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Experiments Involving the Transmission of Layered Video

Over a Local ATM Network

By

Larry Andrew Baptiste

A thesis submitted in conformity with the requirements for
the degree of Master of Applied Science
Graduate Department of Electrical and Computer Engineering
University of Toronto

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Abstract

In this thesis we investigate the feasibility of transmitting layered video over a local ATM network. Two layered video schemes were looked at: subband coded layered video and MPEG-2 SNR scalable layered video. In both cases the video was transmitted as a constant bit rate (CBR) base layer and an unspecified bit rate (UBR) enhancement layer, we then introduced background traffic (both CBR and UBR) in order to determine their effect on the layered video in terms of delay, jitter and loss.

The results obtained for the subband coded video were quite promising, with competing cross traffic in the network having very little effect on base layer jitter or loss. We also found that specifying the cell delay variation tolerance (CDVT) was a factor in preventing base layer loss. Losses in the enhancement layer occurred only when total utilization exceeded 100%, also maintaining synchronization between the two layers did not prove to be problematic even at very high network utilization.

Similar results were obtained for the MPEG-2 scalable video, however due to the much higher bit rates in this case, we found maintaining synchronization between the two streams to be more difficult.
Acknowledgments

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Chapter One: Introduction

1.1 Introduction

Recent technological advances in fiber optics and switching systems have provided the technological basis for the development of high capacity Broadband Integrated Services Networks (B-ISDNs). B-ISDNs are very high speed communication networks, which are capable of supporting transmission speeds of several hundred Mbits/s, and can carry various types of traffic. Hence, in addition to the traditional data and voice services, B-ISDNs will also support the transport of images, teleconferencing, video, and large volumes of interactive computer data. Accordingly, B-ISDNs will support services with both constant and variable bit rates, as well as connection oriented and connectionless transfers. Asynchronous Transfer Mode (ATM) is the choice of the pertinent standardizing body - the International Telecommunications Union Telecommunications Standardization Sector (ITU-T)- for the implementation of B-ISDNs due to its flexibility and efficiency. Simply put, ATM is a connection oriented, packet-switching and multiplexing technique that uses short fixed 53-byte cells to transfer information over a network. ATM is a compromise that allows the integration of different services with different characteristics and requirements. Being a compromise solution, ATM is not optimized for any particular service, be it voice, data, or video; however, since video transmission is one of the main driving forces for the deployment of B-ISDNs the characteristics of video over ATM is an important issue.

1.2 Video Transmission over ATM Networks

There are a number of issues that are fundamental in the deployment of ATM networks. They include:

- Accurate characterization of the video traffic.
- Efficient bandwidth allocation methods for the video traffic.
- Specification of the video services that ATM networks should support.
• Implementation of efficient congestion control mechanisms, call admission procedures, and usage parameter control (UPC) algorithms.

Accurate characterization of the multimedia traffic is essential in order to develop a robust set of traffic descriptors. Unfortunately, there are no comprehensive measurements to permit designers to adequately address the characteristics of various types of B-ISDN applications in a realistic manner. This is especially true for variable bit rate (VBR) traffic.

The characteristics of voice traffic has been studied for decades, and are relatively well understood. Data sources can be viewed as well understood, but unpredictable. Image and VBR video transmission over communications networks are relatively new research areas, and as such current knowledge of their source behavior is limited and implementation specific. Although significant research effort has focused on the development of efficient information multiplexing schemes for the ATM environment, most of the practical problems related to real-time applications remain unresolved.

A VBR encoder attempts to keep the quality of video output constant, while at the same time reducing bandwidth requirements by transferring the minimum of information. However, because VBR video is both highly variable and delay sensitive, high speed networks (such as ATM) are generally implemented by assigning peak rate bandwidths to VBR video applications, and using the residual bandwidth for non-real time traffic. This approach, however, is inefficient, and in order to define bandwidth allocation schemes which provide an adequate quality of service (QoS) for VBR applications and minimize bandwidth wastage, the effects of the applications on the network must be investigated.

In general, the ITU-T describes four types of video service that an ATM network can provide. These are:

• Distribution services - these are essentially broadcast services, where a source may distribute a video signal to many users. Communication in this case is usually unidirectional. Examples are TV and cable services.

• Conversational services - in this case information may flow from many sources to many destinations. Examples of this type of service are videotelephony and videoconferencing.
• Messaging services - an example of this is the moving picture analog to voice mail.
• Retrieval services - these involve the retrieval of images or video sequences from libraries and databases. Video on demand (VOD) is an example of this service.

These service offerings have broad implications on the function that an ATM network must provide. For example, distribution and conversational services require point-to-multipoint and multipoint-to-multipoint call setup and maintenance procedures. The most difficult set of problems have to do with service guarantees, traffic contracts with the users, traffic descriptors, network resource allocation, and network traffic control.

One solution to the resource allocation problem is to constrain the bit rate of the video produced at the source to a constant rate. However, in order to produce constant bit rate (CBR) video, a smoothing buffer is necessary to regulate the bit rate at the network interface. The buffer size is determined by the delay characteristics of applications. Since they can hold a finite amount of data, smoothing buffers are subject to overflows. However, video services are not tolerant to cell losses. To solve the overflow problem, the bit generation rate at the encoder is reduced when the number of bits at the smoothing buffer reaches a threshold. This is achieved at the expense of increased quantization levels, causing degradation of the image quality. Constant bit rate traffic is easy to manage, a constant bit rate is reserved for each connection throughout its duration, whether the source is actively transmitting or silent. Video compression schemes however result in traffic that is highly variable, being encoding scheme and scene activity level dependent. Presenting the VBR traffic produced to the ATM network as CBR has the drawbacks of under-utilization of network resources and QoS degradation. Several VBR video sources can be multiplexed together using VBR services to achieve a multiplexing gain as compared to CBR services in the network. For example, Maglaris et al. [1] provide justification, using analysis and simulation, for the multiplexing of VBR sources; while Kishino et al. [2] present experimental verification of multiplexing gains, for the multiplexing of video sources. VBR transmission also has the advantages of near constant picture quality, and reduced processing delays in codecs.

Finally, a video frame contains a large amount of information and is transmitted over a number of ATM cells. Losing a cell in the middle of a frame may cause synchronization
to be lost, or may corrupt a large part of a frame. Unfortunately cell losses occur in bursts in ATM networks, this leads to complications as it is difficult to compensate for consecutive cell loss. In order to minimize the effect of consecutive cell loss it might be necessary to request strict service from the network. Unlike data, cell loss compensation by retransmission is not a viable alternative for real-time services such as video due to the strict delay constraints.

1.3 Layered Video

VBR coding has the potential advantages of providing consistent picture quality, bandwidth savings and delay reduction. However, statistical multiplexing of VBR sources can lead to network congestion, leading to cell losses and delay jitter. The coding procedure can be controlled so that the generated traffic characteristics conform to the service contract specification, however rate control of the encoding procedure will cause picture quality degradation. Layered source coding (LSC) can be used to overcome this drawback and maintain the advantages of VBR coding.

Figure 1.1 : Layered Coding System.

Figure 1.1 illustrates the layered video concept; an encoder produces video made up of a number of layers, typically two layers are produced: a base layer and an enhancement layer. The base layer can be decoded to produce output video of adequate quality, while both the base and enhancement layers can be decoded together to produce a high quality video output.
The following list describes some of the motivating factors behind layered video:

- The use of layered video allows the transmission of VBR video, while simultaneously allowing CBR reservations to be made in the network. This is done by using rate control when encoding the base layer, while encoding the base and enhancement layers to a constant quality. The base layer can then be sent using guaranteed service in the network, while the enhancement layer can be sent without any reservations.

- Digital networks are made up of a heterogeneous mix of subnetworks, which may support varying bandwidths and qualities of service, encoding video into a range of layers allow subnetworks to subscribe to the appropriate amount of layers that they can support.

- Different endpoints may simply want to subscribe to a smaller number of layers because that is all they are willing to pay for.

Various methods for producing layered video [1,2,4,5,6] have been described in the literature. Some of these methods are reviewed in appendix C.

1.4 Thesis Outline

In chapter 2, we present some background into ATM, look at the fundamentals of video coding, describe important ATM parameters which relate to video, and finally, review some experimental results obtained from ATM networks. Chapter 3 examines cell level measurements for cell transfer delay (CTD) for constant bit rate (CBR) and unspecified bit rate (UBR) traffic. In chapter 4 we present results obtained for subband coded layered video, while in chapter 5 similar results are presented for MPEG-2 layered video. Finally, in chapter 6 we summarize the results we obtained and state our conclusions.
Chapter Two: Background

2.1 Introduction

In this chapter we first look at asynchronous transfer mode in some detail; we then very briefly introduce the fundamentals of video coding. We then look at some video related ATM parameters, and finally at some experimental results from actual ATM networks.

![ATM protocol reference model for B-ISDN](image)

**Figure 2.1:** ATM protocol reference model for B-ISDN.

2.2 Asynchronous Transfer Mode [13,16,17]

The ATM protocol reference model for B-ISDN is shown in Figure 2.1. It consists of three planes: user, control, and management. The user plane transmits end-to-end user information between two or more communicating end nodes. The control plane protocols deal with call establishment and release, and other connection control functions necessary for providing switched services. The management plane provides for operations and management functions, as well as facilitating communication between the user and control planes.
This plane is further divided into the layer and plane management planes; layer management involves layer specific management functions such as the detection of failures and protocol abnormalities. While plane management provides management and coordination functions related to the complete system.

All these planes utilize the physical and ATM layers, while the ATM adaptation layer (AAL) is service-specific and may or may not be used depending on the particular application.

When used, the AAL provides an interface between the user layer and the ATM layer for applications with similar service requirements. Different AALs are defined to support different types of traffic. The ATM layer provides the switching and multiplexing functions for the traffic. There is no awareness of the application specifics at this layer, this is necessary in order to facilitate the high-speed network links. Finally, the physical layer transports ATM cells between two adjacent ATM layers.

2.2.1 ATM Adaptation Layer

The ATM adaptation layer (AAL) resides between the ATM and the next higher layer in both the user and the control planes. It is used to enhance the services provided by the ATM layer to the next higher layer. Hence the AAL is service dependent. It isolates the higher layers from the ATM layer by mapping the higher data units into the ATM cell payload and vice versa. Generally, the ATM forum recommends that traffic be classified into constant bit rate (CBR), variable bit rate (VBR), available bit rate (ABR), or unspecified bit rate (UBR). It is impractical to address the diverse set of applications served by ATM networks in a single AAL framework, therefore the functionality required by various applications are grouped into a small number of classes which have similar service requirements and traffic characteristics.

As shown in Table 2.1 [13], four AALs are defined based on the following three parameters:

- Timing relationship between source and destination (required or not required).
- Bit rate (constant or variable)
- Connection mode (connection-oriented or connectionless services)
### AAL Type

<table>
<thead>
<tr>
<th>AAL Type</th>
<th>1</th>
<th>2</th>
<th>3/4</th>
<th>5</th>
</tr>
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<td>Class B</td>
<td>Class C or D</td>
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<td>Variable</td>
<td>Variable</td>
</tr>
<tr>
<td>Connection Mode</td>
<td>Connection-oriented</td>
<td>Connection-oriented</td>
<td>Connection-oriented or Connectionless</td>
<td>Connection-oriented or Connectionless</td>
</tr>
</tbody>
</table>

**Table 2.1: Currently defined AALs.**

The functions of the AALs are grouped into two sublayers; the segmentation and reassembly (SAR) sublayer, and the convergence sublayer (CS). The SAR sublayer deals with the segmentation and reassembly of data units, and maps them into fixed-length cell payloads. The CS performs a set of AAL service specific functions. It is further divided into a service-specific convergence sublayer (SSCS) and a common part convergence sublayer (CPSS). The SSPS may be null for applications that do not require any service-specific function.

The following is a brief description of the various AAL types:

- **AAL type 1** - this was designed to support constant bit rate services such as voice or CBR video. It supports the service requirements of the user layer by providing timing recovery, timing indication, synchronization, and an indication of lost information. AAL type 1 utilizes pre-established (via signaling) AAL connections.

- **AAL type 2** - this was designed to support variable bit rate signals which require the transfer of timing information between the source and the destination. It is envisioned for use with variable bit rate video signals. No protocol formats have been agreed upon for AAL type 2.

- **AAL 3/4** - this provides for the end-to-end transport of variable bit rate data packets. The protocol model for this layer is shown in Figure 2.2. The common part of the convergence utilizes the services of the SAR to provide for the unassured transport of variable-data units. The SSCS provides for assured mode through the use of sequence
numbers and re-transmission of lost or corrupted data units. When used to provide unassured data transport, the SSCS is null. AAL 3/4 is defined for connection-oriented and connectionless VBR services that require no timing relationship between source and destination. It can support connectionless network access such as that between two LANS over a WAN, or connection-oriented frame relay service.

- **AAL 5** - this was designed to support service class C, connection oriented variable bit-rate data. It was initially designed to provide for the transport of IP data packets, however it became apparent that this AAL could be used to support any class C service. The key difference between AAL type 5 and AAL type 3/4 is that the former was designed with the explicit intention of transporting TCP/IP, and therefore assumed that some of functions included in the assured mode of AAL type 3/4 (such as frame retransmission) would be provided by a higher protocol layer.

![Figure 2.2: AAL type 3/4 protocol model.](image)
2.2.2 The ATM Layer

The ATM layer transfers cells between peer ATM layer entities. It receives a 48-byte from the ATM layer user and appends a 4 byte header, which together with the header error check (HEC) byte makes up the ATM cell.

The ATM cell header at the user network interface (UNI) (Figure 2.3) consists of several fields:

- Generic flow control (GFC).
- Virtual path identifier (VPI).
- Virtual channel identifier (VCI).
- Payload type (PT).
- Reserved field (Res).
- Cell loss priority (CLP).
- Header error check (HEC).

![ATM header at the UNI](image)

The generic flow control (GFC) field supports access to a shared medium or support for multiple levels of priority for terminals sharing a single interface.

In ATM, end-to-end virtual channels are established between end stations before traffic flow can be initiated. Routing of cells is done on a per cell basis at each ATM switch.
The VPI and VCI carry the routing information for a cell. The two levels of routing hierarchies, virtual paths (VPs) and virtual channels (VCs) are defined as follows:

- In order to establish an end-to-end connection between a source and a destination, a link or series of links is required. Each link is referred to as a virtual channel, with each virtual channel identified by a common unique identifier value referred to as a VCI.
- A virtual path (VP) is a generic term for a bundle of VC links, all the links in the bundle have the same endpoints. A VP in a cell is identified by a virtual path identifier.

Users are provided a choice of either a virtual path connection or a virtual channel connection, defined as follows:

- Virtual Path Connection (VPCs) - these are switched based upon the VPI value only.
- Virtual Channel Connections (VCCs) - These are switched upon the combined VPI and VCI value.

The payload type indicator specifies whether the contents of a payload carries user or management data. The CLP is a 1-bit field used for cell loss priority, cells are classified as either low or high priority. A buffer which is congested would discard a low priority cell before a high priority cell.

Finally, the HEC field is used mainly for two purposes; discarding cells with corrupted headers and for cell delineation. The HEC used in ATM has the capability to detect and correct single-bit errors. It can also detect double-bit and some other combinations of bit errors. However, if the error correction capability is applied to an errored cell header, then it is possible that the cell header is still errored after the correction if there were originally multiple-bit errors. The ATM Forum [13] recommends that cells with detected errors at the cell header be discarded, without an attempt at error correction. The HEC is added at the physical layer.

Within an ATM transport network there is no ATM layer user in the user plane, and cells are passed from the receiving ATM layer entities to the transmitting counterparts at each...
switching node between the source and destination end stations. Hence, the ATM layer mainly provides the switching function of ATM networks. Cells with errored headers are discarded at intermediate nodes, and congestion may lead to cells being dropped, however the ATM layer is unreliable, i.e. there is no retransmission of errored and lost cells inside the ATM network, and it is up to the end stations to ensure the integrity of the data carried in ATM cell payloads.

2.2.3 Physical Layer

The physical layer transports ATM cells between two adjacent ATM layers. There is a wide variety of physical link types such as coaxial-cable, single mode fiber and multimode fiber. The physical layer in ATM has more functionality that is typically associated with this layer, since it delivers cells (not bits) to the ATM layer, which requires determining the cell boundaries.

2.3 Traffic Control and Resource Management in ATM Networks [3,21]

In ATM networks because resources are shared, it is essential to monitor and control the flow of traffic. Without sufficient control, demand for insufficient resources can seriously degrade network throughput and efficiency, hence traffic control is necessary to protect both the level of services perceived by users and also ensure the efficient utilization of network resources. As previously stated, ATM networks provide end-to-end sequential cell transport, a user makes a request for a virtual connection which is accepted or rejected by the network. After the acceptance of a connection, carried cells may experience two types of impairments within the network, namely delay and loss. Quality of service (QoS) refers to the set of parameters such as cell delay, cell delay variation and cell loss rate that are related to the user’s perception of the network service.

Traffic control is difficult in ATM networks because of conflicting objectives, we would like to multiplex traffic flows for efficiency, while simultaneously isolating traffic flows for QoS protection. A gain in efficiency can be achieved by statistically multiplexing VBR connections whose total combined rate at any given time is less than the link rate. If the traffic streams are numerous and independent, then the probability that their instantaneous total rate will exceed the link rate will be small. To realize the statistical
multiplexing gain it is desirable to maintain a high utilization factor and to maximize the degree of sharing of network resources. Unfortunately, statistical multiplexing may lead to the QoS of one connection being adversely affected by traffic in the other connections, for example a burst in one stream could fill the multiplexer buffer and thereby increase cell delays for all streams. Also many traffic streams could have simultaneous bursts and cause buffer overflow. Hence a low utilization factor is desirable to minimize queuing delay and buffer overflow, which conflicts with efficient use of network resources.

A description of the traffic to be sent forms the basis of a traffic contract between the end user and the ATM network provider. An ATM traffic description consists of a set of traffic parameters that can be used to characterize an ATM connection. Real-time connections are described by a number of source traffic descriptors, namely peak cell rate (PCR), sustainable cell rate (SCR) and burst tolerance (BT). The specification of the PCR is mandatory for both CBR and VBR connections, whereas SCR and BT are optional parameters used for VBR connections. The additional specification of SCR and BT may allow the network provider to allocate less resources to a connection while still maintaining the required QoS.

During the lifetime of a compliant connection, the network provider agrees to guarantee the requested QoS to all conforming cells. In order to monitor the conformance of cells to the negotiated contract, the generic cell rate algorithm (GCRA) has been proposed by the ATM forum. The GCRA(I,L) is a counter scheme based on two parameters, the increment value I, and the limit value L. For each arriving cell the counter is incremented by I, and conversely decremented by one each cell slot. The counter limit is given by I+L, and a cell that would cause the counter to overflow is said to be non-conforming. Therefore, the increment parameter I affects the cell rate of a connection, and the limit parameter L the burst length.

Connection requests are made through the exchange of signaling information. Call admission control (CAC) uses the connection request to determine:

- An estimation of the resulting QoS if the new connection is accepted.
- Traffic parameters for user parameter control (UPC).
- The network resources to be allocated.
A new connection is admitted by the network only if it is estimated that the new connection can be established with the required QoS without affecting the guaranteed QoS of existing connections. The connection request is passed along the route, each node decides whether it can allocate the necessary resources, and the connection is accepted if each node agrees to accept it. Acceptance of the new connection implies agreement on a traffic contract specifying the obligations between the user and the network.

2.4 Video Services

Digital representation of video signals requires the representation of continuous image data in a discrete manner. To this end, each value over the continuous range of the image is converted into a finite set of discrete amplitudes that span the intensity range of the image using pulse coded modulation. Typically the raw bit rate requirements of video applications vary anywhere from a few to several hundreds of Mbits/s, hence there is simply not enough capacity available to accommodate the uncompressed video signals. Thus compression is needed. Table 2.2 [14] indicates the bit rate requirements of a number of video services before and after compression.

Video is presented to users as a series of frames at a rate of around 30 frames per second, and fortunately video sequences contain a significant amount of redundancy, both within a frame (intraframe) and between consecutive frames (interframe). The intraframe spatial redundancy is due to the high probability of neighbor pixels within a frame being similar. The interframe temporal redundancy is due to the common occurrence of minor variations between frames close together. It is, thus, possible to reduce the application bit rate by taking advantage of these redundancies.
<table>
<thead>
<tr>
<th>Service</th>
<th>Example</th>
<th>Raw (kbits/s)</th>
<th>Compressed (kbits/s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Real time (1/4 screen low resolution)</td>
<td>(128 x 120 pixels) (9 bits/pixel; 15 frames/s)</td>
<td>2074</td>
<td>64</td>
</tr>
<tr>
<td>Real time (1/4 screen high resolution)</td>
<td>(128 x 240 pixels) (9 bits/pixel; 15 frames/s)</td>
<td>4147</td>
<td>384</td>
</tr>
<tr>
<td>Real time (full screen high resolution)</td>
<td>(128 x 240 pixels) (9 bits/pixel; 30 frames/s)</td>
<td>8294</td>
<td>2000</td>
</tr>
<tr>
<td>Video</td>
<td>Non real-time low-resolution server</td>
<td>7603</td>
<td>384</td>
</tr>
<tr>
<td></td>
<td>(352 x 240 pixels) (9 bits/pixel; 10 frames/s)</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>VCR-quality server</td>
<td>60825</td>
<td>1100</td>
</tr>
<tr>
<td></td>
<td>(352 x 240 pixels) (24 bits/pixel; 30 frames/s)</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Studio-quality server</td>
<td>221184</td>
<td>4000</td>
</tr>
<tr>
<td></td>
<td>(640 x 480 pixels) (24 bits/pixel; 30 frames/s)</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>High-definition TV (HDTV)</td>
<td>800000</td>
<td>60000-127000</td>
</tr>
<tr>
<td></td>
<td>(1125 lines ; 24 bits/pixel; 30 frames/s)</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Table 2.2: Bit Rate Requirements of Various Video Formats.

Compression algorithms are classified as either lossless or lossy. In lossless coding, the original quantized sample values are recovered exactly in the absence of bit errors. They result in small compression ratios of around 10:1 to 50:1, and are used mainly for data applications. Conversely, lossy algorithms do not exactly replicate the original samples, leading to some degree of distortion; they, however, produce large compression ratios (around 50:1 to 200:1), and are ideal for video services that can tolerate some losses and some degradation in quality. The actual compression ratio used to encode video frames may be a function of several factors such as available bit rate in the network, the resolution required, and the degree of redundancy in the video bit stream.
2.5 Important Video Related ATM Parameters

There are various transfer parameters that are important to the applications that use ATM networks. They include bit error ratio (BER), cell loss ratio (CLR), cell insertion ratio (CIR), end-to-end cell transfer delay (CTD), and cell delay variation (CDV).

2.5.1 Bit Error Ratio

This is defined as the ratio of the bit errors in the information field to the total number of bits transmitted in the information field. BER in ATM networks is smaller than in previous networks due to the use of fiber technology in the transmission medium. Table 2.3 [15] presents the recommendations for the bit error rates of various BISDN services.

<table>
<thead>
<tr>
<th>Application</th>
<th>Bit Rate (Mbits/s)</th>
<th>BER*</th>
<th>BER~</th>
</tr>
</thead>
<tbody>
<tr>
<td>Videophone</td>
<td>2</td>
<td>$3 \times 10^{-11}$</td>
<td>$1.3 \times 10^{-6}$</td>
</tr>
<tr>
<td>Video Conference</td>
<td>5</td>
<td>$10^{-11}$</td>
<td>$1.8 \times 10^{-6}$</td>
</tr>
<tr>
<td>TV Distribution</td>
<td>20-50</td>
<td>$3 \times 10^{-13}$</td>
<td>$6 \times 10^{-7}$</td>
</tr>
<tr>
<td>MPEG-1</td>
<td>1.5</td>
<td>$4 \times 10^{-11}$</td>
<td>$2.5 \times 10^{-6}$</td>
</tr>
<tr>
<td>MPEG-2</td>
<td>10</td>
<td>$6 \times 10^{-12}$</td>
<td>$1.5 \times 10^{-6}$</td>
</tr>
</tbody>
</table>

* Without error handling.
~ Single bit error correction on cell basis.

Table 2.3: Recommended BER values for several B-ISDN applications.

2.5.2 Cell Loss Ratio

Cell loss ratio (CLR) is the ratio of the number of lost cells to the total number of cells sent within a specified interval. Various CLR objectives for a number of services are shown in Table 2.4 [55]. Due to the randomness of the traffic in ATM networks, it is possible that a cell arriving at a switching node may find the buffer full and be lost. Although this probability can be controlled to negligible values, there is always a non-zero probability of buffer overflow in ATM network.
Application | Bit Rate (Mbits/s) | CLR
---|---|---
Videophone | 2 | \(10^{-8}\)
Video Conference | 5 | \(4 \times 10^{-9}\)
TV Distribution | 20-50 | \(10^{-10}\)
MPEG-1 | 1.5 | \(10^{-8}\)
MPEG-2 | 10 | \(2 \times 10^{-9}\)

Table 2.4: CLR objectives for various VBR B-ISDN applications.

2.5.3 Cell Insertion Ratio

This is defined as the ratio of cells delivered to the wrong destination, to the total number of cells sent. Cells are delivered to a wrong destination when an error occurs at the header which is not detected by header error checking. If the new address in the header corresponds to the address of another connection, then the cell would be misrouted to a wrong destination. Cells that are misrouted may cause loss of terminal synchronization and also increase the traffic flow on links other than the links used by the connection to which they belong.

2.5.4 Delay

The cell transfer delay (CTD) between two network points is defined as the elapsed time from the first bit of a cell leaving the first point, to the last bit of the cell passing the second point. CTD in ATM networks is made up of the following components:

- Propagation Delay - this is due to the speed of light in the transmission medium, and depends on the distance between the source and the destination.
- Transmission Delay - this is the time needed for all the bits of a cell to arrive from the transmission link, and depends on the speed of the link.
Switching Delay - this is the total delay it takes for a cell to traverse an ATM switch. It depends on the internal switch speed and the amount of overhead added to the cell for routing within the switch.

Queuing Delay - this is the delay incurred by a cell when it has to be buffered before being transmitted by a switch.

In addition to CTD the following delays contribute to total end-to-end delay:

- Coding Delay - this is the time required to convert a non-digital signal to a digital bit pattern.
- Packetization Delay - this is the delay incurred while accumulating the required number of bits to form an ATM cell. It depends on the AAL used and the source bit rate.
- Reassembly Delay - this is the delay due to several cells of frame being collected at the receiver before they are passed up to the application.

Table 2.5 [14] gives average end-to-end delay values of some B-ISDN services.

### 2.5.5 Cell Delay Variation (Jitter)

This is defined as the variance of the cell transfer delay for a connection, due to random nature of the various delays outlined previously. Table 2.5 [14] gives some of the CDV objectives for a number of video services.

<table>
<thead>
<tr>
<th>Application</th>
<th>Average Delay (ms)</th>
<th>Jitter (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>64-kbits/s video conference</td>
<td>300</td>
<td>130</td>
</tr>
<tr>
<td>1.5 Mbits/s MPEG NTSC Video</td>
<td>5</td>
<td>6.5</td>
</tr>
<tr>
<td>20 Mbits/s HDTV Video</td>
<td>0.8</td>
<td>1</td>
</tr>
<tr>
<td>16 kbits/s compressed voice</td>
<td>30</td>
<td>130</td>
</tr>
</tbody>
</table>

Table 2.5: Delay and delay variation objectives for various services.
Jitter can be controlled at the receiver by storing the arriving cells in a jitter removal buffer, so that the departure rates of cells from the buffer is close to the inter-exit times of cells at the source.

2.6 Experimental results from ATM networks

Preliminary performance results of ATM networks have been mainly in terms of throughput and delay. Relatively little is known about the ability of ATM to support realistic application traffic patterns such as a mixture of CBR traffic, and time-sensitive bursty traffic such as VBR video. Real-time VBR traffic such as coded video, is considered one of the most difficult types of traffic to support. Research on the performance of ATM networks to support VBR sources has focused on either analytical models or simulation studies to predict performance. However, accurate analytical models are quite often intractable, and simulation cannot capture all aspects (or even most) of an actual distributed environment. This is due to the many components, hardware and software, whose complex interactions cannot be naturally captured or predicted by a fixed model. Hence actual measurements from ATM testbeds are necessary.

In the following we review results obtained from measurements in actual ATM networks:

In [7] the performance of JPEG, MPEG-1 and MPEG-2 coded video over a local ATM network was investigated. Transmission control protocol (TCP) and user datagram protocol (UDP) were used as the transport protocols. The performance in terms of delay (jitter) and frame loss, as a function of load, are presented and discussed.

The authors concluded:

- TCP performed better than UDP, and was able to efficiently support many multiplexed video streams at near peak steady state rates(approximately 23 Mbits/s). (TCP’s sliding window protocol effectively smoothed out traffic burstiness as well as delivered packets within their deadlines. It should be noted, however that UDP with flow control would perform better than TCP, especially over a wide area network where the delays incurred due to TCP’s sliding window would be prohibitive)
- UDP over ATM AAL5 is suitable for use with video only if the network is lightly loaded with video traffic. When 8 or more MPEG streams were multiplexed, unacceptable frame losses occurred (greater than 10%) when transmitting periodic burst streams.
- Controlling burstiness resulted in significantly less packet losses.
- Assuming frame delays less than or equal to 3 fixed frame intervals are tolerable, TCP can support up to 20 multiplexed streams with only 7% of the received intervals exceeding 3 fixed frame intervals.

As noted above due to the sliding window employed by TCP, if the transfer delay is too large (e.g. for wide area network (WAN) traffic), throughput would be adversely affected. For example, in [44] a WAN employing ATM switches with a 34 ms round-trip delay attained a maximum throughput of only 8.5 Mb/s (down from 51 Mb/s for inter-LAN TCP transmission). UDP throughput for both the WAN traffic and LAN traffic on the other hand was the same at 60 Mb/s.

In [8] the traffic characteristics of several applications in an ATM network are investigated. The goal was to study bursty traffic behavior, and also the effects of multiplexing VBR traffic. The traffic consisted of file transfer protocol (FTP), audio, and video applications. The peak rates for audio and video were determined, and found to be greater than 22 Mbits/s for video, and 3 Mbits/s for audio. It was found that the peak rate depended on the time intervals over which the data were gathered, the smaller the interval, the higher the peak rate due to the bursty nature of the VBR traffic. The multiplexing effects of applying FTP to transfer large files and small files was also investigated. It was found that the bandwidth requirement for multiple FTP connections did not scale as the number of connections increased. The effect of multiplexing small files was similar to that of having a sequence of FTP connections. This was due to the small ratio of the file size to the high link speed in an ATM network. The effect of multiplexing large files was found to be similar to that of having a single FTP connection for a concatenated file.
In [9] the performance of real-time and multimedia applications in an ATM environment was examined. The test measurements focused on the end-to-end user level performance metrics such as message throughput and round-trip delay, as well as video frame jitter under no-load and load conditions. The experiments revealed a number of performance bottlenecks and open issues in using ATM switches for practical applications. Their conclusions were:

- For end-to-end performance, the primary bottlenecks are in the protocol processing at layers above ATM, and the host operating system’s (UNIX) performance for burst data transfers.
- The current video-processing hardware and its integration with the host operating system are also severe limiting factors.
- ATM is limited for practical application and experimentation by the lack of suitable analysis tools.

In [10] a comprehensive study of the results obtained, between “mid-94 to end-95”, on the ATM Pilot Network is reported. The ATM Pilot network was the first European-wide VP cross-connected network. Two main traffic types were reported during the tests:

- CBR sources carrying voice or video applications
- VBR sources with multimedia applications or LAN to LAN interconnection.

The following performance measurements were obtained for the network:

Availability of Network Ratio : 97.1\% to 99.98\%.
Cell Error Ratio : $10^{-10}$ to $10^{-8}$.
Cell Loss Ratio : $5 \times 10^{-10}$ to $5 \times 10^{-9}$.
Cell Misinsertion Rate : 0.

They also characterized various parameters such as CDV, peak cell rate (PCR), and interarrival time as a function of time, for CBR sources, VBR bursty sources, and VBR non-bursty sources. Their results indicated that an accurate knowledge of the network traffic is required in order to set up the usage parameter control (UPC) parameter values.
An error in these parameter settings can cause ATM cells to be lost and thus cause a degraded QoS to be experienced by the user.

Finally, in [47] the authors investigate the modeling of CBR and VBR traffic, with respect to cell transfer delay (CTD). They found that CBR reference traffic at low utilization was adequately modeled by a gamma distribution. They also concluded that the CTD distribution at high utilization was the sum of multiple gamma distributions.
Chapter Three : Cell transfer delay measurements

3.1 Introduction

In this chapter we examine the effect of background load on the cell transfer delay (CTD) of CBR and UBR (unspecified) foreground traffic. We first look at the architecture of the ATM switch used in the experiments, we then examine the experimental setup, and finally we look at CTD under the following scenarios:

- CBR foreground traffic with CBR background traffic.
- CBR foreground traffic with UBR background traffic.
- UBR foreground traffic with CBR background traffic.
- UBR foreground traffic with UBR background traffic.

3.2 Fore Switch Architecture [18, 19, 20]

The ATM switch used for the experiments performed was the FORE Forerunner ASX-200WG. Figure 3.1 [19] shows the switch architecture whereby cells are processed in three steps. At the first step the input cells enter an ingress port, are sent through two stages of VPI/VCI translation, and are then checked for compliance with their traffic contract. The second step entails cell distribution using time division multiplexing (TDM) to multiple shared memory switch outputs. The TDM agent allocates specific time periods to transmit cells over the high speed common bus. The Fore switch architecture is non-blocking, in that the switching mechanism can handle the aggregate capacity of all input ports without possibility of cell loss; no input or internal buffering is required. Finally at the network module, cells are buffered in the shared memory in the case of output port congestion. The network module creates a queue per virtual channel (VC) and applies weighted-round-robin queuing and weighted fair queuing algorithms to manage the queues.
Figure 3.1: Fore Switch Architecture.

The Forerunner ASX200-WG switch that we used has 4 network modules, each module provides 4 input/output interfaces which share a 13312 cell output buffer. Since buffers are allocated dynamically, the entire 13312 cells of buffer space can be allocated to a single port.

3.2.1 Switch Traffic Policing

Traffic policing or usage parameter control (UPC), is a method of determining if cells entering the switch conform to pre-established traffic bandwidth contracts. Those cells that exceed the specified contract are “tagged” (as not conforming) or “dropped”. This ensures that non-conforming cells do not adversely affect conforming cells for other connections. The ASX-200WG utilizes a combination of “leaky bucket” or generic cell rate algorithm (GCRA) hardware in the switch fabric and user configurable parameters to perform these functions according to UNI 3.1 [13] specifications.
3.2.1.1 Leaky Bucket Algorithm

As mentioned in the previous chapter, leaky buckets are a mechanism to monitor cells entering the switch fabric to ensure UPC traffic contract compliance. The following parameters are monitored:

- Peak Cell Rate (PCR) - the maximum number of cells per second.
- Cell Delay Variation Tolerance (CDVT) - the tolerance for variation in the inter-arrival time of cells.
- Sustainable Cell Rate (SCR) - the average rate of cell transmission for this connection, taking bursts into account.
- Maximum Burst Size (MBS) - the maximum amount of cells that can be transmitted at the PCR.

The leaky bucket is a timing mechanism that measures the cells entering the switch fabric in terms of the parameters listed above. As a cell arrives, it determines whether the cell is early, on time, or late. If the cell is on time or late, the cell is allowed to pass. If the cell is early, however, the cell is either dropped or tagged. The switch uses a dual leaky bucket where the first bucket measures the PCR of the connection for compliance, whilst the CDVT is determined by the depth of the bucket. The second bucket measures the SCR or the rate of bucket drainage, and the MBS is determined by the depth of the second bucket.

In our experiments we used CBR and UBR traffic. For CBR traffic only the first leaky bucket is used to measure PCR and CDVT. All non-compliant cells are dropped. UBR traffic on the other hand is best effort traffic with no bandwidth guarantees provided, hence this type of traffic is not checked for contract compliance. Of course, the cost of setting up a CBR connection is expected to be significantly more expensive than that for a UBR connection.
3.3 Experimental Setup for cell-level measurements

The basic setup for cell-level measurements is shown in Figure 3.2, where the interWATCH 95000 real-time ATM traffic analyzer [48] captured and timestamped cells at the output port of the ATM network interface card (NIC) of the source workstation Saturn, which was a Sun UltraSparc I running at 143 Mhz. The cells were then forwarded by the analyzer to the ATM switch, where they were processed and sent out over a loopback connection, whereby an output port of the ATM switch was connected via OC3 fiber optic cable to one of the input ports of the ATM switch. The use of loopback connections facilitated the introduction of competing background traffic at the output port of the loopback, and also allowed the switch to emulate a network with multiple hops. Finally, the cells of the foreground traffic were then forwarded to another port of the ATM analyzer, where they were again captured and timestamped. They were then forwarded to the input port of the destination workstation Jupiter, another UltraSparc I
running at 143 Mhz. One hundred thousand cells were captured at both the source and destination workstations by the ATM analyzer. The analyzer was also used to introduce competing background traffic over the loopback connection as shown in the diagram. ATM connections are either pre-established using management functions or are set up dynamically on demand using signaling. When connections are pre-established they are referred to as permanent virtual circuits (PVCs); whereas connections set up dynamically are referred to as switched virtual circuits (SVCs). All our connections were set up using PVCs.

3.3 Cell Transfer Delay measurements for CBR foreground traffic and CBR background traffic

In this experiment, we investigated the effect of multiplexed CBR background traffic loads, on the cell transfer delay (CTD) for CBR foreground or reference traffic at a bandwidth of 450 kbits/s (the bandwidth and packet size corresponded to that of the base layer of the subband video described in the next chapter). The packet size was 1324 bytes. The CBR background load levels that we used are shown in Table 3.1, together with the corresponding bandwidths and number of multiplexed channels.

<table>
<thead>
<tr>
<th>% UTILIZATION</th>
<th>BANDWIDTH (MBITS/S)</th>
<th>NO OF CHANNELS</th>
</tr>
</thead>
<tbody>
<tr>
<td>30</td>
<td>44.8</td>
<td>3</td>
</tr>
<tr>
<td>50</td>
<td>74.9</td>
<td>5</td>
</tr>
<tr>
<td>70</td>
<td>104.8</td>
<td>7</td>
</tr>
<tr>
<td>90</td>
<td>134.8</td>
<td>9</td>
</tr>
<tr>
<td>95</td>
<td>142.3</td>
<td>10</td>
</tr>
<tr>
<td>97.5</td>
<td>146</td>
<td>11</td>
</tr>
</tbody>
</table>

Table 3.1: Background traffic description.
Figure 3.3 shows the CTD distribution with varying CBR background traffic load levels; from the graphs shown we see that as background load increases, the tail of the corresponding distribution grows. (It should be noted that the resolution of the timestamps taken by the ATM analyzer was to the nearest microsecond, leading to an error in each reading of between +1 us and -1us.) At no load the variation in CTD ranges over 10 us from 25 to 35 us, while at 97.5 % background load the goes from 25us all the way up to 155us. Since propagation delay introduced by the fiber optic cables can be assumed to be negligible, and no measurable delay is introduced by the ATM analyzer, we can assume that the delay variation is introduced by variable processing and queuing delays at the ATM switch. As more background load is introduced we can expect both the processing and queuing delays to increase, leading to the increased tail in the distribution shown. Figure 3.4 shows the variation in the mean and standard deviation values of the CTD as the background load is increased, here we see that the increase in these statistics are gradual as load is increased up to 90 % background utilization, but exponential increases take place in going from 90 to 95 % utilization and from 95 % to 97.5 % utilization. This indicates that results obtained for load levels greater than 90 % would probably be less predictable than for lower load levels.

Figure 3.3: CTD for CBR foreground traffic with CBR background (1 hop).
Figure 3.4: Mean and Standard Deviation for CBR foreground vs. CBR background utilization (1 hop).

Figure 3.5: CTD for CBR foreground Traffic (with CBR background) for varying number of hops.
### 3.3.1 Increasing the Number of hops

Figure 3.5 shows the comparison at 70% background utilization as the number of contention hops is increased from 1 to 3. In this case, the incoming background traffic was multicast to all outgoing loopback ports as shown in Figure 3.6 for the case of two loopback contention hops. This was meant to emulate a multi-hop network, where the foreground traffic would have contending traffic at each output port.

As would be expected the tail of the distribution increases with increasing number of hops, with a greater percentage of cells moving into the tail of the distributions as shown in Figure 3.5.

![Diagram showing experimental setup for the two hop case.](image)

**Figure 3.6:** Experimental setup for the two hop case.

Figure 3.7 shows how the mean and standard deviation increases with number of hops for 70% background utilization, the graph shows that the mean increases almost linearly with increasing no of hops, while the standard deviation also increases indicating a greater
spread in the distributions. This is to be expected since with more hops, the attendant processing delays, queuing delays and other distributed effects would also increase. The combined effect of the distributions at each hop would thus lead to an increase in both the mean and standard deviation.

![CTD mean and std for CBR foreground traffic with varying no. of hops (70% CBR bckgrd traffic)](image)

**Figure 3.7**: CTD mean and standard deviation vs. increasing number of hops (CBR traffic).

### 3.3.2 CBR foreground traffic with UBR background traffic

In this section, we examine the effect of unspecified bit rate (UBR) background traffic on CTD of CBR foreground traffic. We would expect the UBR traffic to have minimal effect on the CTD of the CBR foreground traffic, since UBR traffic is sent only when the switch has nothing scheduled to send. As in the previous experiment, the bandwidth of the foreground traffic was 450 kbits/s. The background traffic was produced by multiplexing
various combinations of the background channels shown in Table 3.2. The channels have high peak rates (close to the line rate) and much smaller average rates, hence the traffic produced was bursty in nature, in keeping with the fact that the switch (or network interface cards) would transmit UBR traffic at close to line rates once there was available bandwidth in the network.

<table>
<thead>
<tr>
<th>PEAK BIT RATE (MBITS/S)</th>
<th>SUSTAINABLE BIT RATE (MBITS/S)</th>
<th>MAXIMUM BURST SIZE (BITS)</th>
</tr>
</thead>
<tbody>
<tr>
<td>148</td>
<td>15.0</td>
<td>15300</td>
</tr>
<tr>
<td>148</td>
<td>7.5</td>
<td>15300</td>
</tr>
<tr>
<td>148</td>
<td>6.0</td>
<td>15300</td>
</tr>
</tbody>
</table>

**Table 3.2**: Traffic descriptors for UBR Background Traffic.

Figure 3.8 shows the distribution obtained for 1 hop with various background traffic load levels. Here we observe that as the background load increases, the tail of the distribution grows only slightly. This is probably due to increasing overhead as loading on the output port increases, and/or foreground traffic cells being delayed when they arrive and a background traffic cell is in the process of being transmitted. In any case, the UBR background traffic has much less effect than the corresponding CBR background traffic case.
Figure 3.8: CTD for CBR foreground, UBR background (1 hop)

Figure 3.9 shows a comparison of the distributions obtained with CBR foreground traffic for the cases of no load and 70% UBR background load (the distributions for 1 hop and 3 hops are both shown).

For the case of 1 hop we see that the distribution with the 70% UBR background traffic extends approximately 2 us beyond that for the no load case. A probable explanation for this is that the CBR foreground traffic is not affected by the UBR background traffic, unless a CBR cell is ready to be transmitted at the contention hop, and a UBR cell is in the process of being transmitted. In this case, we would expect the distribution for the UBR background traffic case to extend by at most 2.73 us beyond that of the no load case. Similarly for the case of 3 contention hops, we would expect the distribution for the UBR background case to extend by at most 8 us (2.73 x 3) beyond that of the no load 3 hop distribution. The graph shows that the distribution extends approximately 7 us beyond the no load case.
Figure 3.9: CTD for CBR foreground - No background vs. 70% UBR background traffic (1,3 hops).

3.4 Experiments with UBR foreground Traffic

In these experiments we tried to determine the effect of CBR and UBR background traffic on the CTD distribution of UBR foreground traffic. The ATM switch provides no guarantees in terms of delay or loss for UBR traffic, hence we would expect the CTD distribution in the case of UBR foreground traffic to be substantially different to the CTD distribution for CBR foreground traffic. Figure 3.10 shows the CTD distributions for CBR foreground and UBR foreground traffic over one hop, with no background traffic loading. The bandwidth used for the UBR foreground traffic was 1.4 Mbits/s, while that in the CBR foreground traffic case was 450 kbits/s. These bandwidths corresponded to the actual values used for the subband coded video enhancement and base layers respectively, as described in the following chapter. The packet size was 1324 bytes, also corresponding to the size of the packets using for the subband video. The results show that the two graphs do not differ significantly, with the UBR distribution roughly the equivalent of the CBR distribution shifted 2 us to the right. The small difference could be
due to the fact that the UBR traffic is bursty, with cells from the same packet being sent at approximately 150 Mbits/s; whereas CBR cells from a particular packet are sent equally spaced, with the time between cells inversely proportional to the bandwidth reserved (for the case of 531 kbits/s this time is approximately 800 us). Hence we conclude that with no competing traffic, the ATM switch treats UBR traffic roughly equivalent to the CBR traffic.

![CTD distribution comparison for CBR and UBR foreground traffic (1 hop)](image)

**Figure 3.10**: No load CTD comparison for CBR and UBR traffic (1 hop).

The results are quite different however, even at a relatively low load level of 30% CBR background traffic. Figure 3.11 illustrates the wide variation obtained for the CTD distribution in the case of UBR and CBR foreground traffic, with 30% CBR background traffic over one hop. In the case of the CBR foreground traffic the variation in the CTD is only about 10 us; while in the UBR foreground case the variation is over 400 us, and the CTD varies almost randomly over this interval. This clearly illustrates the difference in the treatment afforded to the two types of traffic by the ATM switch.
Figure 3.11: CTD comparison for UBR and CBR traffic with 30% CBR background traffic (1 hop).

Figure 3.12 shows the variation of the mean and standard deviation for UBR traffic (1 hop and 3 hops) with varying CBR background traffic.

Figure 3.12: CTD Mean and Standard Deviation for UBR foreground with CBR background (1,3 hops).
Here we see that these statistics increase exponentially with increasing background load, at 97.5% background utilization the mean delay is about 25 ms. In fact the largest delay observed was around 70 ms (for both cases of 1 hop and 3 hops), which is approximately 69.9 ms longer than the largest delay observed in the comparable CBR foreground traffic case. Also the number of hops had little effect on the mean and standard deviation. A possible reason for this is the fact that the load levels at the three hops are the same. Thus, the service rates received by the reference traffic are also about the same at the three hops. Hence, the burst of traffic arriving at the first hop would cause significant delay to the cells at the back of the burst. However, at the second and subsequent hops, the arrival rates and the service rates are roughly equivalent, so the queuing delay at these hops are much smaller.

![Mean and Standard Deviation for UBR foreground with UBR background](image)

*Figure 3.13: CTD Mean and Standard Deviation for UBR foreground with UBR background (1,3 hops)*.

Finally, Figure 3.13 shows the mean and standard deviation for the case of UBR foreground traffic with UBR background traffic, the maximum CTD in this case is a lot
less than for the case of CBR background traffic shown in Figure 3.12, with the maximum delay observed being around 2.3 ms.

However, there is still a lot of variation in the CTD as can be seen from the fact that the standard deviation is of the same order of magnitude as the mean. Also, unlike the CBR background traffic case, the mean and standard deviation increase with increasing number of hops.

3.5 Summary

The following conclusions can be stated regarding the results obtained in this chapter (it should be noted that these results apply only to the particular ATM switch used and under the conditions described in the chapter, under different conditions - e.g. different number of background channels, different background channel characteristics - different results may be obtained):

- The effect of CBR background traffic on CBR foreground cells, in terms of CTD, was very predictable except at very high utilization (> 90 % background traffic).
- The effect of UBR background traffic on CBR foreground cells was negligible.
- CBR background traffic had a very pronounced effect on the CTD distribution of UBR foreground cells, also the number of hops had little effect on the mean and standard deviation observed in this case. We suggested a possible explanation for this.
- The effect of UBR background traffic on UBR foreground cells, in terms of CTD, was significant but much less than that of CBR background traffic.
Chapter Four: Experiments with Subband Coded Layered Video

4.1 Introduction

Subband coding of video exploits the fact that different frequency subbands within video streams have varying levels of visual importance. Thus, lower frequency subbands which are more important to the human visual system can be given a higher priority. The subband video used in our experiments was obtained using the codec developed by E.Chang et al. [54]. It produces scalable video with a fine granularity of bit rates, and is of low enough complexity to permit real-time software implementation over existing workstations.

The coding technique consists of the following steps:

1. Three dimensional subband analysis on the video sequence to generate a set of spatio-temporal subbands. The approach adopted is to consider a classical spatial decomposition hierarchy and extend it to 3-D by applying temporal decomposition structures to each of the resulting spatial subbands.

2. Progressive quantization of each subband coefficient using embedded deadzone quantizers. This entails the application of a series of quantizer bins, where the number of bins increases with each quantizer application. The middle bin in each pass is called a deadzone because its quantized output is zero.

3. Hierarchical block coding of the quantized coefficients. This involves partitioning an image into 16 x 16 blocks. If a block contains all zeroes, the block is coded as a "0", and the next block is processed. Otherwise, the block begins with a "1", and the block is subdivided into four 8 x 8 blocks, each of which are coded the same way. Thus, the coding proceeds in a recursive manner down to 1 x 1 blocks. This method is applied to the first quantization layer only. To code higher layers the information from previous layers is used in order to avoid coding redundant bits. Specifically, any blocks that are marked "1" in the previous layer are also assumed to be "1" in the following layer.

4. Packetization of the resulting bit streams for transmission, separating the various layers, and putting headers to allow the network to route, and the decoder to combine the streams appropriately.
The decoder reverses the steps outlined above, it first removes the data from the packets, it then performs hierarchical block decoding and subband reconstruction. Finally, 3-D subband synthesis is performed in order to generate the video frames.

4.2 Experimental Setup

The experimental setup (Figure 4.1) was similar to the setup described in the previous chapter for cell level measurements, however, in this case packet level measurements were taken at the destination computer. As shown in the diagram, Saturn again served as the source, and Jupiter served as the destination computer.

Figure 4.1: Experimental Setup (1 hop).
Pre-encoded subband video was transmitted using permanent virtual circuits (PVCs) from Saturn to the ATM switch (using AAL5 as the ATM adaptation layer), over switch loopback(s), and finally to Jupiter. Figure 4.1 shows the setup with a single loopback, which we will refer to, as in the previous chapter, as a one hop scenario.

The subband video used consisted of 52 sublayers numbered 0 to 51, any subset of the 52 sublayers could be transmitted. The lower the sublayer number, the higher the priority of that sublayer - for example sublayer 0 has highest priority. Also, if sublayer n is absent, all higher numbered sublayers are useless. Each sublayer consisted of packets, each of size 1324 bytes, which constituted four frames of coded video. The sublayer packets were combined to form one group for playback.

The following video sequence was used:
- Mother - this shows a woman sitting with a little boy on her lap, she is saying something while gently stroking his hair. She occasionally looks at the boy, but most of the time she is looking directly at the camera. The resolution of the video was 352 x 240.

The video sequence consisted of 75 groups (300 frames), which at the frame rate used (25 frames/s) gives 12 seconds of video. In our experiments, however, we repeatedly transmitted the sequence until 10000 groups were transmitted. At a frame rate of 25 frames/s, each sublayer generated a bit rate of 75 kbits/s.

We partitioned the video to be transmitted into two layers, a higher priority base layer (consisting of sublayer 0 to sublayer n-1, to make up n sublayers) which was transmitted on one permanent virtual circuit (PVC) using the constant bit rate (CBR) traffic class with appropriate peak rate; and a lower priority enhancement layer (consisting of sublayer n to sublayer n+x-1, giving x total sublayers) which was transmitted on a separate PVC using the unspecified bit rate (UBR) traffic class. The PVCs implemented the usage parameter control functions as specified in UNI 3.1 [13].

On arrival of a packet at Jupiter, the number and the sublayer were recorded, together with a timestamp corresponding to the arrival time of the packet. For our experiments a base layer of 6 sublayers was chosen, this was the smallest base layer size possible using
the switch signaling (which was used for sending the video over the PVCs at the source, and receiving the video at the destination), which allocated bandwidth to CBR connections in increments of 531 kbits/s. Six sublayers corresponded to a bandwidth of 450 kbits/s, using a frame rate of 25 frames/s. The enhancement layer was chosen to be 19 sublayers, giving a total of 25 sublayers at a total bit rate of 1875 kbits/s.

In order to generate the background traffic, the InterWatch95000 analyzer was used, with the same traffic descriptors for the background load as detailed in the previous chapter.

4.3 Jitter Measurements

4.3.1 Base Layer Jitter

In order to measure base layer jitter, we used the inter-arrival times of the last base sublayer packet in each group, and subtracted 160 ms (corresponding to the ideal inter-arrival time). We varied background traffic load level and number of hops to gauge their effect on the jitter. The last base packet was used to measure the jitter since on receiving this packet, decoding could begin for the corresponding group. Figure 4.2 shows the results for group jitter for the case of 1 hop, for varying levels of CBR background traffic. As previously stated, a total of 10000 packets were used in the analysis. The results show that increasing the background traffic had very little effect on the distribution of group jitter over the +2 ms to -2 ms range shown in the graph. This is not surprising since in the last chapter we saw that increasing the CBR background traffic up to 97.5% only caused a 165 us maximum cell delay, this implies that no individual base sublayer packet will be delayed by more than 165 us due to increased background traffic. Hence for the 1 hop scenario we would expect little variation in jitter distribution.
Jitter Distribution for varying CBR Background Traffic Utilization (1 hop)

Figure 4.2: Base layer jitter distribution.

Similar results were obtained for the case of 2 hops (where the load is as described in the previous chapter, and the load is multicast to 2 output ports) and 3 hops, those graphs are shown in Appendix B, these results show that increasing number of hops in addition to increasing background load had little effect on the jitter distribution. The major contribution to delay and jitter for the base layer was not within the network but at the source and destination workstations. This delay consisted mainly of time to copy packets between buffers, interrupt calls, and packet segmentation and reassembly delay.

4.3.2 Enhancement Layer Inter-Arrival Time

In this section we will examine the enhancement layer inter-arrival time, as a measure of the effect of load on the enhancement layer in terms of jitter. Figure 4.3 shows the variation in average inter-arrival time (defined as the time between arrival of consecutively numbered packets belonging to the same group) for enhancement sublayer packets as CBR background traffic is increased for 1, 2, and 3 hops.
The results show that there is little increase in inter-arrival times up to 70% background traffic load level. However, there is an almost 4-fold increase in going from 70% to 95% utilization, and another 2-fold increase in going from 95% to 97.5% utilization. It should be noted however, that the increase in inter-arrival times with increasing load, has little effect on the quality of the received video. This is due to the large group inter-arrival time of 160ms; to receive the 19 enhancement sublayer packets with an average inter-arrival of 4 ms (in the worst case for 97.5% background load), would take on the order of 80 ms. Hence, provided that the delay experienced by any individual cell is less than about 80 ms (in the last chapter we saw that the maximum UBR foreground cell delay with CBR background traffic over was on the order of 70 ms), the last packet received would be the last packet in the base layer. There was little change in the inter-arrival time as the number of hops was increased, this is in agreement with cell level measurements from the previous chapter which showed that increasing the number of hops had little effect on the mean and standard deviation of the CTD for UBR foreground cells with CBR background traffic.
traffic. A typical sublayer packet arrival order for 97.5% background load over 3 hops was as follows:

0 1 2 6 7 8 9 10 11 12 13 14 15 16 17 18 19 4 20 21 22 23 24 5

Hence, the extra delay incurred due to even extremely high levels of CBR background traffic, did not affect the quality of the received video in terms of jitter.

In order to provide a measure of the effect of load on the relationship between the two streams, we decided to use the "distance" (difference) between the packet number of the last base layer packet in each group, and the packet number of the first enhancement layer packet that directly followed it. A negative value for this distance implied that a majority of time the base layer frames lagged behind the enhancement layer frames with the same packet number, while a positive value implied the opposite. Figure 4.4 shows that the distance varies very little with increasing load level, and is about -0.9 in magnitude, verifying that in the vast majority of instances that the last sublayer packet that arrived in any given group was indeed the final base sublayer packet.

**Figure 4.4**: Average Distance vs CBR background traffic load level.
Figure 4.5: Average Enhancement packet inter-arrival time vs. UBR background load.

Figure 4.5 shows the effect of increasing UBR background traffic on the enhancement layer packet inter-arrival times. Here we see that increasing the background traffic and number of hops has little effect on the inter-arrival times, which are on the order of the inter-arrival times for CBR background traffic at low load levels.

4.4 Base Layer Loss

Losses due to bit errors are expected to be very small in ATM networks (on the order of $10^{-10}$), hence when we examine loss we imply loss due to congestion (buffer overflow), or loss due to the usage parameter control (UPC) action of the ATM switch. Figure 4.6 shows the percentage of sublayer packets lost from the base layer, with increasing CBR background traffic for up to 3 hops.
Figure 4.6: Base Layer loss with increasing CBR background load level.

The graph shows that in all three cases, there was no loss for background traffic loads of up to 95% utilization. However, at 97.5% background utilization there were losses in the base layer, approximately 1% for 1 hop, 13.5% for 2 hops, and 22% for 3 hops. This is quite a surprising result since reservations were made for the base layer (after all the whole point in making reservations is to avoid loss and bound delay), and the total reservations made for both the base layer traffic and background traffic, was less than the total bandwidth available.

We suggest the following reason for the loss - the CBR background traffic affected the spacing between foreground traffic cells, so that some foreground cells no longer conformed to the spacing implied by the reservation level made (i.e. cell clumping occurred) at the switch, causing the UPC function at the switch to drop cells in the foreground traffic stream. We will investigate this further in the next section. No losses were observed in the base layer with UBR background traffic load levels of up to 107%.
4.4.1 Cell Delay Variation Measurements

The 1-point cell delay variation (CDV) as defined by the ATM forum [13] is as follows:

The 1-point CDV for cell \( k \), describes the variability in the pattern of cell arrival events observed at a single measurement point with reference to the connection’s peak rate (i.e. \( 1/T \)). Accordingly, the difference between the cell’s reference arrival time \( c_k \) and the actual arrival time \( a_k \) at a measurement point, is taken as a measure of its 1-point CDV \( y_k \) (i.e. \( y_k = c_k - a_k \)). The reference arrival time is defined as follows:

\[
c_0 = a_0 = 0 \\
c_k + T \quad \text{if } c_k \geq a_k \\
c_{k+1} = \begin{cases} 
    c_k + T & \text{if } c_k \geq a_k \\
    a_k + T & \text{otherwise}
\end{cases}
\]

Positive values of the 1-point CDV correspond to cell clumping, which implies cell inter-arrival times smaller than \( T \).

We used cell measurements at the output ATM switch port connected to the destination workstation in order to investigate the 1-point CDV at different CBR background traffic load levels, to investigate why the UPC function of the ATM switch dropped cells at 97.5% CBR background traffic load level. We used appropriate traffic descriptors at the switch such that no losses occurred. As in the previous chapter, 100000 cells were used in the analysis. Let us preface our results with a reminder that one of the parameters that can be specified when setting up a PVC connection is the cell delay variation tolerance (CDVT). When we set up our PVC connections, we did not specify a value for CDVT, hence the default value was used. Thus for our connections, a positive value of \( y_k \) as defined above, larger than the default value means that the UPC function of the switch would drop the non-conforming cell (a single cell loss would lead to the loss of the entire corresponding sublayer packet, each of which was 1324 bytes, using AAL5 to transmit the packets resulted in 28 cells per sublayer packet. Thus technically, a 3% cell loss could lead to a 100% packet loss if exactly one cell was lost per packet). Conversely, if \( y_k \) is
less than the default value for a particular cell, it can be said with 100% certainty, that it would not be dropped by the UPC function of the ATM switch due to cell clumping. Figure 4.7 shows the distribution of the 1-point CDV as CBR background traffic is increased for 1 hop. As the load level increases the graphs shift to the right, implying more positive values of \( y_k \), and more cell clumping.

![1-point CDV distribution with increasing CBR background traffic](image)

**Figure 4.7**: 1-Point CDV for various CBR background loads.

Figure 4.8 shows the 1-point CDV distributions for 97.5% background load for 1, 2, and 3 hops. As the number of hops are increased, the distribution shifts to the right and more cells have a larger value for \( y_k \), and as we saw in the results for the base sublayer packet results, greater cell and subsequent packet loss. Setting the CDVT to a value that is lower than the maximum that actually occurs at a particular load level, will cause cell loss to occur. This may be desirable in some applications such as videoconferencing, where loss might be preferred to jitter (in this case where we have only a few hops the delay is so small this would not apply), however if loss is to be avoided at all costs then a large value of CDVT should be specified. We verified that these losses were caused by UPC action by the switch by setting the CDVT for a 1 hop connection to 25 us, and observing that
with a background load of 97.5 %, a loss percentage of 99.5 % was observed in the base layer. On increasing the CDVT to 150 us, no losses were observed.

Figure 4.8: 1-Point CDV for 97.5 % CBR background load (1-3 hops)

Figure 4.9: Base Layer loss vs. CDVT.

Figure 4.9 shows the base layer packet loss against CDVT value for the case of 1 hop at 97.5 % CBR background traffic load level. From this graph we see that as the CDVT value is increased, the percentage of base layer packets lost decreases. No losses occur for
CDVT values above 125 us. A CDVT of 125 us is very small compared with the application level jitter, so it is well within the tolerance of the application. Hence, using a larger default value for the CDVT would be appropriate.

![Figure 4.10: Enhancement Layer loss vs. UBR background load level](image)

**Figure 4.10**: Enhancement Layer loss vs. UBR background load level

### 4.5 Enhancement Layer Loss

No losses were observed in the enhancement layer for CBR background traffic levels of up to 97.5%. Also no losses were observed in the enhancement layer when UBR background traffic of 98.3% or less. Losses were observed, however, when the total traffic load level exceeded 100% utilization, at 100% and 107% background traffic utilization. Figure 4.10 shows the loss levels observed with increasing UBR background traffic for 1, 2, and 3 hops. We note that as expected, as the number of hops increase, there is greater loss at 100% and 107% background traffic utilization. In the next section we will examine the effect of this loss on the quality of the received video.
4.6 **Loss Index**

In order to address the fact that the loss of a packet from sublayer n, caused all packets received from sublayers above n to be useless in decoding a particular group, we decided to define a *loss index* as follows:

The loss index of a group is the number of sublayer packets received without loss of a higher numbered sublayer packet.

For example, for a particular group, if 24 of the 25 packets were received, but sublayer packet 8 were missing, this would imply that the group has a loss index of 8.

The loss index provides a measure of the effect of loss (relative loss rather than absolute loss, as it accounts for the priority of the sublayer packet that is lost) on the quality of the received video. In this case, where a total of 25 sublayer packets were sent, an average loss index of 25 implied the received video is equal in quality to the video sent (i.e. no loss of packets occurred).

![Average Loss Index vs CBR background load level](image)

**Figure 4.11**: Average Loss Index vs. CBR background load level.
Figure 4.11 shows the average loss index (average for all 10000 group packets) for the case of CBR background traffic. There is no change in the loss index up to 95% background traffic, because there was no loss in the base or enhancements layers; at 97.5% utilization however, there is a large drop in the loss index for the cases of 2 hops and 3 hops. This corresponds to the losses that occurred in the base layer discussed previously. These losses lead to a big drop in the overall quality of the received video streams, as well as large variations in the quality of the video when it is played back.

\[\text{Average Loss index vs UBR background load level}\]

![Graph showing loss index vs UBR background load level](image)

**Figure 4.12**: Loss Index vs. UBR background traffic load level.

Finally, Figure 4.12 shows the average loss index with increasing UBR background load level; here we see that the quality of received video is perfect up to 98.3% background traffic. However, a large decrease in the loss index is observed when the background traffic is increased first to 100%, and then to 107%. This decrease is due to the losses in the enhancement layer discussed previously.
4.7 Quality of received video in terms of PSNR

In this section we compare the decoded subband video frames to the original uncompressed video frames, in terms of Peak Signal to Noise Ratio (PSNR). PSNR is a measure of the difference between the original image and the decoded image, measured in decibels (dB). It is calculated using the following formula:

\[
PSNR = 10 \log_{10} \left( \frac{1}{N} \sum_{x=0}^{N} \frac{255^2}{(I(x) - J(x))^2} \right)
\]

where \( I(x) \) is a pixel from the original image; \( J(x) \) is a pixel from the decoded image; and \( N \) is the number of pixels in the image.

Figure 4.13 shows a comparison in terms of PSNR, when all 25 layers are received without loss, compared to when loss occurs at 97.5% CBR background traffic over two hops. The first three hundred frames received for the sequence are shown, using only the luminance component of the image.

Here, loss occurred only in the base layer, hence there were wide fluctuations in the quality of received video. Also, any sublayer packet loss affected four frames, since there were four frames in a group. In practical terms, this would be very disturbing to the human visual system (HVS) and thus should be avoided at all costs. The possibility of having 97.5% CBR traffic at an ATM network node is probably extremely low, but loss can still occur at lower load levels when there are many hops in the network. Hence the load level and number of network hops should be taken into account and an appropriate CDVT specified (the upper bound on the CDVT would be the expected inter-arrival time of base layer cells).
Figure 4.13: PSNR Comparison - No loss vs. loss with CBR background traffic.

Figure 4.14 shows a similar comparison, but in this case loss occurred due to 100% UBR background traffic over 2 hops. Losses occurred only in the enhancement layer, and hence the fluctuations in PSNR were considerably smaller than in the CBR background case, implying less disturbance to the HVS. Also because no losses occurred in the base layer, there is a guarantee on the minimum number of sublayers of received video.
Finally, Figure 4.15 shows the PSNR comparison when 25 layers were sent with a CBR base layer (6 layers), and UBR enhancement layer, to when the same 25 layers were sent without any reservations (i.e. all layers sent UBR). The background load level was again 100 % UBR traffic over 2 hops.

Here we see that the base layer provides a guarantee on the minimum quality of the received video, whereas with no reservations, sometimes no layers were received.

![PSNR comparison - 25 layers, base(6) + enhancement vs no reservations](image)

**Figure 4.15 :** PSNR - No reservations vs. base layer reservations.

### 4.8 Summary

In this chapter, we examined the effect of UBR and CBR background traffic on base and enhancement layers for subband coded video in terms of packet jitter and loss. We found the following:

- Background load had little effect on the base layer (CBR) group inter-arrival times at the receiver.
- CBR background load caused a significant increase in the enhancement layer (UBR) packet inter-arrival times. However due to the fact that each group was sent over a 160 ms time period, this increase in inter-arrival times had negligible effect on the quality of the received video at the receiver.
• Loss was observed in the base layer for a CBR background of 97.5 % utilization. Examination of the 1-point CDV for the base layer cells indicated that this loss was probably due to UPC action on the part of the switch. Hence, appropriate setting of the CDVT parameter is an important factor in preventing loss at high load levels in an ATM network. No losses were observed in the base layer for UBR background load levels of up to 107 % utilization.

• No losses were observed in the enhancement layer for CBR background loads of up to 97.5 % utilization, losses were however observed for UBR background loads greater than 100 % utilization.

• Losses in the base layer had a very pronounced effect on the quality of the decoded video in terms of PSNR and should be avoided at all costs; losses in the enhancement layer had a smaller effect on the PSNR and thus could be tolerated.
Chapter Five : Experiments with MPEG-2 Layered Video

5.1 MPEG Video

The Moving Picture Experts Group (MPEG) [27,28,35,51] started its activities in 1988, and so far they have formulated two international standards, informally referred to as MPEG-1 and MPEG-2, with a third standard, MPEG-4, due in early 1999. These standards describe the coded representation of moving pictures, audio, and their combination.

5.1.2 MPEG-1 Standard

The MPEG-1 standard supports coding of video and associated audio at a bit rate of about 1.5 Mbits/s. It has three key components:

- Systems - this specifies how to combine one or more data streams form the video and audio with time information to form a single data stream.
- Video - this specifies the coded representation for compressed video sequences of up to 1.5 Mbits/s.
- Audio - this specifies the audio representation.

The MPEG-1 standard does not specify a particular design of video encoder. Instead, the syntax of the coded bitstream is specified, and a model decoder is described. Hence the developer is free to implement the details of the encoder, but for MPEG-1 standard compliance, the coded bitstream must follow the correct syntax, and must be compatible with the model decoder. Video frames are referred to as pictures, and there are three main picture types:

- I pictures (intrapictures) - these are intracoded (not coded with reference to another frame), and are points for random access, refreshing the frame sequence and preventing error propagation across frames. They can tolerate only moderate compression.
- P pictures (forward predicted pictures) - these are interframe (coded with reference to another frame) from the previous I or P pictures. They can be compressed more than I pictures.
- **B pictures** (bi-directionally predicted picture) - these are bi-directionally coded with respect to both previous and future I and P pictures. They provide the highest amount of compression.

The three pictures types are grouped together in group of pictures (GOPs), where a GOP consists of one I picture followed by a number of P and B pictures. An example of a GOP structure is shown in Figure 5.1, where each I or P picture is followed by two B pictures.

![GROUP OF PICTURES](image)

**Figure 5.1**: MPEG group of pictures structure.

The use of B pictures leads to a delay in the encoding and decoding processes, since both the previous and the next I/P picture must be received and stored before a B picture can be encoded or decoded. In order to minimize the delay at the decoder, the pictures are reordered by the encoder prior to transmission or storage, such that the I and/or P pictures required to decode each B picture are placed before the B picture. For example the following GOP sequence:

I₁ B₂ B₃ P₄ B₅ B₆ P₇ B₈ B₉ I₁₀

would be reordered as follows:

I₁ P₄ B₂ B₃ P₇ B₅ B₆ I₁₀ B₈ B₉,
5.1.3 MPEG-2 Standard

The MPEG-2 standard extends MPEG-1 in order to allow efficient encoding of video and associated audio at a wide range of resolutions and bit rates. It supports a wide range of applications for digital video storage media and digital video communications with various features such as constant bit rate transmission, variable bit rate transmission, random access, scalable decoding and high definition television (HDTV) standard formats. MPEG-2 has three main distinct characteristics from MPEG-1:

- Coding rate - from about 1.5 Mbits/s to in excess of 60 Mbits/s.
- HDTV format support.
- Hierarchical and scalable features.

The scalable features are as follows:

- Temporal scalable extension - this provides a migration path from lower temporal resolution systems. Video frames are partitioned into multiple layers, the lower layer provides the basic temporal resolution. The higher layer is coded with temporal prediction based on the lower layer. The full temporal resolution video is obtained by temporally multiplexing the decoded video data from the lower and higher layers.

- Spatial scalable extension - this is designed for video applications that support a minimum of two layers of spatial resolution. Two spatial resolution layers are generated from a single video source, the lower layer provides the basic spatial resolution, the higher (enhancement) layer contains the coded difference signal from a prediction based on the lower layer. The full spatial resolution is obtained at the higher layer by spatial interpolation based on the lower layer.

- Signal to noise ratio (SNR) scalable extension - this is designed for video applications that require a minimum of two layers with different video quality requirements. Two layers with the same spatial resolution are generated from a single video source. The
higher (enhancement) layer is coded to enhance the lower layer. A higher quality resolution video is obtained by adding the higher layer to the lower layer.

- Data partitioning extension - this entails the production of two streams of data. The first stream is used for the critical information and the second stream is used for the less critical information.

### 5.2 Layered MPEG-2 Video

To produce the layered video, we used MPEG2Tool [49], an MPEG-2 encoder toolkit developed at the University of Pennsylvania, which can produce SNR scalable video. The video sequence used consisted of a camera panning past a set of multicolored flowers, with a row of houses in the background. Encoding was difficult because some objects in the scene were moving faster than others. One hundred and forty-eight frames of the video were used. The video was in CCIR-601 format at a resolution of 720 x 486 pixels. Table 5.1 shows a comparison of the I, P, and B frame sizes for the Flower sequence for both the base and enhancement layers.

<table>
<thead>
<tr>
<th></th>
<th>Average size (bytes)</th>
<th>Standard deviation</th>
<th>Largest (bytes)</th>
</tr>
</thead>
<tbody>
<tr>
<td>I pictures</td>
<td>38556.3</td>
<td>2525.7</td>
<td>41055</td>
</tr>
<tr>
<td>P pictures</td>
<td>30537.2</td>
<td>7049.8</td>
<td>42028</td>
</tr>
<tr>
<td>B pictures</td>
<td>10833.4</td>
<td>2516.5</td>
<td>19136</td>
</tr>
<tr>
<td>All pictures</td>
<td>18277.3</td>
<td>11335.6</td>
<td>42028</td>
</tr>
</tbody>
</table>

**Flower Base Layer Picture Sizes.**

<table>
<thead>
<tr>
<th></th>
<th>Average size (bytes)</th>
<th>Standard deviation</th>
<th>Largest (bytes)</th>
</tr>
</thead>
<tbody>
<tr>
<td>I pictures</td>
<td>103459.4</td>
<td>6914.2</td>
<td>112749</td>
</tr>
<tr>
<td>P pictures</td>
<td>81123.4</td>
<td>5403.6</td>
<td>90987</td>
</tr>
<tr>
<td>B pictures</td>
<td>64808.2</td>
<td>5166.1</td>
<td>78614</td>
</tr>
<tr>
<td>All pictures</td>
<td>72341.4</td>
<td>13080.6</td>
<td>112749</td>
</tr>
</tbody>
</table>

**Flower Enhancement Layer Picture Sizes.**

Table 5.1: Picture size comparison for Flower sequence.
Due to the fact that I, P, and B pictures can be compressed to differing degrees, if we made CBR reservations based on the size of the largest frame, most of the other frames would be significantly smaller and a lot of bandwidth would be wasted. ATM networks can compensate for this variation in bit rate through the use of the VBR traffic class, where the peak cell rate, average cell rate, and maximum burst size are specified when making a reservation. This allows the switch to allocate less resources than for a CBR connection with the same peak cell rate. Unfortunately the ATM switch implemented UNI 3.1, which only defines a non real-time VBR traffic class (the latest version, specification UNI 4.0 also specifies a real-time VBR traffic class, which would be more appropriate for video). Hence when making the reservations for the MPEG-2 base layer we had two choices; use the CBR traffic class or use the non real-time VBR traffic class. We decided to go with the former choice due to the fact that video is real-time, and the results in terms of delay for real-time VBR (when it is implemented) could be expected to be similar to the results for CBR (which is also real-time), contingent on the selection of appropriate traffic descriptors for the real-time VBR case.

The GOP structure used was as follows:

\textit{IBBPBBPBBPBB}

Therefore we had a larger I or P frame, followed by two smaller B frames. We chose the reservation level based upon the largest group of IBB or PBB frames in the entire sequence. What this implied was that there would be some overlap of the time to send the I or P frame into the time period to send the following B frames. For example using a frame rate of 25 frames/s, there are 40 ms between frames, the effect of choosing our reservation levels was to allow the larger P or I frame to take more than 40 ms to be sent as shown in Figure 5.2, but still ensure that the largest set of three frames took at most 120 ms to send. The reservation level obtained for the base layer using this scheme was 5.6 Mbits/s. The advantage of our reservation scheme was that the peaks caused by the I and P frames were smoothed to some extent. The disadvantage of our scheme was that we needed to buffer at least three frames before playback at the decoder.
Because we expected little or no losses in the base layer, and the size of the largest base picture was smaller than the maximum AAL5 packet size of 65535 bytes, we used the base layer packet size equal to the picture size. The enhancement layer was sent best effort, and because some losses could be expected here, we used 10 packets for each picture so that the loss of one cell would not cause the entire frame to be lost. The average bit rate of the base layer was 3.84 Mbits/s, while the average bit rate of the enhancement layer was 13.74 Mbits/s. At the beginning of each 40 ms time period, the base packet was sent followed immediately by the enhancement layer packets.

5.3 Results with CBR Background Traffic

5.3.1 Base Layer Inter-arrival Times

Figure 5.3 shows the results for the cumulative probability, that base layer frames arrive a given time, after their reference arrival time. Twenty thousand frames were used in the analysis by repeatedly transmitted the same 148 frames, and the setup was similar to the one described in the previous chapter for subband video.
The results show that an increase in load had little effect on the inter-arrival time distribution of the packets. In this case because the frames are of different sizes, then the theoretical arrivals times vary with each frame; the percentage of frames that arrive after multiples of the inverse of the frame rate, 40 ms (i.e. at 40, 80 ms etc.) is a useful parameter. It helps in determining how much buffer space is needed, and also how much time to wait initially before starting to decode the video. Figure 5.3 shows that frames arrive at most 120 ms after their reference arrival time, hence it would be reasonable to buffer 3 frames before initiating decoding and playback of the base layer. This ensures that preceding frames would always be available for playback while waiting for a particular frame to arrive. The results shown are for 1 hop no load; and also for 93% CBR background load over 3 hops. The two graphs are indistinguishable, illustrating that the high load level and increased number of hops had little effect on base layer packet jitter.

5.3.2 Enhancement Layer Inter-arrival Times

Figure 5.4 shows the effect of load on the enhancement layer inter-arrival times. In this case the packets vary in size, and were much larger than in the previous chapter. The average enhancement layer packet inter-arrival time should be small enough to ensure
synchronization between the base and enhancement streams. As shown in the graph the inter-arrival times are sometimes greater than 10 ms at higher load levels. As we saw in Chapter 2, at high load levels, the cell delay in the ATM network can be very large, which may lead to a loss in synchronization between the two streams. This loss in synchronization could lead to the enhancement layer being useless (unless a large amount of buffering was used).

![Figure 5.4: Enhancement layer packet average inter-arrival times vs. CBR background load.](image)

In order to provide a measure of the effect of load on the relationship between the two streams, we decided to use the "distance" (as defined in Section 4.3.2) between the packet number of the base layer packet, and the packet number of the first enhancement layer packet that directly followed it. As in the previous chapter a negative value for this distance implied that a majority of time the base layer frames lagged behind the enhancement layer frames with the same packet number, while a positive value implied the opposite.

Figure 5.5 shows the distance statistics at increasing CBR background traffic load levels. At low load levels the values for the mean are small (less than one in magnitude) and negative, indicating that the majority of time the base layer packet arrived after the enhancement layer packets. While at higher load levels the distance is positive and increases with load and number of hops.
At a background load level of around 73% the difference changes from a negative value to a positive value, and at a load level of 93% can be as high as 20. What this means is that although enhancement packets are received at higher load levels, they may be useless due to the delay they experience in the network.

**5.3.3 Enhancement Layer Loss**

Figure 5.6 shows the percentage of enhancement layer packets lost, with increasing load over 1 to 3 hops. The total bandwidth of the reference MPEG-2 video was just less than 12% of the available 149.7 Mbits/s bandwidth, and as shown in the figure, losses generally began occurring at about 88% background traffic where the total load becomes 100%. This indicated that once the total utilization is less than 100% very little or no loss occurs; this is similar to the results obtained in the previous chapter for subband video. In this case however, our enhancement layer had a much higher bandwidth and burst size. The low probability of loss at less than 100% utilization points to the ATM switch having an effective buffering scheme.
Figure 5.6: Enhancement layer loss vs. CBR background traffic load level.

5.4 Results with UBR background traffic

5.4.1 Enhancement Layer Inter-arrival times

Figure 5.7 shows the effect of UBR background traffic on the inter-arrival times (defined as the time between arrivals of consecutively numbered packets belonging to the same frame) of the enhancement layer packets. If we compare this graph with the one obtained for CBR background traffic (Figure 5.4) we note that the results are not significantly different except at the highest load levels. In the previous chapter we found that with smaller packets and a smaller enhancement layer bit rate, the enhancement layer inter-arrival times varied significantly using UBR background traffic, when compared to using CBR background traffic.
One possible reason for the different results in this case is that with a larger number of cells per packet, the probability of a packet being delayed is increased (since if one cell is delayed all the following cells from the same packet would also be delayed to the same extent, and the packet delay is determined by the maximum cell delay), thus leading to larger inter-arrival times. We did not make cell level measurements because the analyzer was unable to capture UBR traffic at these high bit rates. Hence, we were unable to verify this hypothesis.

Figure 5.8 shows the distance mean and standard deviation as UBR background load is increased. There is a large increase in distance in going from 86% background traffic to 90% background traffic, this coincides with going into "overload" (over 100% utilization), and also the initiation of packets being lost. Hence if we are not going to buffer the appropriate amount of packets, so that the enhancement layer would be useful at high utilization levels, a good rule of thumb is to check if packets are being consistently lost, and if they are, then ignore the enhancement layer.
5.4.2 Enhancement Layer Loss

Figure 5.9 shows the effect of UBR background traffic on the enhancement packets in terms of loss.

As previously stated, the bandwidth of the foreground traffic was approximately 17.5 Mbits/s, which is about 12% of the link capacity. As in the CBR case there was no loss at total load levels of less than 100%, but there was a precipitous increase once the 100% utilization threshold is crossed in going from 86% background load to 90% background
load. We also observe that there was an increase in the number of packets lost as the number of hops was increased. As we shall see in next section, the loss of enhancement packets has a very pronounced effect on the quality of the decoded video.

### 5.5 PSNR of decoded video

Figure 5.10 shows the PSNR values for the base layer, base and enhancement layers when both are received without loss, and also base and enhancement layers when there was 27% loss in the enhancement layer for the case of 1 hop 90% UBR background load. The PSNR for the first 148 reconstructed frames are shown, these frames were decoded offline using the MPEG-2 decoder developed by the Software Simulation Group (SSG) [50]. The luminance components from both the original raw video and the reconstructed decoded video were used to calculate the PSNR.

![PSNR for first 148 frames](image_url)

Figure 5.10: PSNR for various scenarios.

The graph shows that the quality of the decoded video is extremely low when there are losses in the enhancement layer, hence as previously suggested it would be better to monitor if enhancement layer loss occurs at the decoder, and decode only the base layer if loss is observed. The reasons for the low quality of the video were twofold; firstly, when enhancement layer packets were lost, the decoder was unable to combine the base and enhancement frames correctly because headers essentials to their proper recombination were missing. Secondly, when using I and P frames as reference for other frames, these
reference I and P frames were reconstructed using both the base and enhancement layers. Hence if these frames contained errors, they would propagate to other frames. The average PSNR for the base layer only was 25.95 dB, 38.42 dB for the base and enhancement with no loss, and 14.43 dB for the base and enhancement layers when there were losses in the enhancement layer.

5.6 Summary

In this chapter we examined the effectiveness of transmitting SNR scalable MPEG-2 video using a CBR base layer, and a UBR enhancement layer. The total bit rate of the combined layers was approximately 18 Mbits/s, which was much higher than that for the subband video examined in the previous chapter. We made the following observations:

- Background load level had no significant effect on the base layer in terms of packet inter-arrival time, and hence would not effect the buffer space needed at the decoder.
- High levels of background load had a significant effect on the “distance” between the base and enhancement layers. For CBR background traffic the average distance was as large as 20, whilst for UBR background the average distance was on the order of 6. What these results implied was that the background load expected in the network would have an effect on the buffer space needed at the decoder, and also how the decoder should wait for the enhancement layer packets before assuming they had been lost.
- No losses were observed in the base layer.
- Losses were observed in the enhancement layer, but only when the total utilization exceeded 100%.
- Losses in the enhancement layer had a disastrous effect on the quality of the decoded video. This suggested that if loss of enhancement layer packets was observed at the decoder, then it would be more effective to decode the base layer only for that group of pictures.
Chapter Six : Conclusions

6.1 Discussion

The primary goal of this thesis was to investigate the feasibility of transmitting layered video over a local ATM network. Two layered video schemes were looked at, namely subband coded layered video and MPEG-2 SNR scalable layered video. In both cases the video was transmitted as a CBR base layer and a UBR enhancement layer, we then introduced background traffic (both CBR and UBR) in order to determine their effect on the layered video in terms delay, jitter and loss. Because no reservations were made for the enhancement layer, this allowed the video to be coded to a constant quality without close consideration of what the final bit rate would be, however without reservations, there were no delay or loss guarantees for the enhancement layer.

The use of the base layer, which did have quality of service guarantees, maintained a minimum quality for the received video. Also the fact that UBR enhancement layer cells were sent in bursts at line rates helped to compensate for the lack of delay guarantees, since ATM cells from the CBR base layer were staggered over the period to send a frame the enhancement layer could potentially arrive before the base layer.

The results obtained for the subband coded video (base layer at 450 kbits/s, enhancement at 1.35 Mbits/s) were very positive. We found that background traffic load level had no apparent effect on base layer packet jitter at the receiver. Also the enhancement layer arrived before the base layer even at the maximum CBR background traffic load level examined, and hence would introduce no extra delay at the decoder. We found out that losses could occur in the base layer due to usage parameter control (UPC) action on the part of the switch, at very high CBR background traffic load levels. This could be prevented by appropriate selection of the CDVT parameter when setting up connections. Losses in the enhancement layer only occurred when the total utilization (at any output port along the route) exceeded 100 %.

The effect of losses in the base layer had a pronounced effect on the quality of the received video in terms of peak signal to noise ratio (PSNR), with large fluctuations observed, and hence base layer loss should be avoided at all costs. Enhancement layer
loss had a less pronounced effect on the quality of the received video, and hence whatever is received (base + enhancement) could be decoded without excessive disturbance to the human visual system.

The results for the much higher bit rate MPEG-2 video (base layer 3.84 Mbits/s, enhancement 13.74 Mbits/s) were also quite good. As for the subband video we found that background load level had no apparent effect on the base layer in terms of jitter. However due to the much higher bit rates, synchronization between the two streams was more difficult to maintain than for the subband coded video. The results were still quite good however, with synchronization (enhancement layer arriving first or at the same time as the base layer) between the two layers for total utilization on the order of 90%. Losses in the enhancement layer only occurred when the total utilization exceeded 100%; these losses were extremely detrimental to the quality of the decoded video, hence, a good rule of thumb would be to check if any losses were occurring in the enhancement layer, and, if so, decode only the base layer for that GOP.

6.2 Suggestions for further work

Some suggestions for further work are as follows:

- The ATM switch that we used currently implements UNI 3.1, which does not support a real-time VBR traffic class. If the switch was upgraded to UNI 4.0, which does support a real-time VBR traffic class, then either the base or the enhancement layer could be sent using real-time VBR reservations.
- The network interface cards (NICs) that we used did not allow the simultaneous setup of two or more CBR connections, this facility would be quite useful if, for example, we wanted to send spatial scalable video using CBR connections for both streams.
- In our experiments, we used at most, two independent streams to produce the background traffic. A more realistic scenario would involve the use of many more independent background streams to provide the background traffic.
Appendix A - List of Acronyms

AAL - ATM Adaptation Layer.
ABR - Available Bit Rate.
ATM - Asynchronous Transfer Mode.
BER - Bit Error Ratio.
B-ISDN - Broadband Integrated Services Digital Networks.
BT - Burst Tolerance.
CBR - Constant Bit Rate.
CDV - Cell Delay Variation.
CDVT - Cell Delay Variation Tolerance.
CIR - Cell Insertion Ratio.
CLR - Cell Loss Ratio.
CPCS - Common Part Convergence Sublayer.
CS - Common Sublayer.
CTD - Cell Transfer Delay.
FTP - File Transfer Protocol.
GCRA - Generic Cell Rate Algorithm.
GFC - Generic Flow Control.
GOP - Group Of Pictures.
HDTV - High Definition TeleVision.
HEC- Header Error Check.
HVS - Human Visual System.
ITU-T - International Telecommunications Union -Telecommunications standardization sector.
JPEG - Joint Photographic Experts Group.
LAN - Local Area Network.
LSC - Layered Source Coding.
MBS - Maximum Burst Size.
MPEG - Moving Pictures Expert Group.
NIC - Network Interface Card.
PCR - Peak Cell Rate.
PSNR - Peak Signal to Noise Ratio.
PT - Payload Type.
PVC - Permanent Virtual Circuit.
QoS - Quality of Service.
SAR - Segmentation And Reassembly.
SCR - Sustainable Cell Rate.
SNR - Signal to Noise Ratio.
SSCS - Service-Specific Convergence Sublayer.
SSG - Software Simulation Group.
SVC - Switched Virtual Circuit.
TCP - Transmission Control Protocol.
TDM - Time Division Multiplexing.
UBR - Unspecified Bit Rate.
UDP - User Datagram Protocol.
UNI - User Network Interface.
UPC - Usage Parameter Control.
VBR - Variable Bit Rate.
VC - Virtual Circuit.
VCC - Virtual Circuit Connection.
VCI - Virtual Circuit Identifier.
VOD - Video On Demand.
VP - Virtual Path.
VPC - Virtual Path Connection.
VPI - Virtual Path Identifier.
WAN - Wide Area Network.
Appendix B - Jitter Distribution for Base Layer Subband Video

Figure B.1 Jitter distribution for varying CBR background traffic load level (2 hops)

Figure B.2 Jitter distribution for varying background traffic load level (3 hops).
Appendix C - Review of layered coding implementations

In the following we review some layered coding methods put forward in the literature:

In [4] a method of two-layer coding of VBR video signals is proposed by Ghanbari. The first layer codes important structural information in the image, to be accommodated in the guaranteed capacity. The second layer provides the additional information necessary for a quality finish. The performance of the coder was tested with common interchange format (CIF) standard sequences and broadcast-quality pictures. The portion of the VBR channel allocated to the lower layer as guaranteed bandwidth was examined; it was found that the coder performed well for a guaranteed channel rate as low as 10-20% of the total bit rate.

In [53] a two-layer coding scheme is proposed based on reducing temporal redundancy by using the background information of the image sequence. The scheme is described as follows, while coding the I-frame, a classifier is designated to detect the background region. The cells representing them are placed in the enhancement layer; on the other hand, blocks recognized as motion are placed into the base layer. The P- and B-frames are not modified in any way. The authors provide experimental results comparing their scheme to the "traditional" two-layer coding method (based on essential information in the base layer, and high frequency and edges of the picture in the enhancement layer). Their scheme outperformed the traditional one in terms of PSNR.

In [2] a layered coding technique using discrete cosine transform (DCT) coding is described for ATM networks. The technique involved separation of the DCT coefficients into most significant parts (MSPs) and least significant parts (LSPs). MSPs consist of the lower spatial frequency coefficients, while LSPs consist of the higher frequency components. The authors found, via computer simulation, that the influence on picture quality by cell loss can be reduced by using the layered coding technique.

In [5] Luo and Zarki propose an adaptive data partitioning (DP) for use with the leaky bucket (LB) algorithm. It involves a two layer scheme with a high priority (HP) and low
priority (LP) being generated. The authors measure their scheme by following two criteria:

- Whether the HP layer constitutes a high enough visual quality image.
- Whether the transmission of the HP layer can be guaranteed by a UPC such as the LB in a simple and efficient manner.

They propose an adaptive DP scheme based on the analysis of MPEG's compression limit, its traffic characteristics, and the psychophysical properties of the human visual system. Their scheme generates results that provides an "acceptable compromise between high visual quality and low bit rate requirements during times of long bursts".

In [6] the same authors propose two prioritized transmission (PT) schemes, namely coupled-concatenated transmission (CCT) and coupled interlaced transmission (CIT). These are extensions of the adaptive DP technique. CCT involves transmitting the HP cells of a frame first, followed by the LP cells, while CIT involves the implementation of an intelligent self-tagging scheme for the LP cells. The results obtained, via computer simulation, indicated that both schemes outperform non-layered coding in terms of SNR, with CIT giving the best performance of the three.

In [43] Taubman and Zakhor present a video compression strategy, based on 3-D subband coding. The scheme is multiresolution, whereby an encoding bit stream provides some measure of bit rate scalability through the selection of different display resolutions. It is also multirate, which allows for extraction of subsets of the bit stream, covering a range of several tens of kbits/s through several Mbits/s, with a fine gradation of available data rates through this range. This allows for transmission through heterogeneous networks with many decoder sizes, maximum frame rates, and varying bandwidth requirements throughout the network. The system is based on 3-D subband coding, progressive quantization of the subband coefficients, followed by layered arithmetic coding. The relatively simple traffic properties for the layers produced allowed for simple network control with layers or groups of layers, while allowing VBR by varying the number of layers transmitted. Their scheme outperformed MPEG-1 (in terms of SNR) during
regions when the camera was still or panning, during camera zoom sequences, however, MPEG-1 outperformed their scheme.

In [22] another layered coding technique based on subband coding is presented. It involved the encoding of medical images for real-time transmission over heterogeneous networks. The method consisted of performing 3-D subband analysis, followed by progressive quantization using embedded deadzone quantizers, then by hierarchical block coding. The authors compared their scheme to MPEG-1 at various bit rates, they found that their scaleable encoder achieved comparable quality (in terms of PSNR) to MPEG-1. Their encoder was much faster than MPEG-1 encoding, while their decoder was slower than MPEG-1 decoding but still ran in real time. The authors conclude that for software implementations of real-time applications involving both encoding and decoding, such as video conferencing, their scaleable coding technique is more promising than MPEG-1.

Finally, in [52] an “intelligent” packetizing scheme in which MPEG coded video data is split into separate streams of ATM cells according to the picture coding type of each frame is proposed. The encoded I-frames within the data are placed in one stream, P-frames are placed in a second stream and B-frames in a third stream. The separate streams are transmitted over different VCCs in the ATM network. The results obtained via simulation using this technique indicated that the B-frames had the highest tolerance to loss, and the I-frames the lowest, in terms of both PSNR and subjective quality assessments. The authors concluded that instead of providing a single, low cell loss rate for all the data, the B-frames could accept a higher cell loss rate than the I- and P-frame streams. Alternatively, cells within the B-frame stream could be “tagged” as low-priority data which could be dropped without severely affecting the quality of the decoded sequence. They also found that the three separate streams were each significantly less bursty than combined MPEG data. This means that less network capacity needs to be reserved for the separate streams than for the combined data.
References


