shown Figure 7.9. The bandwidth holding time period until an unused route is released is exponentially distributed also with mean $1/\alpha$ $(1/\alpha \le 1/\mu)$. When the incoming burst duration is shorter than the reservation process duration, it is assumed that $\alpha = \mu$. Thus, if more than one alternative route found, all the routes appear to be used, while the destination is using only one of them. This happens, for example, when a short burst is sent proceeded by reservation request that tries to reserve sufficient resources *on-the-fly* [29][44] [174].

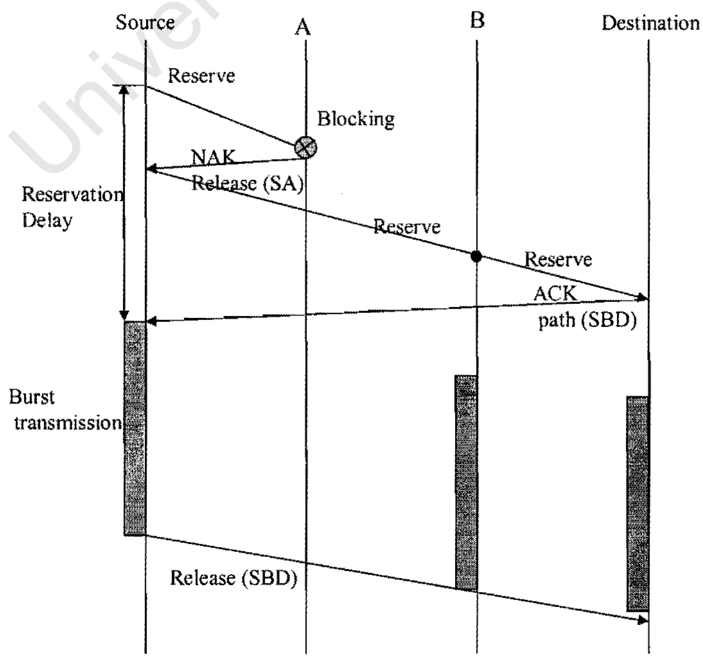


Figure 7.7: Single-path reservation scheme

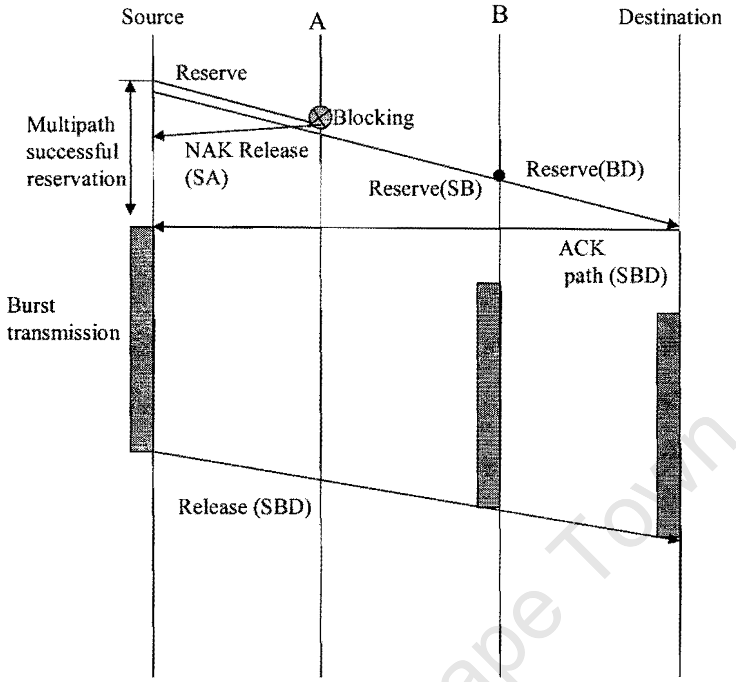


Figure 7.8: Multipath reservation protocol

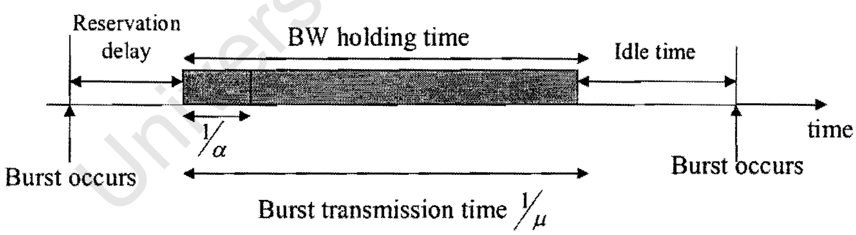


Figure 7.9: Bandwidth holding time

7.5 Performance Analysis

In multipath routing-based dynamic bandwidth allocation protocol, a burst that arrives at the network tries to capture bandwidth simultaneously for each link on its route. Only bursts that succeeded in the entire reservation contribute to the useful throughput. The network has finite number of links. The links are modelled by

G/M/N/N queues and thus, the total number of system states is finite. The network that uses multipath routing strategy can be modelled by a continuous-time Markov chain with $n(n+3)/2$ states [192], where n is number of routes candidates on the *route-map*. The infinitesimal transition rates matrix Q, uniquely describes the states. Each state is represented by (σ,ω) where σ the number of routes that are used for burst transmission, and ω is the number of alternative routes that were found and not used for burst transmission. The elements of infinitesimal transition rates from the state (σ,ω) to (σ',ω') are:

$$q_{(\sigma,\omega),(\sigma-1,\omega)} = \sigma\alpha \tag{7.1}$$

$$q_{(\sigma,\omega),(\sigma,\omega-1)} = \omega\mu \tag{7.2}$$

$$q_{(\sigma,\omega),(\sigma+i,\omega+1)} = \delta(i,\sigma+\omega)\lambda \qquad (0 \le i \le \min\{r-1, n-(\omega+\sigma+1)\}) \tag{7.3}$$

where,

$$\delta(i,m) = \binom{n-m}{i+1}\binom{m}{r-(i+1)} \bigg/ \binom{n}{r} \tag{7.4}$$

Our aim is to find the probability of unsuccessfully finding an alternative route in the *route-map* to accommodate the incoming burst. However, the burst blocking probability and reservation delay are dependant on the location of the burst blocking in the network.

The network steady state probabilities; $\pi_{\sigma,\omega}$ can be found by solving system equilibrium equations (Appendix A). The steady state probabilities are given by:

$$\vec{\pi}Q = 0 \tag{7.5}$$

where Q is infinitesimal transition rates matrix, which is constructed from the transition rates of Equations (7.1-7.3), $\vec{\pi}$ is the vector of steady state probabilities.

The equations at Appendix A are solved together with probability conservation relation numerically using Gaussian eliminations method to find the transition probabilities states $\pi_{\sigma,\omega}$,

$$\sum\nolimits_{\sigma,\omega} \pi_{\sigma,\omega} = 1. \tag{7.6}$$

$$A\pi_{\sigma,\omega} = B \tag{7 .7}$$

$$A = \begin{pmatrix} a_{0,0,,0,0} & a_{0,0,,0,1} & a_{0,0,,0,2} & a_{0,0,,0,3} & \cdots\cdots & a_{0,0,,0,n} \\ a_{0,1,,0,0} & a_{0,1,,0,1} & a_{0,1,,0,2} & a_{0,1,,0,3} & \cdots\cdots & a_{0,1,,0,n} \\ a_{0,2,,0,0} & a_{0,2,,0,1} & a_{0,2,,0,2} & a_{0,2,,0,3} & \cdots\cdots & a_{0,2,,0,n} \\ a_{0,3,0,0} & a_{0,3,0,1} & a_{0,3,0,2} & a_{0,3,0,3} & \cdots\cdots & a_{0,3,0,n} \\ \vdots & \vdots & \vdots & \vdots & & \vdots \\ a_{0,n,0,0} & a_{0,n,0,1} & a_{0,n,0,2} & a_{0,n,0,3} & \cdots\cdots & a_{0,n,0,n} \end{pmatrix} \tag{7.8}$$

$$B = \begin{bmatrix} 0 \\ 0 \\ 0 \\ 0 \\ \vdots \\ 1 \end{bmatrix} \tag{7.9}$$

where,

$$a_{(\sigma,\omega),(i,j)} = \begin{cases} -q_{(i,j),(\sigma,\omega)} & i = \sigma \ \& \ j = \omega \\ q_{(i,j),(\sigma,\omega)} & elsewhere \end{cases} \tag{7.10}$$

7.5.1 Multipath Burst Reservation Blocking Probability

This section uses the analytical approach to derive the probability of unsuccessfully finding an alternative route to accommodate the incoming burst (blocking probability) P_{Bloc} for multipath routing-based dynamic bandwidth allocation protocol. The P_{Bloc}, the blocking probability is given by the ratio between the rate of bursts that have been served, λ_{out}, and the rate of incoming bursts λ_{in}.

$$P_{Bloc} = \textit{1- Rate of bursts have been served /Rate of incoming burst}$$

$$P_{Bloc} = 1 - \frac{\overline{N}\mu}{\lambda} = 1 - \frac{1}{\rho}\sum_{\sigma=0}^{n-1}\sum_{\omega=0}^{n} \omega \cdot \pi_{\sigma,\omega} \tag{7.11}$$

where $\overline{N}$ is the average number of served connections in the system (*little's law*), and $\rho = \lambda / \mu$.

To obtain P_{Bloc}, the system steady-state probabilities $\pi_{\sigma,\omega}$ is obtained from solving the matrix Equation (7.7).

7.5.2 Single–Path Burst Reservation

Another possibility is to work around congestion problem, and provide the bursty traffic sources with the required extra bandwidth, by applying the burst level dynamic bandwidth allocation using sequential single path searching strategy, which has been illustrated in Figure 7.7. In this strategy, only one path/route at the time is considered for routing at burst level, but upon failure, along this route/path, an additional number of crank-backs are allowed to try to reserve resources on other alternative routes sequentially. By allowing crank-back, the actual arrival rate λ_{ef}, that the system observes is higher than λ, the arrival rate of requests from the outside.

This analysis assumes that the time period between consecutive crank back attempts (reattempts) is exponentially distributed and the combined arrival process of new incoming request and repeating requests is Poisson. It also assumes that upon failure, the next route is selected randomly for reservation [191].

In the case of single-path reservation $r = 1$, and $\sigma = 0$, thus the system can be modelled by $n+1$ state birth-death process with transition rates, as shown in Figure 7.10.

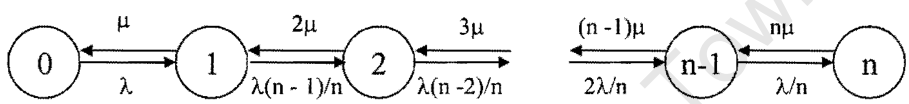


Figure 7.10: Birth-death process with transition rates

$$\lambda_\omega = \frac{n-\omega}{n}\lambda \qquad \mu_\omega = \omega\mu \tag{7.12}$$

The average number of active connections is given by

$$\overline{N} = \sum_{\omega=1}^{n} \omega \cdot \pi_\omega = \frac{\rho n}{n+\rho} \tag{7.13}$$

where $\rho = \lambda / \mu$.

Thus;

$$P_{Bloc} = 1 - \frac{\lambda_{out}}{\lambda_{in}} = 1 - \frac{\overline{N}\mu}{\lambda} = \frac{\rho}{n+\rho} \tag{7.14}$$

Let us consider the case when single-path reservation when an infinite crank-back is allowed until the success is achieved $(a=\infty)$. If the arrival rate λ is larger than $n\mu$, this system is unstable.

The effective arrival rate is given by,

$$\lambda_{ef} = \lambda \sum (1-P_{Bloc}) \cdot i \cdot (P_{Bloc})^{i-1} = \lambda/(1-P_{Bloc}) \tag{7.15}$$

Using Equation (7.14), we can compute the burst blocking probability per attempt as follows,

$$P_{Bloc} = \frac{\rho}{n} \tag{7.16}$$

When we allow the crank back once in the single path reservation (i.e. two-reservation trails) $(a=2)$, the effective arrival rate is,

$$\lambda_{ef} = \lambda + \lambda P_{Bloc} \tag{7.17}$$

Substituting in Equation (7.14) yields the equation,

$$\rho(1-P_{Bloc})^2 - (n+2\rho)(1-P_{Bloc}) + n = 0 \tag{7.18}$$

$$P_{Bloc} = \sqrt{\left(\frac{n}{2\rho}\right)^2 + 1} - \frac{n}{2\rho} \tag{7.19}$$

If the number of allowed crank back attempts is two (i.e. three reservation trails) $(a=3)$, the effective arrival rate is given by:

$$\lambda_{ef} = \lambda\left(1 + P_{Bloc} + P_{Bloc}^{2}\right) \tag{7.20}$$

Substituting λ_{ef} in (7.14) and solve the equation for blocking probability,

$$P_{Bloc} = \frac{0.873n}{h\sqrt{\rho}} + \frac{h}{2.62\sqrt{\rho}} \tag{7.21}$$

where,

$$h = \sqrt[3]{-9\rho^{3/2} + \sqrt{3}\sqrt{4n^3 + 27\rho^3}}$$

7.5.3 Single-Path Reservation Delay

In traditional fast reservation protocols, the protocol tries to reserve the burst required peak bandwidth, on a *path-by-path* basis at burst level sequentially [46][47][49]. We assume that the reservation delay of single path reservation without allowing crank-back attempt is exponentially distributed with mean D. We also assume the burst reservation delay includes the propagation, queuing, and switching delay. If the scheme fails to reserve the required bandwidth in a certain path then it cranks-back and reattempts again after a back-off period on alternative route on the *path-by-path basis*. This period is assumed as exponentially distributed with mean τ_b.

The single-path reservation delay D_{reatt} with one $(a = 2)$, two $(a = 3)$ and infinite $(a = \infty)$ crank back attempts (i.e. successive trails until success) is also compared with a single path reservation without permitting the crank back is D.

$$D_{reatt} = \frac{1}{1 - p_{Bloc}^{a}} \sum_{i=1}^{a} i \cdot (\tau_b + D) P_{Bloc}^{i-1} (1 - P_{Block}) \tag{7.22}$$

where P_{Block}, is computed for one and two crank back attempts using Equations (7.19) and (7.21) respectively. The single path reservation delay of Equation (7.22) is plotted in Figure 7.20.

7.5.4 Multipath Reservation Delay

The reservation delay of the multipath routing-based dynamic bandwidth allocation protocol D_m is calculated and compared with the reservation delay of the traditional single-path reservation schemes. The average reservation delay of first success when r multiple reservations done in parallel is D/r, as it is a competition between r exponential processes. Thus the overall average reservation delay to successfully reserve the required burst bandwidth is bounded below by:

$$D_m = \frac{1}{1-(1-(1-P_{Bloc})/r)^r}\left[\frac{D}{r}(1-P_{Bloc})\frac{(1-P_{Bloc})}{r}\right.$$
$$+\frac{D}{r-1}\left(1-\frac{1-P_{Bloc}}{r}\right)\frac{(1-P_{Bloc})}{r}+\ldots. \tag{7.18}$$
$$\left.+D\left(1-\frac{1-P_{Bloc}}{r}\right)^{r-1}\frac{(1-P_{Bloc})}{r}\right]$$

where, P_{Bloc} is calculated by Equation (7.6) and r is the number of routes candidates considered for routing at burst level.

7.6 Numerical Results

This section reports the performance results of the multipath routing-based dynamic bandwidth allocation protocol for short and long bursts in terms of average reservation delay, burst blocking probability, and throughput characteristics. The burst blocking probability represents the probability of unsuccessfully finding an alternative route to accommodate the incoming burst when the original route that established at call level cannot accommodates the incoming burst. The influence of traffic load and the number of alternative paths from source to destination on the performance of the multipath routing-based protocol is presented. The analysis is conducted under the assumption that the network traffic is heavy loaded; in other words the burst bandwidth requirements is more than the bandwidth already allocated at call level. Figures 7.11-7.18 depict the probability of unsuccessfully finding an alternative route to accommodate burst, and throughput for short and long bursts. The analysis has been evaluated for a network that has 4 and 8 alternative routes candidates used by the multipath routing protocol. For all the calculated parameters, single path reservation $r = 1$, where reservation for a connection is attempted along a single route, always achieves the highest burst blocking probability when compared with multipath alternative $r > 1$, where reservation is attempted along several routes in parallel.

The performance of single path reservation without allowing crank back and with allowing 2, 3, and infinite crank backs to reserve the required bandwidth to accommodate the incoming bursts have been evaluated in terms of burst blocking probability as shown in Figure 7.19. The search for an alternative route in traditional single-path reservation like ATM PNNI routing protocol is carried out in sequential fashion, which means the reservation process delay will be longer than the bursts are able to tolerate. The average reservation delay for traditional single path reservation and multipath reservation are presented in Figure 7.20 and Figure 7.21.

Figure 7.21 also quantifies the effect of parallelism for multipath routing-based dynamic bandwidth allocation protocol on the reservation delay. The average success reservation time in Y-axis has been normalized with single path reservation delay without reattempt D. As expected, when the number of routes candidates for routing at burst level increase, the reservation delay decreases dramatically.

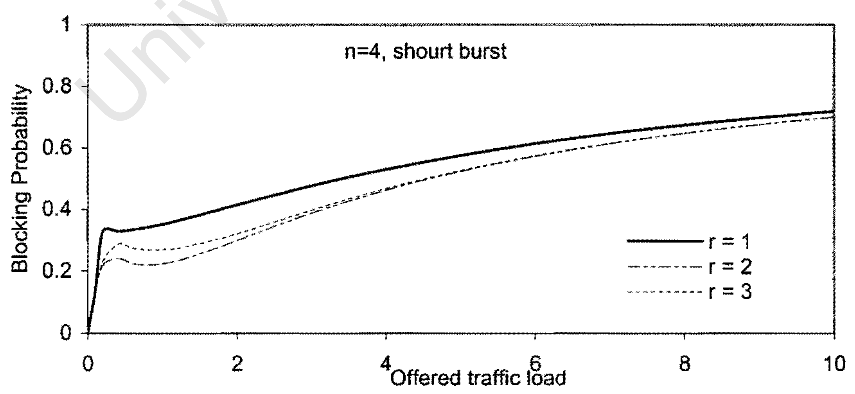


Figure 7.11: Multipath short bursts blocking probability with 4 alternative routes

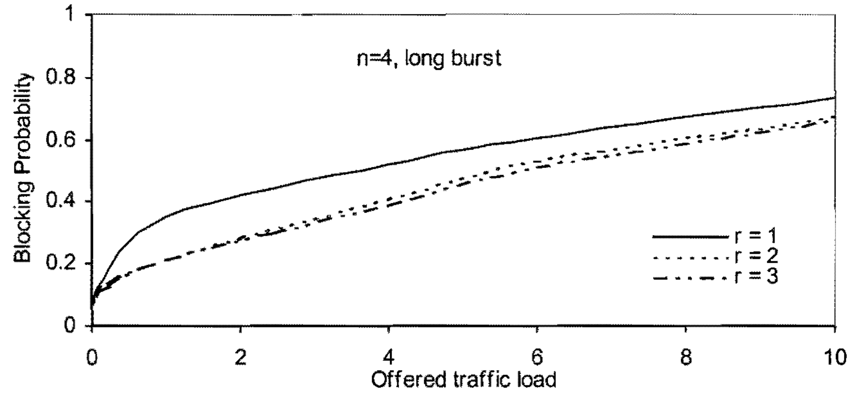


Figure 7.12: Multipath long bursts blocking probability with 4 alternative routes

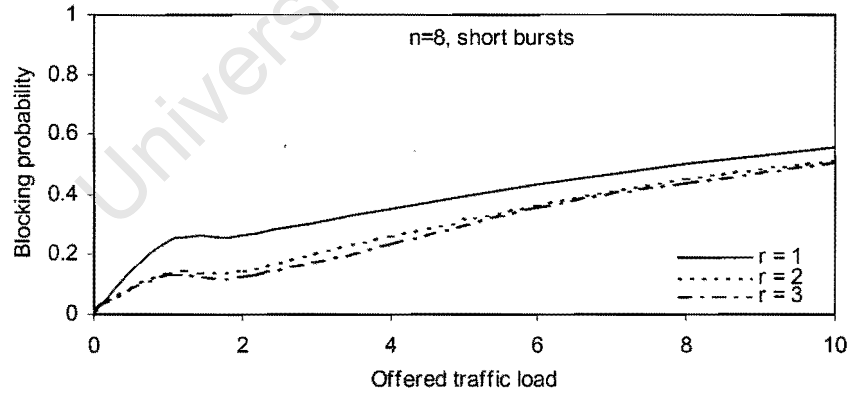


Figure 7.13: Multipath short bursts blocking probability with 8 alternative routes

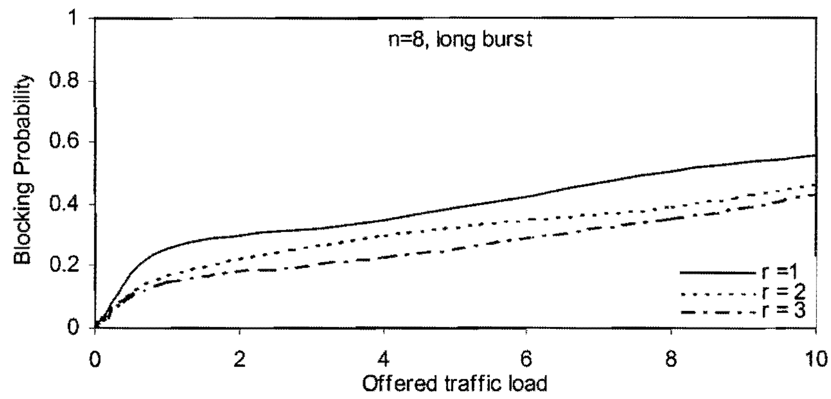


Figure 7.14: Multipath long bursts blocking probability with 8 alternative routes

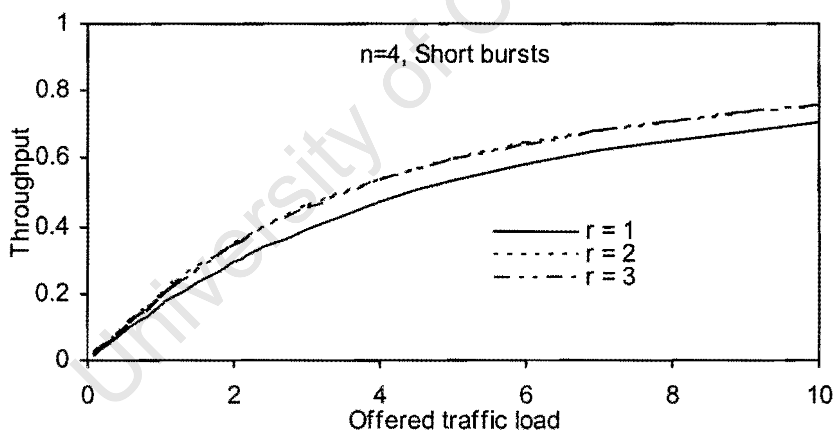


Figure 7.15: Throughput of short bursts with 4 alternative routes

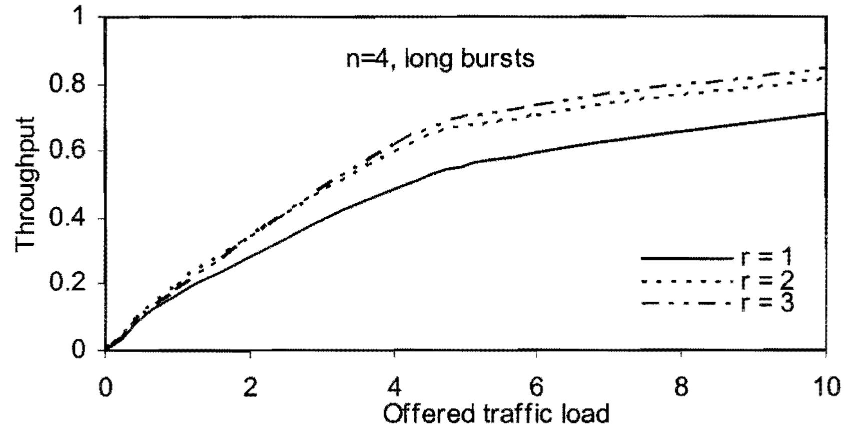


Figure 7.16: Throughput of long bursts with 4 alternative routes

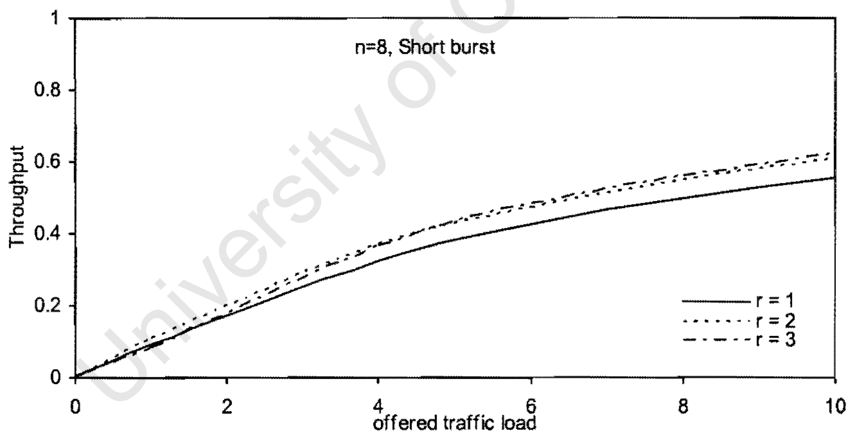


Figure 7.17: Throughput of short bursts with 8 alternative routes

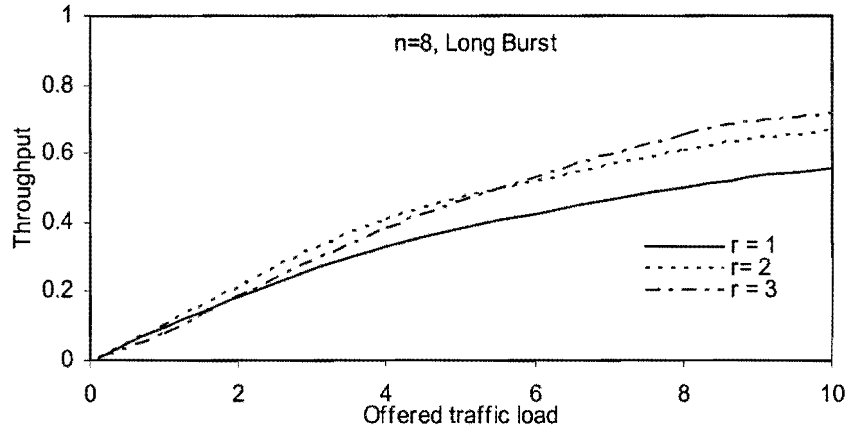


Figure 7.18: Throughput of long bursts with 8 alternative routes

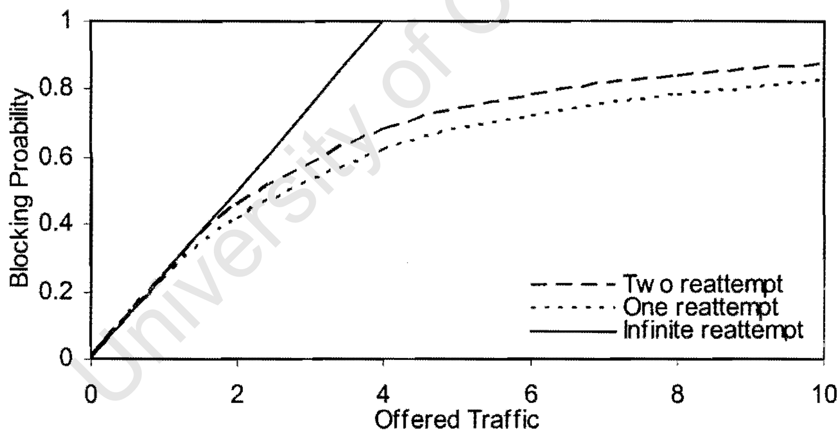


Figure 7.19: Single path burst blocking probability

Figure 7.20: Average single path reservation delay

Figure 7.21: Multipath average reservation time

7.7 Survivability to Failures

The prospect of broadband telecommunication networks loaded with massive quantities of information due to the wide range of services being supported causes extra concern for network survivability. This chapter has proposed a multipath routing-based dynamic bandwidth allocation protocol, which involves a routing process at burst level to provide the extra bandwidth required to accommodate bursty traffics. The routing process uses network nodes and links, which are subject to failure and total congestion. As the effect of node failures could be drastic, rapid restoration strategies are imperative (see Figure 7.22). This section shows how the presented protocol can be modified to handle link failures and maintains the integrity of network such as to guard against disruption of services due to unavoidable events, such as total nodal congestion, component and link failures, etc. Protection and restoration techniques must ensure short restoration times, efficient link utilisation, and as such must not be unnecessarily costly to implement. Furthermore, during the restoration process, simple bandwidth computation procedures must be implemented to ensure that low nodal processing loads of affected switches.

Different restoration techniques may be implemented at different layers or parts of a network, as was presented in [185], and illustrated in Table 7.3. In this regard, the key design consideration towards achieving a reliable ATM network with acceptable quality of service guarantees (QoS) is to enhance its capability to survive against any unexpected failures. Several protection methods have been studied including protection methods involving re-channelling of data only to a parallel link, without (necessarily) actually performing any routing. Hitless protection is one that protects the network without any errors or loss of data [185]. The approach taken in the design of the link-failure handling is to conserve the operation principles and the message complexity of the protocol. The protection must ensure hitless automatic protection switching (APS) [186], i.e. restoration with minimal errors. The restorations time should also be minimal. It makes no sense to try and 'save at any price' the reservation process when a failure occurs by adding complicated code, even for the simple reason that for some failures, e.g. those that dissect the *route-map*, nothing can be done to save the reservation process. It is important to note that once the connection is established, a link failure cannot be viewed as part of the reservation protocol, but should be treated by a general connection tear-down algorithm [187].

Table 7.3: Optional restoration techniques

Higher layers		Restoration methods
ATM Layer	VC level	PS
	VP level	PS, SHR, Mesh
SDH Layer (Physical)	Transmission. path level	APS, SHR, Mesh
	Digital section level	APS,SHR
	Regenerator section level and physical media	---
Optical layer		

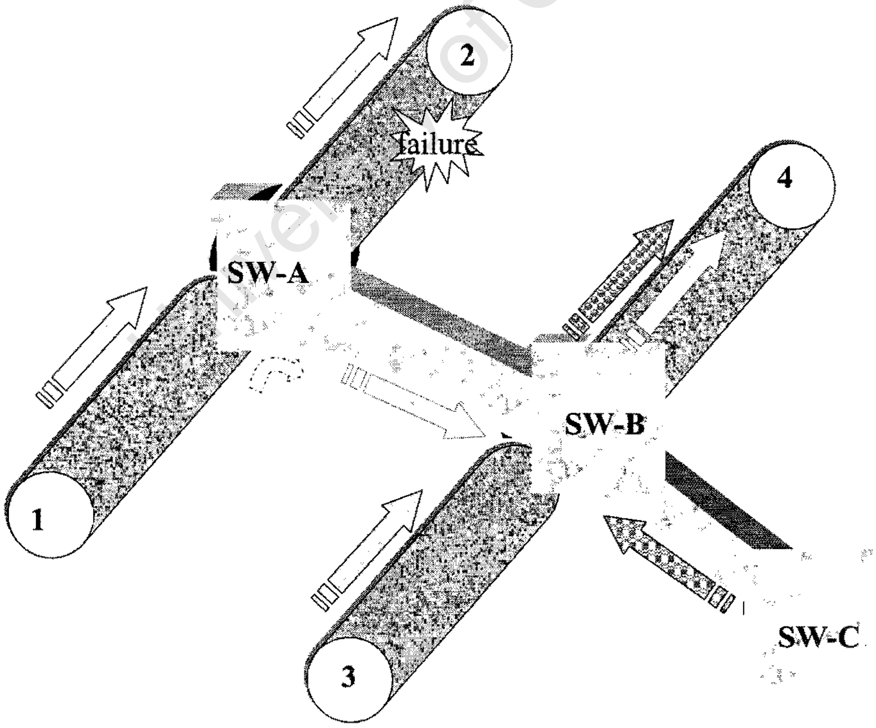


Figure 7.22: VP protection

The high reliability of fibre-optics high-speed networks may tempt us to handle the rare link failures with simple reset procedures that terminate the protocol operation. Such procedures are presented in section 7.8 and can be applied to our protocol presented in this chapter. However, since a *route-map* contains more than a single path between source and destination, it is desired that a failure in one (or more) of the links in the *route-map* will not automatically abort a progressing connection establishment process.

7.8 Protocol Reset Procedure

The procedures described in this section are designed to terminate the protocol operation and release the bandwidth allocated to the protocol when a failure or several failures occur. No effort is made to continue the search for a path to accommodate the burst. When a failure occurs, the two nodes at the endpoints of the failed link start a flooding algorithm such as PIF [182] with the Fail control cell. All nodes along the path receive Fail cells on their master-links, stop sending messages belonging to the protocol, and forwards the Fail cells to all their neighbours. The Fail cell must reach the source and the destination nodes that learn this way about the failure of the reservation process. When the node receives Fail cells from all its neighbours it releases the allocated bandwidth and sends Fail to l.

Due to imprecise global network information to set up a *route-map*, loops also can be formed when the REQUEST cells reach the destination, and control cells may get strayed and show up later after the burst has been accommodated. Our protocol can deal with this situation by rejecting all REQUEST cells that involve looping. When the second time a REQUEST cell reaches a node forming a loop, it will be rejected since it cannot provide better QoS than the first time it reached the node. When a stray cell shows up after a route has been released, it cannot re-search for a new alternative route since each node keeps a record for its latest master-link. Thus, when the stray cell reaches the destination, the destination will know that the alternative route has been found and will reject the request. Notice that the REJECT, ACCEPT and RELEASE cells always travel along a route without loops.

7.9 Conclusions

The use of high-speed networks to carry bursty traffic, such as IP packets, image, compressed video, etc, over ATM networks requires a new thinking. Instead of controlling connections, we are faced with the problem of controlling the bursts. Thus the new problem is how to control the bursts and how the resources are allocated optimally for the bursty applications. This chapter has proposed a multipath routing-based dynamic bandwidth allocation protocol to allocate bandwidth effectively for bursty traffic.

The protocol combines the QoS routing and bandwidth allocation strategies in order to achieve dynamics in bandwidth allocation, and consequently improves network utilisation. The multipath routing protocol is invoked at burst level, when the original path, which has been established at call setup phase, cannot accommodate the incoming burst. The protocol uses a flooding-like approach to search for feasible route on a *route-map* on the *burst-by-burst* basis.
Before a burst gets transmitted, the source must reserve the required resources end-to-end. This makes the protocol suit most non-interactive applications because of routing cost at burst level.

The performance of the multipath routing-based dynamic bandwidth allocation protocol has been evaluated analytically under an assumption that the network is heavily loaded. The performance results presented on Figures 7.11-7.21 serves as a motivation to adopt the multipath routing-based dynamic bandwidth allocation, and can effectively be used for maximizing network resource utilisation. It gives the node a choice of next hops for the same destination.

The protocol achieves good performance in terms of burst blocking probability and throughput when only one or two paths of the *route-map* are searched in addition to the original path for the required extra bandwidth to accommodate the incoming bursts on *burst-by-burst* basis. This fact and other practical considerations limit the multipath routing-based dynamic bandwidth allocation protocol to search for more than one or two paths in parallel, especially when the incoming bursts are short in size.

The single path reservation schemes with crank back show increase on average reservation delay as the number of permitted crank back attempts increase. This is because the effective arrival rate experienced by the network, λ_{ef}, increases and consequently the blocking probability increases as well (Figure 7.20 and Figure 7.21). This increase in burst reservation delay will negatively effect delay sensitive applications. However, the multipath routing-based dynamic bandwidth allocation protocol shows a very significant reduction in reservation delay to accommodate the incoming bursts compared to the single path reservation approach (Figure 7.22).

Chapter 8

Conclusion and Future Research Directions

8.1 Summary

The diversity of traffic types in high-speed ATM networks introduces challenging problems on the underlying network. No complete solution has yet been provided for multimedia applications, i.e. video and audio. The difficulty lies in the nature of the multimedia application where the traffic is highly bursty and correlated. Traditionally, the static bandwidth allocation task does allocate fixed bandwidth for the entire duration of the connection at call level (call setup phase) only, using CAC mechanisms. In this case, the user must accurately specify its required QoS parameters. Then the network uses the QoS requirements to allocate a permanent amount of bandwidth for the connections, either deterministically or statistically, at call setup phase. However, when the traffic offered to the connection is bursty, traditional static bandwidth allocation schemes would not be efficient for at least three reasons. Firstly, when the application is delay sensitive, it is impossible to characterise the incoming traffic precisely. Secondly, the traffic is inherently bursty, which makes peak-rate deterministic bandwidth allocation schemes excessively conservative and costly in terms of bandwidth utilisation. Thirdly, VBR traffic is highly correlated, and as such it exhibits long-range dependency, which degrades statistical multiplexing gains, thus resulting in poor resources utilisation.

This research has proposed an alternative approach to managing and allocating network resources in a more flexible and efficient ways to solve the problem of poor network resources utilisation caused solely by bursty traffic. This research suggests that the bandwidth is allocated dynamically in order to control the burstiness, and consequently improve network utilisation. This goal has been achieved by two approaches. In Chapter 5, the adaptability on bandwidth allocation was achieved by seeking the required extra bandwidth to accommodate the bursts on the same transmission path that was established during call setup phase. Thus, the proposed protocol periodically renegotiates the bandwidth that has been allocated at call setup phase. The on-line predictors are used to forecast the future traffic behaviour. The prediction process was derived from measurements of the traffic that have been observed so far and the QoS that has been experienced. This protocol increases the perceptual video quality, especially for real-time video, and improves network utilisation. The proposed protocol assumes that the traffic source can accept a graceful degradation of QoS in the case of network congestion, when the required bandwidth cannot be provided. In this model, the traffic source is a scalable video, compressed on multilayer format.

The renegotiation-based dynamic bandwidth allocation protocol has achieved a better performance in terms of cell loss, throughput, and utilisation than the traditional static bandwidth allocation schemes using non-layered video compression model, as have been reported in Chapter 6. The protocol can also achieve, a significant improvement on the network performance, provided that the renegotiation point intervals are selected carefully. The tradeoff between the renegotiation time intervals and network performance of the protocol was also quantified in Chapter 6. We have noticed that, the more frequent are the renegotiations, the better network performance, but at the expense of higher signalling overhead. Thus, the renegotiation time intervals have to be selected based on traffic characteristics and signalling overhead. While the renegotiation-based dynamic bandwidth allocation protocol doesn't involve any routing process at burst level, which will have a negative impact on delay sensitive applications, the chance to find the required extra bandwidth on the original connection might fail.

However in Chapter 7, we have chosen another approach for dynamic bandwidth allocation for non-interactive applications. A multipath routing-based protocol has been proposed. The protocol seeks the required extra bandwidth on alternative routes to accommodate the bursts, when the burst can not be accommodated on the original path that was established at call setup phase. The protocol is based on a multilevel bandwidth allocation strategy combined with multipath routing at burst level. Thus the bandwidth is allocated and managed at call, burst, and cell level. This protocol efficiently handles bursty traffic from non-interactive applications by searching for extra required bandwidth on alternative routes in parallel. In this protocol, neither the renegotiation of source's traffic parameters, nor the degradation on QoS of the traffic source is considered. Therefore, the adoption of the proposed protocol to the multipath routing strategy seems natural for bursty traffic.

The effect of parallelism on the performance of the multipath routing-based dynamic bandwidth allocation protocol was also quantified in Chapter 7. The proposed protocol is appropriate for an ATM network that supports the transfer of high-speed bursty traffic for non-interactive applications.

8.2 Future Research and Implementation Requirement

The requirement for the renegotiation-based dynamic bandwidth allocation protocol is that the network allows renegotiation of the source's traffic descriptor. An extension to the current ATM signalling protocols is required to achieve efficient renegotiation at lower signalling cost, and, in the meantime, the renegotiation interval must be carefully controlled to be initiated at an optimum time scale.

The ITU-T Q.2963 recommendations have defined an extension of the ITU-T Q.2931 signalling to support renegotiation messages [92][93]. However, the ATM Forum has not yet defined renegotiation messages, but renegotiation is planned as a future work item for the next releases. Further work should proceed to cover the implementation issues for the renegotiation-based dynamic bandwidth allocation protocol. Also, further work is also needed, on the video compression techniques to develop a standard and more efficient multilayer video encoder that is able to produce a layer video stream with lower overhead and higher compression rate.

In this work we have realised that if the CLP field on the ATM header is modified to be at least 2-bit size, this will result in 2^2 different priorities, which will simplify the implementation complexity of our protocol significantly. Thus, instead of establishing a VC for each layer of the multilayer video stream to carry the base and the enhancement layers concurrently, all substreams could be transmitted on a single VC with a different priority tagging.

The implementation of the multipath routing-based dynamic bandwidth allocation protocol requires that the protocol be informed about bursts occurrence times and its bandwidth requirements. The burst occurrence is determined either at the source or at the network edge. When the burst occurrence is determined by the source, the source should be intelligent enough to find out when it becomes bursty. The source can either use predictors for real-time applications (see section 5.5) or compile the non-real time application to determine in advance the bursts occurrence times. When the network edge is responsible for determining bursts occurrence, a performance measurement-based algorithm with certain threshold at the network edge determines when the source is bursting. So, further work is required at source node and network edge as well to characterise bursty traffic and predict its behaviour in a more accurate manner. This will involve fuzzy and neural networks to make the source node more intelligent to predict its burst occurrences time and the extra amount of bandwidth required to accommodate it.

To implement the multipath routing-based dynamic bandwidth allocation protocol, the traffic source must maintain a list of connections that are currently in the setup phase, and assign each connection a unique id. The source id and connection id must be carried by all the messages of the protocol, so they can be uniquely identified. Also, all the switches on the *route-map* and signalling protocols like PNNI must modified to support the multipath routing on *burst-by-burst* basis.

Further work should be carried out with a test bed over ATM switches.

8.3 Possible Application Areas

The two areas of application that seems to have the highest potential to be influenced by the adoption of renegotiation-based dynamic bandwidth allocation protocol are multimedia transmission over wireless ATM networks and multiple destinations (multicasting).

The proposed multipath routing-based dynamic bandwidth allocation protocol can be integrated both into ATM and into TCP/IP. In ATM, according to the PNNI standard, each node keeps a network map that becomes less detailed the further the described area is from the node. This effect is due to the route-hierarchical structure that hides details of areas that are not descendents of the current switch in the hierarchy. The source node builds a route that becomes less detailed as the route extends away from the source towards the destination. Intermediate switches are responsible for finding the exact route in their neighbourhood. In case of reservation failure, PNNI supports crank-back, which are considered too time consuming. However, allowing the traffic source to seek the required extra bandwidth on multiple routes in parallel, as we have proved in Chapter 7, can increase the reservation success probability, increase the route optimality, and setup the route faster than in the current implementation. The results reported in Chapter 7 show that the multipath routing-based dynamic bandwidth allocation protocol achieves better network performance compared with single path routing schemes. Good performance has been achieved even when one or two paths were searched in addition to the original path for the extra bandwidth to accommodate the incoming bursts. This fact, and other practical considerations, limit the multipath routing-based dynamic bandwidth allocation protocol to searching for more than one or two paths in parallel, especially when short bursts are considered.

Appendix-A

In order to calculate the probability of unsuccessfully finding an alternative route to accommodate the incoming burst (burst blocking probability). The system steady state probabilities, $\pi_{\sigma,\omega}$ should be found first by solving the system equilibrium equations, $\vec{\pi}Q = 0$, where Q is the infinitesimal transition matrix derived from Equations (7.1-3), together with the probability conservation relation Equation (7.5). The appendix uses the Markov chain and the transition rates of Equations (7.1-3), to write the following $n(n+1)/2-1$ equilibrium equations:

$$\begin{aligned} q_{(\sigma,\omega),(\sigma,\omega)}\pi_{\sigma,\omega} = {} & q_{(\sigma-1,\omega-1),(\sigma,\omega)}\pi_{\sigma-1,\omega-1} + q_{(\sigma,\omega-1),(\sigma,\omega)}\pi_{\sigma,\omega-1} + \\ & q_{(\sigma+1,\omega),(\sigma,\omega)}\pi_{\sigma+1,\omega} + q_{(\sigma,\omega+1),(\sigma,\omega)}\pi_{\sigma,\omega+1} \\ & 2 \le \sigma \le n-2, \quad 0 \le \omega \le n-(\sigma+1) \end{aligned} \tag{A.1}$$

$$q_{(0,\omega),(0,\omega)}\pi_{0,\omega} = q_{(0,\omega-1),(0,\omega-1)}\pi_{0,\omega-1} + q_{(1,\omega),(0,\omega)}\pi_{1,\omega} + q_{(0,\omega+1),(0,\omega)}\pi_{0,\omega+1} \quad 1 \le \omega \le n-1 \tag{A.2}$$

$$q_{(0,0),(0,0)}\pi_{0,0} = q_{(1,0),(0,0)}\pi_{1,0} + q_{(0,1),(0,0)}\pi_{0,1} \tag{A.3}$$

where $q_{(\sigma,\omega),(\sigma,\omega)}$, the transition rate out of state (σ,ω), is given by:

$$q_{(\sigma,\omega),(\sigma,\omega)} = \sigma\alpha + \omega\mu + \sum_{l=0}^{\min\{n-(\sigma+\omega+1),1\}} \delta(l,\sigma+\omega)\lambda \tag{A.4}$$

The recursion relations could be written for $\pi_{\sigma,\omega}, \sigma > 0$

$$\pi_{1,\omega} = (q_{(0,\omega),(0,\omega)}\pi_{0,\omega} - q_{(0,\omega-1),(0,\omega)}\pi_{0,\omega-1} - q_{(0,\omega+1),(0,\omega)}\pi_{0,\omega+1})/q_{(1,\omega),(0,\omega)} \qquad \omega = 1,2,\ldots,n-1$$

$$\begin{aligned}\pi_{\sigma,\omega} = (&q_{(\sigma-1,\omega),(\sigma-1,\omega)}\pi_{\sigma-1,\omega} - q_{(\sigma-2,\omega-1),(\sigma-1,\omega)}\pi_{\sigma-2,\omega-1} - q_{(\sigma-1,\omega-1),(\sigma-1,\omega)}\pi_{\sigma-1,\omega-1} \\ &- q_{(\sigma-1,\omega+1),(\sigma-1,\omega)}\pi_{(\sigma-1,\omega+1)})/q_{(\sigma,\omega),(\sigma-1,\omega)} \qquad \sigma = 2,3,\ldots n \,\&\, \omega = 1,2,.n-\sigma\end{aligned} \tag{A.5}$$

When more than one route is considered for searching $r > 1$, Equation (A.1) takes the form,

$$q_{(\sigma,\omega),(\sigma,\omega)}\pi_{\sigma,\omega} = q_{(\sigma+1,\omega),(\sigma,\omega)}\pi_{\sigma+1,\omega} + q_{(\sigma,\omega+1),(\sigma,\omega)}\pi_{\sigma,\omega+1} + \sum_{l=\max\{0,\sigma-(r-1)\}}^{\sigma} q_{(l,\omega-1),(\sigma,\omega)}\pi_{l,\omega-1} \tag{A.6}$$

where $q_{(\sigma,\omega),(\sigma,\omega)}$, the transition rate elements out of state (σ,ω) is given by

$$q_{(\sigma,\omega),(\sigma,\omega)} = \sigma\alpha + \omega\mu + \sum_{l=0}^{\min\{n-(\sigma+\omega+1),r-1\}} \delta(l,\sigma+\omega)\lambda \tag{A.7}$$

$$\begin{aligned}\pi_{\sigma,\omega} = \{&q_{(\sigma-1,\omega),(\sigma-1,\omega)}\pi_{\sigma-1,\omega} - q_{(\sigma-1,\omega+1),(\sigma-1,\omega)}\pi_{\sigma-1,\omega+1} \\ &- \sum_{l=\max\{0,\sigma-r\}}^{\sigma-1} q_{(l,\omega-1),(\sigma-1,\omega)}\pi_{l,\omega-1}\}/q_{(\sigma,\omega),(\sigma-1,\omega)} \qquad \sigma = 1,2,3,\ldots,n \quad \omega = 1,2,\ldots,n-\sigma\end{aligned} \tag{A.8}$$

$$q_{(\sigma,n-\sigma),(\sigma,n-\sigma)}\pi_{\sigma,n-\sigma} = q_{(\sigma,n-\sigma-1),(\sigma,n-\sigma)}\pi_{\sigma,n-(\sigma+1)} + q_{(\sigma-1,n-\sigma-1),(\sigma,n-\sigma)}\pi_{\sigma-1,n-(\sigma+1)} \qquad 1 \le \sigma \le n-1 \tag{A.9}$$

$$\sum_{(\sigma,\omega)} \pi_{\sigma,\omega} = 1$$

When more than route is considered for searching, ($r > 1$), Equation (A.9) takes the form:

$$q_{(\sigma,n-\sigma),(\sigma,n-\sigma)}\pi_{\sigma,n-\sigma} = \sum_{l=\max\{0,\sigma-(r-1)\}}^{\sigma} q_{(l,n-(\sigma+1)),(\sigma,n-\sigma)}\pi_{l,n-(\sigma+1)} \qquad 1 \le \sigma \le n-1 \tag{A.10}$$

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