National Library
of Canada

Bibliothèque nationale
du Canada

Acquisitions and
Bibliographic Services Branch

Direction des acquisitions et
des services bibliographiques

395 Wellington Street
Ottawa, Ontario
K1A 0N4

395, rue Wellington
Ottawa (Ontario)
K1A 0N4

Your file - Votre référence

Our file - Notre référence

NOTICE

The quality of this microform is heavily dependent upon the quality of the original thesis submitted for microfilming. Every effort has been made to ensure the highest quality of reproduction possible.

If pages are missing, contact the university which granted the degree.

Some pages may have indistinct print especially if the original pages were typed with a poor typewriter ribbon or if the university sent us an inferior photocopy.

Reproduction in full or in part of this microform is governed by the Canadian Copyright Act, R.S.C. 1970, c. C-30, and subsequent amendments.

AVIS

La qualité de cette microforme dépend grandement de la qualité de la thèse soumise au microfilmage. Nous avons tout fait pour assurer une qualité supérieure de reproduction.

S'il manque des pages, veuillez communiquer avec l'université qui a conféré le grade.

La qualité d'impression de certaines pages peut laisser à désirer, surtout si les pages originales ont été dactylographiées à l'aide d'un ruban usé ou si l'université nous a fait parvenir une photocopie de qualité inférieure.

La reproduction, même partielle, de cette microforme est soumise à la Loi canadienne sur le droit d'auteur, SRC 1970, c. C-30, et ses amendements subséquents.

Canada

FEC and priority for an ATM Adaptation Layer protocol for VBR video distribution

by

**K. Anand, B. Tech.
Indian Institute of Technology,
Bombay, India**

**A thesis submitted to the
Faculty of Graduate Studies and Research
in partial fulfillment of the requirements
for the degree of
Master of Engineering**

**Ottawa-Carleton Institute for Electrical Engineering
Faculty of Engineering
Department of Systems and Computer Engineering
Carleton University
Ottawa, Ontario, Canada**

August 26, 1992

© K. Anand, 1992

National Library
of Canada

Acquisitions and
Bibliographic Services Branch

395 Wellington Street
Ottawa, Ontario
K1A 0N4

Bibliothèque nationale
du Canada

Direction des acquisitions et
des services bibliographiques

395, rue Wellington
Ottawa (Ontario)
K1A 0N4

Your file Votre référence

Our file Notre référence

The author has granted an irrevocable non-exclusive licence allowing the National Library of Canada to reproduce, loan, distribute or sell copies of his/her thesis by any means and in any form or format, making this thesis available to interested persons.

L'auteur a accordé une licence irrévocable et non exclusive permettant à la Bibliothèque nationale du Canada de reproduire, prêter, distribuer ou vendre des copies de sa thèse de quelque manière et sous quelque forme que ce soit pour mettre des exemplaires de cette thèse à la disposition des personnes intéressées.

The author retains ownership of the copyright in his/her thesis. Neither the thesis nor substantial extracts from it may be printed or otherwise reproduced without his/her permission.

L'auteur conserve la propriété du droit d'auteur qui protège sa thèse. Ni la thèse ni des extraits substantiels de celle-ci ne doivent être imprimés ou autrement reproduits sans son autorisation.

ISBN 0-315-84135-4

Canada

**"FEC and priority for an ATM Adaptation
Layer protocol for VBR video distribution"**

submitted by

**K. Anand,
B. Tech.**

**in partial fulfillment of the requirements
for the degree of Master of Engineering**

Thesis Supervisor

Thesis Supervisor

**Chairman,
Department of Systems
and Computer Engineering**

**Carleton University
August 26, 1992**

Abstract

A major advantage of ATM based B-ISDN networks is their ability to support variable bit rate (VBR) video coding, resulting in consistent picture quality, bandwidth savings and delay reduction. However, one of the major drawbacks of ATM networks is the possibility of consecutive cell losses due to network congestion and the consequent degradation in picture quality. This thesis proposes an ATM Adaptation Layer (AAL) protocol for matching a VBR video source to an ATM distribution network. The VBR source emits data of two levels of priority. The protocol includes a cell loss compensation scheme using a combination of Single Parity Check (SPC) and Reed Solomon (RS) codes for the high priority VBR data. The performance of the compensation scheme is evaluated by analyzing the process of statistical multiplexing of multiple video sources. Both analytic and simulation approaches are adopted to estimate the performance of the compensation scheme as well as that of the priority handling scheme.

Acknowledgments

I would like to take this opportunity to sincerely thank my supervisor, Dr. A. R. Kaye, for his guidance and encouragement throughout the duration of my thesis. This thesis would probably not have seen the light of the day, had it not been for his enthusiasm and interest in the whole project.

I gratefully acknowledge the funding support provided by Carleton University, Telecommunications Research Institute of Ontario (TRIO), and National Sciences and Engineering Research Council (NSERC).

I would also particularly like to thank Prof. T.A. Gulliver for his constant guidance and extremely useful suggestions regarding the coding theory part of the thesis, and Prof. S.A. Mahmoud for co-supervising this thesis.

To My Parents

Table of Contents

Abstract	iii
Acknowledgments	iv
List of Figures	ix
List of Acronyms	xi
 Chapter 1 Introduction	 1
1.1 Background	1
1.2 Thesis Objective	3
1.3 Thesis Organization	5
 Chapter 2 Video Source Coding Schemes	 7
2.1 Overview	7
2.2 Layered Video Coding Schemes	7
2.3 VBR source coding scheme used in the thesis	10
 Chapter 3 ATM based B-ISDN	 15
3.1 ATM Overview	15
3.2 BISDN Layered Protocol	16
3.3 Details of the AAL Layer	19

Chapter 4 Proposal for an AAL structure	23
4.1 Introduction	23
4.2 Causes of ATM cell losses	25
4.3 Performance of adaptation schemes	28
4.4 SAR_PDU format	35
 Chapter 5 High Priority cell loss compensation scheme	 38
5.1 Introduction	38
5.2 Coding for cell loss recovery	40
5.3 Coding scheme options	42
5.4 Coding structure	43
5.5 Error control options	47
5.6 Encoding/Decoding Fundamentals	48
5.7 Cell loss recovery scheme	52
5.8 Lost CLD cell detection and regeneration	54
5.9 Video cell loss detection and regeneration	57
5.10 Performance Analysis	61
 Chapter 6 Performance evaluation of an ATM multiplexer	 64
6.1 Introduction	64
6.2 Handling different classes of priorities	66
6.3 Queueing model of the multiplexer	70

6.4 Analytical and simulation approaches for evaluating ATM multiplexer performance	72
6.5 Evaluation of the priority scheme	88
Chapter 7 Conclusions	96
Appendix A Source distribution for a single video source	99
Appendix B Generation of geometrically (truncated) distributed random numbers	101
References	103

List of Figures

Figure 1.1	Video distribution system	4
Figure 2.1	Video Codec Functions	11
Figure 2.2	Probability distribution for a single video encoder	12
Figure 2.3	Video-PDU format	14
Figure 3.1	Functional partitioning for Video and ATM	16
Figure 3.2	ATM Adaptation Layer Definition	17
Figure 3.3	CS Sublayer PDU format	20
Figure 3.4	SAR Sublayer PDU format	21
Figure 4.1	Adaptation Layer Structure — CRC per cell	24
Figure 4.2	Adaptation Layer Structure — CRC per Video_PDU	24
Figure 4.3	Adaptation Layer structure	37
Figure 5.1	Encoding Matrix Structure	44
Figure 5.2	Parity cell generation for a column	49
Figure 5.3	Cell loss recovery mechanism a)	53
Figure 5.4	Cell loss recovery mechanism b)	54
Figure 5.5	CLD cell structure	55
Figure 5.6	CLD-RS cell structure	57
Figure 5.7	Cell loss detection	58
Figure 5.8	Parity cell structure	59
Figure 5.9	Regeneration of lost video cell	60
Figure 6.1	Model of the ATM multiplexer	65
Figure 6.2	Partial Buffer Sharing (PBS)	66
Figure 6.3	Push-out scheme	68
Figure 6.4	Replacement strategy for Push-out scheme	69
Figure 6.5	Queueing model of the ATM multiplexer	70
Figure 6.6	The pf of N=10 multiplexed video sources	73

Figure 6.7	HP/LP loss probability vs MUX cycles, N=10, without FEC overhead	75
Figure 6.8	HP/LP loss probability vs MUX cycles, N=10, with FEC overhead	77
Figure 6.9	HP loss probability after decoding, (Option 1) for N=10 . .	78
Figure 6.10	HP/LP loss probability without FEC overhead.	80
Figure 6.11	HP/LP loss probability with FEC overhead.	80
Figure 6.12	HP loss probability after decoding (Option 1)	81
Figure 6.13	HP loss probability after decoding (Option 2)	81
Figure 6.14	HP loss rate improvement factor (Option 1)	82
Figure 6.15	HP loss rate improvement factor (Option 2)	82
Figure 6.16	LP loss rate deterioration factor due to adding FEC overhead	83
Figure 6.17	HP loss rate after FEC decoding, using push-out scheme (Option 1)	89
Figure 6.18	HP loss rate after FEC decoding, using push-out scheme (Option 2)	89
Figure 6.19	HP loss rate improvement using push-out scheme (Option 1)	90
Figure 6.20	HP loss rate improvement using push-out scheme (Option 2)	90
Figure 6.21	HP loss rate without FEC but using push-out scheme . .	92
Figure 6.22	HP loss rate improvement using only push-out scheme .	92

List of Acronyms

AAL	ATM Adaptation Layer
ATM	Asynchronous Transfer Mode
BER	Bit Error Rate
B-ISDN	Broadband Integrated Service Digital Network
CCITT	Comite Consultatif International Telegraphique et Telephonique
CIP	Cell Identification Pattern
CLD	Cell Loss Detection
CPL	Control Plane Layer
CRC	Cyclic Redundancy Check
CS	Convergence Sub-layer
DCT	Discrete Cosine Coding
DPCM	Differential Pulse Code Modulation
FBM	Frame Begin Mark
FEC	Forward Error Correction
FIFO	First In First Out
FN	Frame Number
HEC	Header Error Check field (ATM Cell – CCITT)
HP	High Priority
HPBN	High Priority Bytes Number
LAN	Local Area Network
LG	Line Group
LGN	Line Group Number

LI	Length Indicator
LIFO	Last In First Out
LP	Low Priority
LPBN	Low Priority Bytes Number
MUX	Multiplexer
PBS	Partial Buffer Scheme
PDU	Payload Data Unit
PMD	Physical Medium Dependent
QOS	Quality Of Service
RS	Reed-Solomon
SAR	Segmentation and Reassembly sub-layer
SBC	Sub-Band Coding
SN	Sequence Number
SNP	Sequence Number Protection
SONET	Synchronous Optical NETwork
SPC	Single Parity Check codes
STM	Synchronous Transport Multiplex
UPL	Upper Plane Layer
VBR	Variable Bit Rate
VCI	Virtual Channel Identifier
VQ	Vector Quantization

Chapter 1

Introduction

1.1 Background

The adoption of the asynchronous transfer mode (ATM) as the switching protocol and the introduction of synchronous optical network (SONET) as the transmission standard greatly facilitates the implementation and commercialization of the broadband integrated service digital network (B-ISDN) [1], [2]. Efforts are underway to investigate the feasibility of providing video services over ATM networks with the anticipation that data, voice and eventually video will be integrated into the same transport facility, providing bandwidth efficiency and accommodation of a wide range of services.

The main advantage of ATM is its capability to absorb the variations of different types of traffic, and hence facilitate the integration of sources with different bit rates and statistical characteristics. ATM is a connection-oriented transfer technique, wherein fixed size ATM cells from different sources are asynchronously multiplexed onto a communication channel.

Since video services offer a large market for residential users, but at significantly higher bandwidth requirements than voice and data, success in adopting ATM for B-ISDN depends largely on the feasibility of transporting motion video in an efficient manner [3].

With the standardization of the high-speed transmission protocol SONET, the bandwidth of a B-ISDN is greatly expanded [4]. This gives the possibility of switching and transmitting full-motion video signals. With the introduction of ATM, variable bit rate (VBR) coding and transmission of video signals are made possible. Due to the inhomogeneous nature of video sources and the inherent variations of activity from scene to scene, the information rate after compression tends to be highly variable and non-stable. In the STM-based fixed-rate transmission environment, this inhomogeneity and variation

are partially compensated for by buffering and matching to the characteristics of the constant rate channel. This causes certain delay for video scenes with low motion and causes picture quality degradation for video scenes with active motion. The fixed-rate channel also puts some restriction on the capability of data compression schemes since the compression data rate for the busiest scene has to be taken into account. In VBR coding systems, no rate control buffer is required and a time-transparent or rate-free transmission can be realized, although a network buffer is used to absorb possible simultaneous traffic peaks of multiple users. A statistical gain is obtained by taking advantage of the burstiness of variations of the compressed video source traffic [5]-[7]. The fixed rate coding of fixed-bandwidth system is controlled by the coding rate, while distortion (or quality) varies with the activity of instantaneous scenes. The VBR system is controlled by the distortion (or quality), in which the coding rate varies with the activity of the instantaneous scenes. Therefore, the bits-on-demand property of an ATM network can benefit the VBR coding, for which the data rate may vary drastically when picture quality is kept constant.

From the user's point of view, the advantages of VBR coding due to the introduction of ATM include:

- * format-independent, rate free, and time-transparent transmission;
- * reduction of end-to-end delay;
- * consistent video quality even for active motion area/frames;
- * quality control by the user instead of rate control by the channel.

From the network point of view, the advantages of ATM include:

- * dynamic bandwidth allocation;
- * easy multimedia (data/voice/image/video) integration;
- * network delay/throughput performance improvement.

While VBR coding in the ATM environment provides the network with a number of advantages over the fixed-rate coding in STM circuit-switched networks, it also causes some problems that do not exist in STM networks. Cell delay jitter and cell losses are the two potential major impediments to transporting video on ATM [1], [8]. Cell delay occurs due to the non-uniform arrival of the cells at the receiver, which in turn is the result of random queueing delays in the network. This problem can be overcome by time-stamping the individual cells and providing a sufficiently large buffer at the receiver, thus enabling recovery of a stable clock. However, cell losses have a detrimental effect on the video quality, and can occur due to cells being misdelivered or due to cells being dropped in the event of network congestion. In ATM, retransmission of cells/blocks for real time services, e.g., video, is not possible; therefore error recovery should be either built into the cells themselves or the coding algorithm should have mitigating properties to deal with the lost cells.

Since cell-loss, cell-delay jitter and random bit errors are the potential major problems of transmitting video over ATM networks, a video coding algorithm should be able to deal with these problems. Research in packet video coding suggests that layered coding strategies are most suited for ATM [9]. Details of this are given in the next chapter.

1.2 Thesis Objective

This thesis addresses the issues related to video transmission over an ATM distribution network, for entertainment purposes (Fig 1.1).

The basic system includes several video encoders at the head-end transmitting video signals. These are mapped onto the ATM distribution network consisting of switches and multiplexers. At the tail-end are the decoders and the customer's display device.

Figure 1.1 Video distribution system

The two main objectives involved here are: to simplify the decoder complexity and to accommodate as many video channels into a given output transmission link as possible [10].

The cell loss priority bit in the ATM cell header, recently introduced by CCITT [21], provides two different bearer capabilities in ATM networks. These services differ in the required cell loss rate. The video encoder shown in Fig 1.1 separates compressed visual information and assembles them into two types of cells: high and low priority, according to their importance. The loss of low priority cells does not produce significant degradation in the picture quality while the loss of high priority cells can generate annoying artifacts.

The suggested Quality of Service (QoS) specifies the loss rate for both levels of traffic [10]. For high priority the loss rate should be $<10^{-12}$ while the corresponding figure for low priority is 10^{-6} . These will be the performance targets throughout the thesis. Transfer delay is not a stringent condition though a worst case delay of 2.5 ms has been assumed.

A possible way to provide the ATM network with the ability to manage the two bearer capabilities is to operate the access to the network facilities according to a priority scheme. More specifically, the buffers in the nodes can be used to give priority to cells belonging to the most demanding bearer capability. Preliminary performance evaluation of this approach in case of bursty and VBR traffics [44], [45] have shown its effectiveness in improving the network efficiency.

Previous studies [31], [32] have illustrated the fact that employing a cell loss compensation technique (using FEC) considerably overcomes the problem of cell losses due to buffer overflow.

We therefore evaluate the performance of a statistical ATM multiplexer with the following objectives:

- a) To propose an AAL (ATM Adaptation Layer) protocol incorporating error detection for all types of errors and forward error correction (FEC) to protect against cell losses due to congestion.
- b) To propose a suitable multiplexing and priority handling scheme.
- c) To evaluate the merits of the above proposals compared particularly with a scheme not using FEC and/or a priority handling scheme.

1.3 Thesis Organization

Chapter 2 presents a brief description of several video coding schemes, with emphasis on the one used in this thesis. Chapter 3 describes the ATM layered protocols and the corresponding functions and formats.

Because of the susceptibility of low bit rate video codecs to transmission errors, many of them incorporate FEC schemes, which can cope with random or short burst errors. Chapter 4 discusses the problem of random bit errors affecting video transmission and suggests ways of tackling them.

An ATM network transports video data in cells of many bytes; CCITT specifies 53, of which 8 are taken up by overhead, leaving 45 bytes for video data. In this case, network overload causes data to be lost in multiples of 360 bits. Cell loss is therefore a major consideration when operating high compression video codecs over an ATM network.

The most straightforward approach, at least in concept, is to apply error correction to the end-to-end link. It has been shown by Kitami and Tokizawa [32] that employing a cell loss compensation scheme greatly enhances the transmission quality. Chapter 5 therefore looks into applying FEC to tackle consecutive High-Priority (HP) cell losses due to buffer overflow, when the network becomes congested. The technique involves detecting cell losses at the decoder and inserting dummy cells prior to any reconstruction of lost HP cells. The whole process involves delay at the decoder end. This solution is therefore applicable where high quality is required but delay is not a important criterion; video distribution services and broadcast television being examples.

From a practical point of view, the number of video channels that can be provided to a customer by multiplexing onto a common channel, and their respective buffer requirements is of considerable importance. Chapter 6 deals with the performance evaluation of an ATM multiplexer using both analytical and simulation approaches and illustrates the effect on performance in employing FEC (to tackle consecutive HP cell losses) and/or the priority handling scheme.

Finally, Chapter 7 highlights the major results obtained in this thesis.

Chapter 2

Video Source Coding Schemes

2.1 Overview

The concept of classifying video information into different importance classes with different priorities is an evolution of the technique used extensively in packet voice communication. The objective is to restrict cell dropping to those cells assigned with lower priority and guarantee timely and error-free transmission of cells assigned with higher priority, thereby minimizing the degradation of picture quality when the network becomes congested. This selective discarding idea is implemented by so-called layered source coding. In layered video transmission, video quality degrades gradually with the cell loss rate when only the cells in the least important layer are discarded.

2.2 Layered Video Coding Schemes

The first step in layered video source coding is to classify picture information in terms of its importance, and then assign the most important class the highest priority. After the separation, information is layered into a pyramid form and compressed by various bit rate coding schemes. At the receiver, these layered signals are decoded and combined to reconstruct the original signal, subject to certain distortion.

There are many coding schemes that can be used for video transmission in the ATM environment. These can be broadly classified [11] as:

1) Transform Coding: Natural image/video scenes contain a lot of redundancy in the spatial/temporal domain. The purpose of transform coding is to map highly correlated image samples to another domain in which the image coefficients are more statistically independent [12].

A video signal can be considered as a sequence of images, or time-varying imagery. Therefore, any image-compression method can be applied to digital video by applying it to individual video frames. There also exists considerable redundancy in the temporal domain (frame-to-frame), and interframe coding often makes substantial reductions in the data rate possible. Examples of interframe coding schemes include conditional replenishment and motion-compensated coding. Conditional replenishment is an interframe predictive coding scheme based on transmitting and coding differences between the present frame signal and the previous signal. When there is no motion or slow motion involved in the scene, this scheme is efficient. If the object in the picture moves to some extent between the frames, the coding efficiency will be greatly improved if this motion vector can be estimated in some way. This is the principle of motion compensation coding. Based on motion compensation for interframe displacement estimation and transform coding schemes for intraframe coding, a great number of transform-based coding approaches exist for VBR video transmission in the ATM environment [6], [7] and [13].

A video signal can be considered as a sequence of images, or time-varying imagery. Therefore, any image-compression method can be applied to digital video by applying it to individual video frames.

2) Combined-Transform Coding: In transform coding, an image is first divided into many sub-blocks (usually 8×8 or 16×16 pixels). These blocks are then transformed, quantized, coded and transmitted independently. The correlation within each block is substantially reduced or eliminated. The correlation between different blocks, however, remains high. The problems associated with this are two-fold. On one hand, any remaining high correlation affects the compression performance. On the other hand, it creates some artifacts across the block boundary, especially for low rate coding, known as "blocking effect", which is due to the coarse quantization of transformed coefficients. Considerable research in this area has been conducted in the last decade. A simple method for reducing the number of artifacts is by low-pass filtering [14], which tends to

smooth out the block boundary at the expense of a possible loss of some high-frequency information.

3) Predictive Coding: In predictive coding [15], the value of a pixel is predicted based on the value of a neighboring group of pixels related by an autoregressive (AR) model. The AR model is defined by a set of coefficients and a residual image. Data rate reduction is realized by coding the set of coefficients and residual image. The main disadvantage of this type of scheme is its error propagation nature and low compression ratio compared with transform-based coding schemes.

4) Sub-band Coding: The basic idea of sub-band coding (SBC) is to divide the frequency band of the signal into a number of sub-bands by a bank of bandpass filters [16]. Each sub-band is then translated to baseband by down-sampling and encoded separately. Motion vectors can also be estimated on individual bands. For reconstruction, the sub-band signals are decoded and up-sampled back to the original frequency band by interpolation. The signals are then summed to produce a close replica of the original signal.

Sub-band coding can be combined with other coding techniques to further improve the coding efficiency. SBC/DPCM, SBC/DCT, and SBC/VQ schemes have been developed [17], [18]. The VBR sub-band coding scheme has also been proposed and has been shown to be an appropriate scheme for video transmission over packet-switched network due to its robustness and compression performance [19].

5) Vector Quantization: Vector quantization (VQ) operates on a group of pixels instead of a single pixel. There are three main reasons why vector quantization is useful: 1) VQ is capable of removing the interpixel correlation, 2) VQ can be used to remove the interpixel statistical dependence (i.e., VQ can also pick up the redundancy between uncorrelativeness and independence, which is only equivalent for Gaussian sources), and 3) VQ picks up a certain degree of geometric redundancy. The idea of VQ was proposed several years ago but only recently has it become a feasible scheme for image coding after

a computationally efficient code-book search algorithm known as LBG was developed [20]. Many improvements on the LBG algorithm have since been made and although VQ is still not feasible for real-time VBR video transmission, due to the huge efforts in establishing and searching its codebook, the improvement in VQ algorithm and advances in VLSI technology will make it an ideal candidate for future real-time data compression applications.

2.3 VBR source coding scheme used in the thesis

Of the above mentioned compression schemes, the SBC/DPCM coding scheme is used in this thesis. The encoder and decoder functions, used in the problem, are shown in Fig 2.1, [10].

CODEC FUNCTIONS

Figure 2.1 Video Codec Functions

Video coding is done on a frame by frame basis, with each frame conforming to the NTSC standard of 525 lines per picture frame and a frequency of 30 frames per second.

The unit of video information after bandwidth compression coding is called a Linegroup (LG), and is essentially the compressed information of 3 lines of a picture frame. Because the picture frames are repeated every 1/30 sec, a LG would be delivered out of a video encoder every 190.5 μ s. In accordance with ATM terminology, a single LG will henceforth be called a Video_PDU (Protocol Data Unit). The important distinction to be made here is that the length of the Video_PDU will change but not its arrival interval.

We model the Video_PDU length by a truncated geometric distribution with the minimum, maximum and average length being 1, 16 and 5 cells respectively [10]. This translates to a peak and average bit rate of 35.61 and 11.128 Mbits/s respectively for each video encoder. Fig 2.2 shows the probability distribution function for a single video encoder in terms of the number of cells in a Video_PDU. The distribution was arrived at as outlined in Appendix A.

Although these specific maximum, minimum and average values are based on a particular implementation [10], it seems reasonable to suppose that similar models might approximate the characteristics of other video coders of this type.

Figure 2.2 Probability distribution for a single video encoder

Conforming to the current CCITT recommendations regarding priorities [21], two levels of priority are considered and are henceforth termed High-Priority (HP) and Low-Priority (LP). As mentioned earlier, the suggested QOS [10] requires that the HP loss rate be less than 10^{-12} and the corresponding figure for LP is 10^{-6} .

Since each Video_PDU will contain both HP and LP video information, there is a need to indicate to the decoder, for error recovery purposes, the amount of HP and LP information in a Video_PDU. This is done by including a field VDBN (Valid Data Bytes Number) which specifies the number of HP and LP bytes in a single Video_PDU.

A typical Video_PDU would contain

High Priority bits (HP) **1795 bits = 225 octets**

Low Priority bits (LP) **585 bits = 75 octets**

Total **300 octets**

There is a typical ratio of 3:1 between the HP and LP bytes of a Video_PDU. This is based on empirical information provided [10]. In the maximum case when there are 16 cells in a Video_PDU, 12 of these will be HP and the remaining 4 will be LP. The 12 HP cells will each contain 45 bytes of information in the payload. Therefore a maximum of 540 bytes of HP information and 180 bytes of LP information has to be specified by the HPBN and LPBN field respectively. Hence, the HPBN field is taken to be 10 bits and the LPBN field as 8 bits. Fig 2.3 shows the typical Video_PDU format.

Legend

FBM ***Frame Begin Mark***

FN ***Frame Number***

LGN ***Line Group Number***

HPBN ***High Priority Bytes Number***

LPBN ***Low Priority Bytes Number***

VDBN ***Valid Data Bytes Number = HPBN + LPBN***

R ***Reserved***

M ***Synchronization Field***

Figure 2.3 Video-PDU format

Description:

FBM Occurs as the first octet in a frame, and consists of an 8 bit code.

FN Provides a frame identity which repeats itself every 256 frames thus providing further integrity checking.

LGN Occurs in the first octet of a linegroup.

VDBN The number of octets in the linegroup. Consists of HPBN and LPBN. It also provides an indication as to when the next linegroup begins.

M 8 bits are assumed for synchronization purposes.

Chapter 3

ATM based B-ISDN

3.1 ATM Overview

As noted in Chapter 1, an extremely wide variety of services is expected to be provided over BISDN networks. Many of these services are yet to be defined, and more importantly, the needed mix of services will probably vary from one customer to the next as well as over time. For these reasons it is critical that the multiplexing and switching technique used to manage the bandwidth of BISDN facilities be extremely flexible. Traditional time division techniques which divide bandwidth into a number of fixed capacity channels simply do not have this needed flexibility. Instead, BISDN is being based on a new ATM multiplexing structure [22].

With ATM, the transmission-bit stream is divided into a number of fixed-length *cells*. Each cell is made up of a *header* field containing network control and an *information* field containing user data. Unlike synchronous time division techniques which identify calls by the position of a time slot within a frame (i.e. position multiplexing), ATM techniques associate calls with cells by a label in the header (i.e. label multiplexing). Connections are established by setting up translation tables at switching and multiplexing points which map an incoming label into an outgoing link and label. The connections so established are called *virtual circuits* since no bandwidth is consumed unless traffic is actually being sent. A given source could generate cells at a regular rate or at a statistically varying rate. Thus, not only can ATM provide continuous bit rate (CBR) services, it can also exploit the burstiness of many types of information streams such as variable rate coded video signals, as in our case.

3.2 BISDN Layered Protocol

An ATM based BISDN layered protocol for video transmission is illustrated in Fig 3.1.

Figure 3.1 Functional partitioning for Video and ATM

In current CCITT recommendations [23], four classes of sources are defined in terms of their bit rates, connection mode, and timing requirements (Fig 3.2). VBR coded video sources fall into Class B (variable bit rate, connection-oriented, and preserved timing between source and destination). The corresponding functions and formats for the different layers are described briefly [10].

1. Physical Medium Dependent (PMD) Layer

The PMD is responsible for the proper bit transmission and performs functions which are necessary to insert/extract the cell flow into/out of a transmission frame.

Figure 3.2 ATM Adaptation Layer Definition

2. ATM Layer

The ATM layer is the user independent network layer and provides integrated cell multiplexing and switching services, irrespective of the incoming source formats.

The ATM cell is 53 octets in length, consisting of a 5 octet header field (in the above figure, the header field divisions are shown in bits) and a 48 octet information

field. The above cell format will be used at the User-Network Interface (UNI) and the Network-Node Interface (NNI).

The ATM cell structure at the UNI is shown in Fig 3.2. The control information in the header field includes :

Generic Flow Control (GFC) Provides contention resolution and simple flow control.

Virtual Channel Identifier (VCI) Used to establish connections through the use of translation tables, that map an incoming VCI to an outgoing link.

Virtual Path Identifier (VPI) Used similar to the VCI to establish a virtual path connection for one or more VCI. It allows simplified network routing functionality and management.

Payload Type (PT) Differentiates between types of cells that traverse the same virtual circuit.

Cell Loss Priority (CLP) Used to indicate priority of cells, with lower priority cells having a higher probability of being discarded depending upon the network conditions.

Header Error Control (HEC) Provides a CRC calculation over the header field for error detection and correction.

For NNI, the header does not contain a GFC field and those bits are used for the VPI field.

3. Adaptation Layer (AAL)

The ATM adaptation layer (AAL) performs cell assembly/disassembly and individual protocol depending on each service, such as lost cell handling and forward error correction. Its function is to carry service-specific attributes not carried in or preserved by the ATM layer, and to compensate for impairments introduced at the ATM or PMD layers.

The AAL itself has two sub-layers, both of which will be discussed in detail in the next section.

4. User/Control Plane

In the user plane layer (UPL), video signal formatting, VBR encoding/decoding, high-level signal processing, and end-to-end error recovery are performed. The control plane layer (CPL) is responsible for end-to-end signalling and routing.

An ATM cell is formed at the adaptation layer and queued and multiplexed to form a SONET frame in the physical layer.

3.3 Details of the AAL Layer

As mentioned before the AAL layer consists of two sub-layers. These are:

a) Convergence (CS) Sub-layer

On transmission, the Convergence Sublayer accepts variable length User Protocol Data Units (USER_PDUs) from the user layer. The CS prepends a header to the USER_PDU and then appends 0-3 pad octets to the USER_PDU to build it out to an integral multiple of 32 bits. Next, it appends a trailer to the concatenated header, USER_PDU, and the PAD structure. This collection, the header, USER_PDU, PAD, and the trailer is referred to as a CS_PDU. After appending, the trailer the CS passes the CS_PDU to the SAR Sublayer for segmentation and further transmission.

On reception, the Convergence Sublayer validates the collection of SAR_PDU payloads received from the SAR Sublayer by using the information contained in the CS_PDU header and trailer. It removes the PAD octets, if any, and presents the validated CS_PDU payload to the user layer.

Figure 3.3 CS Sublayer PDU format

Description

CS_PDU Type This field identifies the convergence service to be provided.

BE (Begin end) Tag (header) This field allows the association of BOM (Begin of message) SAR_PDU with the EOM (End of message) for the same CS_PDU, and is used at the receiver to detect the loss of ATM cells over several CS_PDUs. For a given CS_PDU the sender inserts the same value in this field, and in the similarly named field in the CS_PDU trailer. The sender simply increments the value placed in the BE tag fields for each successive CS_PDU sent over a given SN value. BE tag values are cycled modulo 256.

BA (Buffer Requirement) Size Used to encode the buffer requirements required to receive the CS_PDU.

USER_PDU The variable length USER_PDU field contains user information.

LI (Length Indicator) Used to encode the length of the CS_PDU Payload. Also used to detect the loss or the gain of cells through misinsertion.

BE Tag (trailer) For a given CS_PDU the sender shall insert the same value in this field as was inserted in the BE tag (header) of the CS_PDU.

RSVD This field is reserved for future use (possibly for conveying protocol control information).

b) Segmentation and Reassembly (SAR) Sub-layer

On transmission, the SAR sublayer accepts a variable length CS_PDU from the Convergence Sublayer and maps the CS_PDU into a sequence of N SAR_PDU's, by placing at most 45 octets of the CS_PDU into a SAR_PDU payload, along with additional control information, described below, that are used to verify the integrity of the SAR_PDU payload on reception and to control the reassembly process.

On reception, the SAR_PDU is validated, and the user data in the SAR_PDU is passed to the Convergence Sublayer.

SAR Sublayer PDU formats

As can be seen from Fig 3.2 there are different possible structures for the SAR_PDU, depending on the AAL Type used. In general however, the SAR_PDU_Header and SAR_PDU_Trailer fields encapsulate the SAR_PDU_Payload, which contains a portion of the CS_PDU. Details of the SAR_PDU format used in this thesis will be discussed in the next chapter.

Figure 3.4 SAR Sublayer PDU format

Description

SN (Sequence Number) Used to detect lost or misinserted cells.

SNP (Sequence Number Protection) Is a (8,4) code on the SN field providing error detection and correction options.

LI (Length Indicator) This field is the length indicator for partially filled cells.

CRC (Cyclic Redundancy Check) Is the error check sequence which detects up to two correlated bit errors.

Chapter 4

Proposal for an AAL structure

4.1 Introduction

A critical function of the adaptation layer (AAL) is to prevent corruption of information carried in ATM cells. This chapter outlines a suitable Adaptation Layer (AAL) structure for the given problem by analyzing the performance of two possible AAL structures from the perspective of error control for the payloads.

We present two possible strategies for the error control of a single Video_PDU payload.

1. Construct an error check sequence for the whole Video_PDU, and then segment it into cell payloads — "CRC per Video_PDU " scheme.

2. Divide the Video-PDU into cell payloads and append a CRC check sequence to each of these payloads. In this case, there is an error check sequence for each cell — "CRC per cell" scheme.

There are three possible types of information corruption in ATM networks that are based on fiber optic transmission systems: 1) Random errors 2) Cell loss due to congestion and 3) Burst errors.

In the following sections we shall perform a quantitative evaluation of the above mentioned two error control strategies for the three types of errors. The CS_PDU and SAR_PDU structure for both strategies are shown in Fig 4.1 and Fig 4.2 respectively. The ensuing performance analysis of the two strategies will be based on these formats.

CS_PDU

SAR_PDU

Figure 4.1 Adaptation Layer Structure — CRC per cell

CS_PDU

SAR_PDU

Figure 4.2 Adaptation Layer Structure — CRC per Video_PDU

4.2 Causes of ATM cell losses

There are three possible types of cell losses in ATM networks [24], assuming fiber optic transmission systems. The causes of these types of errors are explained in the following.

1. Random errors

Random errors occur due to Gaussian noise and other interference in the communication channel. If an uncorrectable error occurs in the address field of an ATM cell, the cell will not be delivered to the right destination. For real time services such as video this cell is considered to be lost at the receiver end and discarded. The random error rate is calculated as follows.

For each ATM cell the header size is 5 octets and the information size 48 octets. Therefore, assuming a BER due to random errors to be $p=10^{-12}$

$$\begin{aligned} P\{\text{Cell loss due to random errors in the header}\} &= 1 - P\{\text{no bits in error}\} \\ &= 1 - (1 - p)^{40} \\ &\simeq 40p = 4.0 \times 10^{-11} \end{aligned}$$

Hence, we can conclude that cell loss due to random errors is quite low and can be taken care of by the (40,32) single error correcting HEC (Header Error Check) that is included in the cell header.

2. Cell loss due to congestion

An ATM network takes advantage of statistical multiplexing but also runs the risk of simultaneous traffic peaks from multiple sources. Although a buffer can be used to absorb the instantaneous traffic peak to some extent, there is still a possibility of buffer overflow in case of congestion. Furthermore, the buffer size has to be traded off with the transmission delay. In case of buffer overflow, the network congestion control protocol

will drop cells. Details of dealing with cell losses due to buffer overflow are described in the next chapter.

3. Burst errors

In long haul fiber transmission systems, there will be burst errors caused predominantly by protection switching [25]. A failed repeater would cause transmission of messages to be switched onto a protection line and during the switching and reframing process the circuit would be essentially open, resulting in a high error rate for a small duration of time. According to [25], the average number of protection switches per day on a 150 Mbps line is 0.24. The duration of each protection switch is about 30 ms. During a protection switch, all the errored cells are lost. Further, when several identical sources are multiplexed, and the switch samples the sources one after the other in a sequential manner, then the cells can be assumed to be lost independently. We can write the probability of cell loss on account of protection switches as follows

$$\frac{\text{No. of protection switches per day} \times \text{No. of cells in 30ms}}{\text{No. of cells per day}}$$

No: of protection switches per day = 0.24

No: of cells in 30ms on a 150 Mbps line = 1.0613×10^{10}

No: of cells in a day on a 150 Mbps line = 2.16×10^{11}

Therefore,

$$P\{\text{Cell hit by burst error}\} = t = 8.34 \times 10^{-8} \quad \dots 4.1)$$

We next determine the probability of cell misdelivery and the probability of loss due to the network detecting an error in the header and discarding the cell.

$P\{\text{Cell misdelivery}\} =$

$P\{\text{Errors in header}\} \times P\{\text{Errors undetected}\} +$

$P\{\text{Errors in header}\} \times P\{\text{Errors detected}\} \times P\{\text{Errors incorrectly corrected}\}$

The individual probabilities are [24]

$$P\{\text{Errors in header}\} = 8.34 \times 10^{-8}$$

$$P\{\text{Errors undetected}\} = 2^{-r} \quad [24].$$

where n = Total no. of bits in the header including r CRC bits.

$$P\{\text{Errors incorrectly corrected}\} = (n+1) \times 2^{-r}$$

Therefore

$$P\{\text{Cell misdelivery}\} = 3.25 \times 10^{-10} \quad \dots 4.2)$$

Similarly,

$$P\{\text{Network discards cells}\} =$$

$$P\{\text{Errors in header}\} \times P\{\text{Errors undetected}\} \times P\{\text{Unassigned VCI}\} +$$

$$P\{\text{Errors in header}\} \times P\{\text{Errors detected}\} \times P\{\text{Errors uncorrected}\}$$

It has been assumed that half the VCI's are unassigned. Therefore,

$$P\{\text{Unassigned VCI}\} = 0.5$$

$$P\{\text{Errors uncorrected}\} = 1 - \{(n+1) \times 2^{-r}\} \quad [24].$$

Hence,

$$P\{\text{Network discards cells}\} = 8.307 \times 10^{-8} \quad \dots 4.3)$$

The probability of cell loss is equal to sum of the probability of cell misdelivery and the probability of error detection and discard by the network. Comparing eqns. 4.2 and 4.3 it can be seen that the probability of cell misdelivery is almost 2-3 orders of magnitude less than the latter. Hence, for the remainder of the chapter losses due to cell misdelivery shall be neglected and cell losses are assumed to be accounted for mainly by protection switches.

4.3 Performance of adaptation schemes

In the rest of the chapter, the above mentioned adaptation schemes are examined from the perspective of error control for the payload [24], [26].

A. RANDOM ERRORS

The measure of comparison is the probability that an errored Video_PDU is accepted as valid by the adaptation layer. The analysis is done first for the CRC per cell scheme.

a) CRC per cell

Suppose that the Video_PDU has on an average N cells. It will then be in error even if one of the cells is in error. The Video_PDU will be accepted as valid if the errors in all the errored cells are not detected. The probability of such a thing happening is

$P\{i \text{ cells in error, no errors detected}\}$

The second possibility is when at least one of the cells from 1 to $N-1$ is lost and the N th cell is corrupted in such a way that a partial Video_PDU appears to be complete. For this to happen the length field of the N th cell must be in error and at the same time match the length of the collected cells, that is:

$P\{i \text{ cells lost and } N\text{th cell is errored so as to make a partial Video_PDU appear complete}\}$

The third possibility is when the last cell is discovered to be in error and dropped. If a cell in the next Video_PDU is in error in such a way that the whole bunch of collected cells appear to form a legitimate Video_PDU, then a faulty Video_PDU is accepted as valid. This can happen if the cell provides the correct length and sequence number. This probability is given as.

$P\{N\text{th cell lost and cell from a subsequent Video_PDU is errored such that the faulty Video_PDU appears correct}\}$

Combining the above three possibilities, we have

$P\{ \text{Video_PDU accepted as valid, though in error} \} =$

$$\sum_{i=1}^N P\{i \text{ cells in error, no errors detected}\} +$$

$$\sum_{i=1}^{N-1} P\{i \text{ cells lost and Nth cell is errored so as to}$$

$$\text{make a partial Video_PDU appear complete}\} +$$

$P\{\text{Nth cell lost and cell from a subsequent Video_PDU}$
is errored such that a faulty Video_PDU appears correct}

Assuming the following,

Cell payload CRC is capable of detecting 1, 2 and 3 bit error patterns

End-to-end bit error rate (BER) = $p = 10^{-8}$

No: of bits in cell payload = $n = 45 \times 8 = 360$

Average no: of cells in a Video_PDU = $N = 5$

A cell will be in error if there are more than 3 errors in the cell. Therefore,

$P\{ \text{Undetected error in cell } k \} = x, \quad \text{for any } k$

$$x = 1 - \sum_{i=0}^3 {}^nC_i p^i (1-p)^{n-i}$$

$$= 3.6 \times 10^{-6}$$

If we choose a CRC of r bits for the cell payloads then

$P\{ \text{Unable to detect cell } k \text{ in error/cell } k \text{ is in error} \} = y = 2^{-r}$

For $r=10$ bits, $y = 0.976 \times 10^{-3}$

Therefore,

$$\begin{aligned}
& \sum_{i=1}^N P\{i \text{ cells in error, no errors detected}\} \\
&= \sum_{i=1}^N {}^N C_i x^i (1-x)^{N-i} \times y^i \\
&= [xy + (1-x)]^N - 1 \\
&= 1.12 \times 10^{-15} \quad \dots 4.4)
\end{aligned}$$

We next determine the cell loss due to error check. Now a cell could be lost due to an error check in the header or an error check in the payload. Since we are considering the error detection and correction mode for the header field and only error detection mode for the cell payloads:

$$\begin{aligned}
& P\{\text{Cell loss due to error check in the header}\} \\
& u = [1 - (1-p)^{40} - 40p(1-p)^{39}] (1 - 2^{-8}) \\
& \text{Thus, } u = 1.55 \times 10^{-13} \quad \dots 4.5)
\end{aligned}$$

$$\begin{aligned}
& P\{\text{Cell loss due to error check on the payload}\} \\
& w = [1 - (1-p)^{384}] (1 - 2^{-r}) \\
& \text{Thus, } w = 3.83 \times 10^{-6} \quad \dots 4.6)
\end{aligned}$$

Therefore,

$$P\{\text{Cell loss due to error check}\} = u + w = 3.83 \times 10^{-6}$$

Using the above derived values,

$$\begin{aligned}
 & \sum_{i=1}^{N-1} P\{\text{i cell lost and Nth cell errored so as to make a partial} \\
 & \quad \text{frame appear complete}\} \\
 &= \sum_{i=1}^{N-1} {}^{N-1}C_i (u+w)^i [1 - (u+w)]^{N-1-i} a x \\
 & \quad \quad \quad = 6.81 \times 10^{-15} \quad \dots 4.7)
 \end{aligned}$$

where $a = P\{\text{Length field \& SN/SNP is errored so that partial Video_PDU appears complete}\} = 2^{-14}$

The third term,

$P\{\text{Nth cell lost and cell from subsequent Video_PDU is errored such that the faulty Video_PDU appears correct}\}$

$$\begin{aligned}
 &= (u+w) ax \\
 &= 0.08 \times 10^{-15} \quad \dots 4.8)
 \end{aligned}$$

Combining Eqns. 4.4, 4.7 and 4.8

$$P\{\text{Video_PDU accepted as valid, though in error}\} = 8.01 \times 10^{-15} \quad \dots 4.9)$$

b) CRC per Video_PDU

Here again the Video_PDU contains N cells. There is, however one CRC sequence (of 4 octets) per Video_PDU. Once again there are three components:

The first component is when there are greater than 3 errors in the frame which are not detected by the CRC.

$$P\{ \text{Video_PDU in error} \} = q$$

$$= 1 - \sum_{i=0}^3 N_n C_i p^i (1-p)^{N_n-i}$$

$$= 1.61 \times 10^{-10}$$

Now,

$$P\{ \text{Video_PDU errors undetected/Video_PDU in error} \} = 2^{-r}$$

where, r is the size of the CRC field for the payload.

Thus the second component is given by:

$$P\{ i \text{ cells amongst the first } N-1 \text{ cells are lost, Video_PDU CRC is unable to detect} \}$$

$$= {}^{N-1}C_i u^i (1-u)^{N-1-i} \cdot 2^{-r}$$

The third component is the probability that the N th cell is lost and a cell of the next Video_PDU is errored in such a way that the collected cells appear to form a complete Video_PDU. For this to happen the cell from the next Video_PDU must be errored to provide the correct sequence number and also pass the CRC check. As shown in the CRC per cell scheme, the probability of this happening is considerably smaller and hence can be neglected.

Combining the above components

$$P\{ \text{Video_PDU accepted as valid, though in error} \}$$

$$= q \cdot 2^{-r} + \sum_{i=1}^{N-1} {}^{N-1}C_i u^i (1-u)^{N-1-i} \times 2^{-r}$$

$$= 2^{-r} \left[q + \left\{ 1 - (1-u)^{N-1} \right\} \right]$$

$$= 1.57 \times 10^{-12} \quad \dots 4.10)$$

Comparing the values for CRC per cell and CRC per Video_PDU (eqns. 4.9 and 4.10), it can be seen that the former is almost 3 orders of magnitude less.

B. CONGESTION

The effect of cell losses due to congestion on the two adaptation layer schemes is considered next. Once again the Video_PDU consists of N cells. The basis of comparison between the two schemes is:

P{ Video_PDU accepted as valid, Video_PDU is hit by cell losses due to congestion }

a) CRC per cell

The first probability stems from the fact that any number of the first N-1 cells could be lost due to congestion. The Nth cell's Length Indicator (LI) field will identify such losses. However, the Nth cell could be in error (with probability x) in such a way that the length matches the partial frame (with probability = a) and the cell CRC is unable to detect those errors (prob. = 2^{-r}). Hence the total probability is the product of these individual probabilities.

The second probability arises when the Nth cell is lost and a cell from the subsequent Video_PDU completes the errored Video_PDU. For this to occur, the cell in the subsequent Video_PDU must have the same LI, SN & SNP field and at the same time pass the CRC check. This probability is much less than the first one. Hence,

P{ Video_PDU accepted as valid, Video_PDU is hit by cell losses due to congestion }

$$\begin{aligned}
 &= \sum_{i=1}^{N-1} {}^{N-1}C_i s^i (1-s)^{N-1-i} 2^{-r} x a \\
 &= x a 2^{-r} \left[1 - (1-s)^{N-1} \right] \\
 &= 1.388 \times 10^{-11} \quad \dots 4.11)
 \end{aligned}$$

where s = cell loss due to congestion.

A figure of $s=10^{-6}$ has been assumed in the above calculations because that is the specified LP loss rate and our system must, at the least, deliver that loss rate.

b). CRC per Video_PDU

Here again there are two possibilities which would cause a Video_PDU affected by cell losses to be accepted as valid.

The first possibility is when the Video_PDU CRC fails to detect cell losses amongst the first 1 through N-1 cells. The second possibility is when the Nth cell gets lost and a cell from a succeeding Video_PDU completes the errored Video_PDU. This probability as mentioned before is much lower than the first. Therefore,

P{ Video_PDU accepted as valid, Video_PDU is hit by cell losses due to congestion }

$$\begin{aligned} &= \sum_{i=1}^{N-1} {}^{N-1}C_i s^i (1-s)^{N-1-i} 2^{-r} \\ &= 2^{-r} [1 - (1-s)^{N-1}] \end{aligned}$$

$$= 3.906 \times 10^{-7} \quad \dots 4.12)$$

Comparing the two schemes (eqns. 4.11 and 4.12), in terms of cell losses due to congestion, we once again find that the CRC per cell schemes outperforms the CRC per Video_PDU scheme.

C. BURST ERRORS

As mentioned earlier, in a fiber optic physical medium burst errors are caused by protection switches. Long haul fiber transmission systems are more prone to burst errors as compared to short haul where burst errors are almost negligible. Short haul fiber systems will be in effect in our case where video on demand services are being considered. However, we do the analysis considering burst errors given that at a later date these services may be extended over long distances.

a). CRC per cell

When a cell is hit by burst errors, there is a large number of errors in the header and information field. If the ATM header has a (40,32) single error correcting HEC then the probability of errors in the header remaining undetected is 2^{-8} . If an r -bit CRC field is used for the information field then the probability that the information field remains undetected is 2^{-r} . Therefore,

P{ Video_PDU accepted as valid, Video_PDU is hit by burst errors }

$$\begin{aligned}
 &= {}^N C_1 t 2^{-8} 2^{-r} \\
 &= 1.59 \times 10^{-11} \quad \dots 4.13)
 \end{aligned}$$

Here it is assumed that only 1 cell is traversing the burst error. In a normal situation during a burst error there would be more cells getting lost and this would mean more CRC checks would have to fail. Hence the probability is much less than the one derived above.

b). CRC per Video_PDU

A cell hit by a burst error is detected by the HEC with a probability of $(1 - 2^{-8})$. Now, the probability that the Video_PDU CRC cannot detect a burst error is 2^{-r} . Therefore,

P{ Video_PDU accepted as valid, Video_PDU is hit by burst errors }

$$\begin{aligned}
 &= t \sum_{i=1}^{N-1} {}^{N-1} C_i (1 - 2^{-8})^i (2^{-8})^{N-1-i} 2^{-r} \\
 &= t 2^{-r} \left[1 - (2^{-8})^{N-1} \right] \\
 &= 3.25 \times 10^{-8} \quad \dots 4.14)
 \end{aligned}$$

Comparing eqns. 4.13 and 4.14, we find that yet again the CRC per cell outperforms the CRC per Video_PDU scheme.

4.4 SAR_PDU format

Having compared the two adaptation schemes, it is concluded that the CRC per cell scheme is superior to the CRC per frame scheme. From these results, a Length Indicator (LI) field of 6 bits and a CRC error check field of 10 bits are proposed as part of each SAR_PDU.

In the CRC per cell case, cell losses are detected by the packet length indicator. The performance of this scheme is robust since such losses will go undetected only when the length indicator (LI), if errored, passes the CRC check and is equal to the collected packet. However, in the case with a CRC per Video_PDU, any cell loss translates into a burst error and the burden of this detection lies on the Video_PDU CRC. Therefore, the performance of the CRC per Video_PDU is almost 3–4 orders of magnitude poorer than the CRC per cell.

The calculations in the above section have been done assuming $N = 5$ (the average number of cells in a Video_PDU). As we increase the value of N , up to a maximum of 16, the performance of the CRC per cell scheme improves further.

By including the LI and CRC field as part of the error checking, the additional bandwidth consumed is about 17.73 Kbps, which is a negligible fraction (about 0.1%) of the average transmission rate of a source – 11.128 Mbps per source.

Having established a structure for the SAR_PDU, the adaptation layer structure is now specified as shown in Fig 4.3.

Figure 4.3 Adaptation Layer structure

It should be noted here that the SN field starts fresh for each Video_PDU. After the SN and SNP field are tagged onto the SAR_PDUs they must be passed to the ATM layer for multiplexing and further transmission. Keeping in mind that error control is a function of the Adaptation Layer (AAL), the SAR_PDUs containing HP information (i.e. HP cells) are passed onto an encoder matrix for error protection against consecutive cell discards before being passed onto the ATM layer, while the LP SAR_PDUs are passed onto the ATM layer directly.

The LP cells are stored in a pre-buffer at the ATM layer until the HP cells for that particular Video_PDU are encoded against consecutive cell loss and passed onto the ATM layer. Following this, all the cells belonging to the Video_PDU (HP followed by LP cells) are multiplexed along with cells from other sources onto a common channel.

Chapter 5

High Priority cell loss compensation scheme

One of the major objectives of this thesis is to determine whether the use of a FEC scheme would be beneficial, given the ATM statistical multiplexing approach and a channel of fixed capacity.

In this chapter, we shall propose first a scheme using FEC to compensate for consecutive cell losses when buffer overflow occurs due to congestion in cell multiplexers. Though the ATM cells, involved this research, are of two different priority levels (low priority and high priority) only high priority cells are encoded to protect against buffer overflows.

5.1 Introduction

One of the major problems inherent in ATM-based networks for video services is the cell discard problem which may seriously degrade picture quality especially when redundancy has already been reduced by VBR coding.

In packet networks such as LANs, packet retransmission is employed to overcome the problem of cell losses. However, this is effective only in case of non-time constrained traffic such as data services and is not practical for ATM networks which would be handling, in addition, services such as voice and video.

There are three possible alternatives for reducing the quality degradation due to cell discard.

1. Proper capacity allocation at call set-up to reduce cell congestion [27], [28].
2. Forward error correction (FEC) to recover discarded cells at terminals in which a recipient can recover lost or erroneously received data based on error control bits

(which are added at the source end to the information sequences). Recovering lost cells reduces the need for retransmission of reliable information and enhances the quality of real time traffic like video which cannot rely on retransmission because of the large delays involved [29]–[35].

3. Layered source coding algorithms, such as interframe predictive coding which prevents severe degradation of picture quality [36]–[38].

In this chapter, we look into the second option of using FEC for cell loss compensation. The idea of using the FEC technique was until recently restricted to cases where burst errors are caused by noise and distortion of the transmission channel. Thus bursty erroneous periods are assumed not to be synchronized with the cell periods. In ATM networks however, the errors due to cell discard are generated on a cell by cell basis and are synchronized with the cell periods.

The following sections deal with a cell loss recovery mechanism that uses the FEC technique. This method consists of cell loss detection and lost cell regeneration. The major features of this technique are:

- (a) It can compensate for consecutive cell loss.
- (b) The average coding/decoding delay time can be reduced due to high speed processing.
- (c) Original cells remain unaltered when encoded. Thus, encoded cells maintain the standard cell format and they are handled in the same way in the ATM network.
- (d) Also, there is no revision of any information related to cell loss compensation within the ATM network.

Cell discard in ATM Networks

There are two major causes of cell discard in ATM networks. The first is random bit errors in cell headers. The second is buffer overflow in multiplex or cross-connect nodes. Cell headers are protected from bit errors by header error check (HEC) while

there is no protection against buffer overflow. Moreover since cell losses due to random transmission bit errors are also generated randomly, the quality degradation is small. On the other hand, since cell losses due to buffer overflow are generated in a consecutive manner, considerable quality degradation is caused.

1. Cell discard due to bit errors

Because ATM networks process cells by their headers, bit errors in the header fields can cause cell discards. The fifth octet in the header field of an ATM cell is used for header error control (HEC) and is calculated from the other four header octets. HEC and cell delineation functions are realized with a (40,32) CRC field. This HEC can correct single bit errors and detect two bit errors. Therefore, the cell loss rate and cell misdelivery rate are greatly reduced and in fact in the previous chapter it was shown that these could be neglected.

2. Cell discard due to buffer overflow

When there is signal congestion in cell multiplexers, some cells are discarded due to buffer overflow. Unlike HEC, there are no recommended standards for buffer overflow protection. However, if a cell loss compensation method is employed, transmission quality can be greatly improved [32].

5.2 Coding for cell loss recovery

General Considerations

Since cells are sent in increasing order of SN (Sequence Number), the decoder identifies missing cells by gaps in the arriving cell SNs, either upon the arrival of a cell or after a pre-determined waiting time. Thus a cell which is missing can be considered as a sequence of bit erasures, i.e., unreliable bits whose location is known. There are several FEC codes that efficiently correct erasures, most notably the Reed-Solomon codes. In fact, [35] suggests using Reed-Solomon codes for consecutive cell loss compensation, in HDTV applications. However, the process of cell loss recovery

described is quite involved. This is because the error correction mechanism proposed consists of interleaving the encoder output in a matrix and then adding the Reed-Solomon codes. This is then transmitted to the decoder where first a de-interleaver sorts the interleaved cells and then passes it on to the decoder. A similar approach may not be appropriate in our case as the hardware implementation of Reed-Solomon codes to deal with data rates of 150 Mbps is extremely complex even with current VLSI technology. This could potentially lead to data processing bottlenecks. Since one of the main objectives is to simplify the decoder complexity, the use of a Single Parity Check with erasure correction for cell loss compensation seems promising [31], [33]. A comparison between the efficiency of Reed Solomon (RS) codes and Single Parity Check (SPC) codes is however also considered, purely for implementation purposes.

The rate at which cells are lost depends on the rate at which the source emits them. Cells are more likely to be rejected because of buffer overflow as their rate of arrival at the buffer increases. Since coding increases the volume of traffic entering the network, the cell loss rate is likely to increase after the control bits are added. Thus, the lower the code overhead, the lower the increase in cell loss rate would be. However, the use of control bits should bring the cell loss rate below the loss rate when no coding is used. The ratio of cell loss rate after decoding to the rate when no coding is used is denoted as the *loss ratio*. This must be smaller than 1 for the code to be useful.

Requirements of the coding scheme:

The coding scheme adopted must

1. Deal with long sequences of lost bits (i.e. bursts of missing cells).
2. Correct occasional bit errors.
3. Result in low probability of undetected errors.

Since cells are transmitted at very high speeds, coding must be done with a minimum

of processing per cell, preferably by low complexity hardware. Both coding and decoding must add minimal delay and should require only a minimum amount of storage for operation.

Based on the above considerations, one possible technique is to group cells into blocks of a pre-determined size and then add to each block, check cells containing error control bits. The number of error check cells and their construction determines the maximum number of cells that can be recovered.

We therefore group the HP cells emitted from the different multiplexed video sources into a matrix of a fixed dimension. Error control coding is then done on a column-by-column basis, while transmission is row-wise.

5.3 Coding scheme options

1. Add error check bits for each Video_PDU as it comes out of the encoder and then transmit it.

This obviously preserves the Video_PDU integrity. However, as the length of each Video_PDU is geometrically distributed, we need to add a varying number of check bits to each varying length Video_PDU, in order to have the same error checking capabilities. This automatically increases the encoder/decoder complexity and the overhead involved is high.

2. Have a single Video_PDU for each row of the matrix.

In this case the maximum number of HP cells per row would be 12 and the minimum 1. If error coding is done for constant length columns then a varying number of dummy cells needs to be inserted. This not only increases the overhead but also makes the detection of dummy cells, if and when losses occur, more complex.

3. To have a matrix of a particular size and start filling it up with HP cells after the SN & SNP fields are tagged to them.

The FBM field in the Video_PDU would indicate to the decoder when a frame ends and when the next one begins. Loss of control information due to the loss of the cell in which it resides can easily be detected by extrapolating from the control information of the next Video_PDU or by simply using the error check mechanism to regenerate the lost cell, as explained later.

Any single matrix would consist of HP cells from N different homogeneous sources that are multiplexed together onto a common output link. The arrivals from these different sources are synchronized.

Because of the following advantages it was decided to adopt the last mentioned coding scheme.

1. The significance of the Video_PDU structure is kept intact. This helps immensely in delineating cell losses, if and when they occur, at the receiver.
2. Overhead due to dummy cells is totally eliminated.
3. The constant length of each row makes it easier to devise a uniform error correcting scheme.

5.4 Coding structure

A detailed description of the coding scheme is given in this section.

Figure 5.1 Encoding Matrix Structure

Each matrix consists of N video cells arranged in M rows. Fig 5.1 shows the encoding matrix.

Each row of cells is terminated by a CLD cell which is used for cell loss detection. Each column has a set of duplicated Parity/RS cells (depending upon whether the columns are all Parity encoded or are all Reed-Solomon encoded) at its bottom. Cell loss recovery is then done using these Parity/RS cells. A set of duplicated CLD-P or CLD-RS cells is located at the rightmost bottom corner for CLD cell loss recovery.

Therefore, $N+1$ is the row length and $M+2$ is the column length of the matrix. One lost cell in each column can be recovered using Parity checks and two lost cells using the Reed-Solomon codes. Hence up to N or $2N$ discarded consecutive HP video cells in a matrix can be regenerated by this technique, depending on the encoding scheme used.

The values of N and M are 8 and 54 respectively. The choice of N is dictated by design parameters (as explained later) while the value of M was arrived at as a compromise between the overhead used and the performance achievable.

The reasoning behind using the CLD cells and duplicated layers of error control cells is provided next.

The idea of using CLD cells for cell loss detection is largely driven by the fact that there are inherent dangers involved in just relying on the SN check at the decoder end. Consider, for example, two consecutive Video_PDUs each having 6 HP cells, marked 1 to 6. It may happen that cells 4–6 from the first Video_PDU and cells 1–3 from the second Video_PDU are lost. At the receiver end it will appear (erroneously) as if there were no losses if only a SN check is used.

A duplicated layer of Parity cells is proposed because the encoding matrix encodes HP cells which carry not only important video information but also more important control information required by the decoder for reconstructing any lost cells. Moreover, if we were to have just a single level of error check cells, and if one of these cells itself were to get lost then our cell compensation process would fail at least for that column. Having two error check cells per matrix column ensures that even if the cell loss rate is p (assuming cells in a column are lost independently as explained later), then the probability that both the error check cells in a column are lost is approximately p^2 and hence considerably smaller. Performance evaluation was also done using a single layer of Parity cells but the results obtained were significantly worse than those obtained using a duplicated layer. Therefore, the increased overhead is justified as it provides improved performance.

Complexities added due to the above scheme:

1. Coding and decoding delay.

Encoding delay is almost negligible because video cells do not have to be buffered. The encoder has to determine the CIPs (Cell Identification Patterns, explained later) and Parity/RS(56,54) cells. These will have to be stored, which involves memory requirements. At the decoder all the cells received have to be buffered to the end of each row. If discarded cells are detected then all the cells following the lost cells are buffered until the end of the matrix. It follows that when there are no video cell discards in a matrix the decoding delay for each cell in the matrix is a maximum of $[(N+1)/\rho] \times [\text{Cell Period}]$ where ρ is the link utilization factor. On the other hand, the delay is a maximum of $[(N+1)(M+2)/\rho] \times [\text{Cell Period}]$ when there is even one cell discard in the matrix. For an average link utilization factor of $\rho=0.742$ and the given matrix size the decoding delay is 0.03 ms when there is no video cell lost and is 1.922 ms when even one video cell is discarded.

Similarly, the delays involved for different values of ρ can be obtained. However, we are not overly concerned about delay as video-on demand services are not very delay sensitive.

2. Delineation of fields

Video_PDU delineation can very easily be done by using the FBM/FN/LGN fields present in the Video_PDU. Even if the cell containing this control information is lost it can be easily extrapolated from the control information present in the next Video_PDU. This is possible because decoding is done row by row. Since the FBM indicates the beginning of each frame and since each frame has a fixed number of Video_PDUs, the loss of any Video_PDU can be detected by simply detecting a break in the LGN field.

5.5 Error control options

An outline of the different options available for error encoding is presented next.

1. Have double layer of Parity cells for each column of video cells and two CLD-P cells for the CLD cells column.

In terms of hardware implementation this offers the best option, as Single Parity Check (SPC) codes are very easy to implement and the processing speeds are much faster.

2. Have a Reed-Solomon (RS) code for all the columns.

Reed-Solomon codes are extremely powerful, error detecting/correcting codes and are superior in performance to SPC, as will be borne out later with respect to improvement in cell loss rate. However the hardware implementation of Reed-Solomon codes to deal with data rates of 150 Mbps is extremely complex and needs to be done using custom built VLSI.

3. Have a double layer of Parity cells for each column of video cells and two CLD-RS cells for the CLD cells column.

This is essentially a compromise between the above two options. In the whole encoding matrix the last column is really the most important one, in terms of reconstructing lost video cells at the decoder end. We therefore use the more powerful Reed-Solomon code for this column alone. The attractive feature about this option is that the hardware complexity involved with Reed-Solomon codes is restricted to the most important column of the matrix. This does not affect the loss rate for the cells in the other columns but ensures that the loss performance for the CLD cells column is significantly improved.

Henceforth, the above three options will be referred to as Options **1**, **2**, **3** respectively. In the following Option **3** is described in detail. Performance evaluation however, will be done for all the three options.

5.6 Encoding/Decoding Fundamentals

1. Single Parity Check (SPC) codes

Consider a video source that sends a sequence of cells to a destination over the ATM network. The source sends cells of two types: video and overhead. The source groups the cells transmitted into columns of length M , where M is a pre-determined number. After all the video cells of a particular column have been transmitted, the error control bits for that column are computed and the source sends those cells as the $(M+1)$ st and $(M+2)$ nd cells of the column.

Encoding and decoding of SPC codes

The parity cell is computed in the following manner. Consider just one column and assume for the sake of simplicity that the cells of the column are numbered $1, 2, \dots, M$ and that the two identical parity cells are numbered $(M+1)$ and $(M+2)$. Denoting by c_{ij} the value of the i th bit of cell j ($1 \leq j \leq M$), the parity cell's i th bit $c_{i,M+1}$ is set so as to complete to even parity the sequence of bits $c_{i,1}, c_{i,2}, \dots, c_{i,M}$, for example

$$c_{i,M+1} = \left(\sum_{j=1}^M c_{ij} \right) \bmod(2) \quad i = 1, 2, \dots, m$$

where $m = \text{length of a cell} = 48 \text{ bytes}$

$c_{i,M+2}$ is identical to $c_{i,M+1}$

The parity cell for a column is generated by an encoder consisting of m *exclusive-or (XOR) gates*, each of which has its output connected to one of its input ports as shown in Fig 5.2.

Figure 5.2 Parity cell generation for a column

Cells are given one at a time, to the encoder, with bits 1, 2, . . . m applied to the input ports of gates 1, 2, . . . m respectively. At each clock pulse, the decoder output port i represents the modulo 2 of bits in position i of the previous and bit i of the current cell.

Since there are $N+1$ ($= 9$) columns in the encoding matrix, there are 9 such sets of connections as shown in Fig 5.2, each representing the operation for a single column. The CLD cell is generated for a particular row and then sent out immediately after the video cells of the row have been sent out. Transmission is done row by row and furthermore there is no need to store the whole column of cells at the source. Thus, immediately following its "contribution" to the encoder, a cell can be sent over the network. Following the application of the M -th cell, the encoder's m output ports of that particular set contain the parity cell, which then is transmitted.

The decoder is similar in structure to the encoder. As cells arrive row-wise, they are buffered at least until the end of the row. Using the CLD cell, any video cell loss for that

particular row is detected. If cell loss has occurred, the decoder identifies the location of the missing cell and considers these missing cells as erasures. Once cell loss is detected, all the cells following that are buffered until the end of the matrix. Then error-correction is done on a column-by-column basis to regenerate the lost cells.

2. Reed-Solomon (RS) codes

Before describing the encoding/decoding procedure using Reed-Solomon codes, a brief summary of these codes is presented [39], [40]. Each cell in the matrix is 48 bytes long.

Fundamentals

Let:

w = Number of bits per symbol; each symbol $\in GF(2^w) = 6$

n = Total number of symbols, including check symbols = $M+2 = 56$

k = Number of information symbols = $M = 54$

m = Degree of generator polynomial = number of check symbols = 2

d_{min} = Minimum Hamming distance of the code = $m+1 = 3$

$G(x)$ = The code generator polynomial

$D(x)$ = Data polynomial

$W(x)$ = Write redundancy polynomial = $[x^m \cdot D(x)] \text{ MOD } G(x)$

$C(x)$ = Transmitted codeword polynomial = $x^m \cdot D(x) + W(x)$

This code $RS(n,k)$ can correct

$$\frac{d_{min} - 1}{2} = 1 \text{ symbol error without erasure correction}$$

$$d_{min} - 1 = 2 \text{ symbol errors with erasure correction}$$

The generator polynomial of a Reed-Solomon code is given by:

$$G(x) = \prod_{i=0}^{m-1} (x + \alpha^{ma+i})$$

where α is a primitive element over $GF2^6$ (Galois Field), m is the number of check symbols and m_0 is an offset, often zero or one. In the interest of simplicity we take m_0 equal to zero. As $m=2$, the generating polynomial takes the following form:

$$\begin{aligned} G(x) &= \prod_{i=0}^1 (x + \alpha^i) \\ &= (x + \alpha^0)(x + \alpha^1) \\ &= x^2 + (\alpha^0 + \alpha^1)x + \alpha^1 \\ &= x^2 + \alpha^3x + \alpha^1 \end{aligned}$$

Encoding and Decoding of Reed-Solomon codes

The encoding and decoding processes for Reed-Solomon codes are beyond the scope of this thesis. The reader is directed to [39] for a detailed account of these codes. Encoding is typically done using an internal-XOR shift register with symbol-wide data paths (6 bits in our example) in order to implement the form of the generator polynomial shown above.

The decoding of Reed-Solomon codes is an extremely complex process, both in terms of hardware complexity and computational intensity. Without going into the details of the decoding process, we shall outline the procedure which consists of the following steps

1. Compute syndromes.
2. Calculate the coefficients of the error locator polynomial.
3. Find the roots of the error locator polynomial. The logs of the roots are the error locations.
4. Calculate the error values.

Since in our case the error locations are known because of the CLD cells, we need to perform only the last step which involves calculating the error values. Once the error locations vector is known, the syndrome equations become a system of linear equations with the error values as unknowns. Determinants can be used to solve the syndrome equations, particularly since the number of errors is low.

The above encoding/decoding process described is for just one column of the matrix which is Reed-Solomon encoded. If we were to use Option 2) whereby the whole matrix is RS(56,54) encoded, we would need $N+1$ ($= 9$) such processes running in parallel.

5.7 Cell loss recovery scheme

A outline of the approach, using Option 3, is as follows. First, if the discard of a CLD cell is detected by a gap in the SN-CLD, then using the CLD-RS cells the CLD cell is regenerated. Video cell loss is then detected using the now regenerated CLD cells. Dummy cells are inserted at those positions where lost cells are detected. Finally, using the Parity/RS(56,54) cells, located at the end of each column, the lost cells are regenerated and these regenerated cells replace the dummy cells.

Fig 5.3 and Fig 5.4 give an outline of the process. A detailed description of the processes is given in the following sections.

Figure 5.3 Cell loss recovery mechanism a)

A : Detected lost cells (dummy cells are inserted)

B : Replace the dummy cells with regenerated cells.

Figure 5.4 Cell loss recovery mechanism b)

5.8 Lost CLD cell detection and regeneration

Structure

The following is the structure of a single CLD cell (Fig 5.5)

The SN field (4 bits) and the CRC (10 bits) field are extracted from the incoming HP

Figure 5.5 CLD cell structure

video cells. These $4+10=14$ bits form the CIP — Cell Identification Patterns. The CIP's of all the video cells in each row are now loaded onto the payload of the CLD cells. Also, there is a 8 bit SN-CLD field which acts as a sequence number for the CLD cells and detects the discards of the CLD cells. The period of the SN-CLD has to be longer than the column length of the encoding matrix. By having a 8 bit SN-CLD field matrices with column length up to 256 can be encoded.

A (384,368) CRC is then calculated over the CIP's and the SN-CLD in each CLD cell, and added as part of the last two octets of the CLD cell payload. This CRC can correct single bit errors in the CIP and SN-CLD field of the cell payload. This reduces the influence of bit errors on cell loss detection.

It is however possible that a CIP of a particular SAR_PDU is identical to the CIP of another SAR_PDU. The probability of this happening is $(1/2)^{14}$

Therefore, an algorithm to prevent mismatch has to be determined. For this the following is suggested.

The payload of the CLD cell is $45 \times 8 = 360$ bits. Each CIP is 14 bits long. In order to prevent mismatch each CIP is triplicated and at the decoder the CIP for a particular cell is taken based on a majority rule. The probability that all the triplicated versions of a CIP are different is negligibly small.

The triplicated CIP of each cell occupies $14 \times 3 = 42$ bits. It's therefore possible to have $\lceil \frac{360}{42} \rceil = 8$ video cells in a particular row of the matrix. The last 24 bits of the CLD payload are padded with '0' bits.

Thus, N is 8 in the matrix. The increased bandwidth required for having this simple majority rule, to prevent mismatch, is about 1.5 MHz. The reason for choosing M as 54 was explained earlier. The encoding matrix is therefore of size 9×56 .

The cell header tagged onto the CLD cells at the ATM layer is similar to that of standard cells. The payload however is not compatible with any AAL type. This however should not pose any problem because the CLD cells are used only for cell loss recovery and are removed before any AAL processing of video cells takes place at the receiver end.

Operation

The Reed-Solomon codes used, enables up to two lost CLD cells to be recovered. If a CLD cell loss is detected from a gap in the SN-CLD, the lost CLD is regenerated before the video cell loss detection occurs. The regeneration of discarded CLD cells is done using the CLD-RS cells and in the same manner as the regeneration of the lost video cells. This is explained later. The above decoding process however is aborted if more than two CLD cell losses are detected; and all the cells are passed onto the decoder.

The structure of the CLD-RS cells is as shown below in Fig 5.6.

Figure 5.6 CLD-RS cell structure

The last two octets of the CLD-RS cell again consists of a (384,368) CRC field for bit error correction/detection.

5.9 Video cell loss detection and regeneration

Cell loss detection

CLD cells are generated and added to the video cell stream in the encoding equipment. Immediately after all the video cells in one row are sent out, the corresponding CLD cell is sent out. Cell losses are detected row by row in the decoder. When a row is received by the decoder, the SN and CRC fields of the cells (CIP value) are compared against the CIP values of the CLD cell that terminates the row. Lost cell positions can be detected in this manner, instead of a simple SN check. Note that all the CIPs are first chosen based on a simple majority rule and then compared against the CIP value of the corresponding video cell.

If all the CIP pairs match, then no cells have been discarded in that row. On the other hand, if some CIP value fails to match any CIP field of the received cells, cell discards in the corresponding positions are detected and dummy cells are inserted in the required positions. For example, if the *i*th CIP value in a CLD cell does not match, the *i*th cell in the corresponding row is confirmed to be a discarded cell. Fig 5.7 shows a case where cell 3 is found to be discarded and is replaced by a dummy cell.

Figure 5.7 Cell loss detection

Delineation of the matrix rows is indispensable for cell loss detection. Therefore, each row is terminated by a CLD cell, and each CLD cell in turn is terminated by a CRC field.

Lost cell regeneration

As indicated in the above figure, cell losses once detected lead to dummy cell insertion at the positions of lost cells to restore the proper location of the other video cells. The dummy cells are replaced by regenerated cells in the lost cell regeneration process.

Lost cells are regenerated using parity cells. The parity cell format is as shown in Fig 5.8. Instead of the SN and SNP field present in the header of the other video cells we have a SN-parity field (8 bits). Parity cell losses are detected by gaps in the SN-parity. The information field of the parity cell is calculated by modulo 2 addition of the information fields of the cells in that column. After having determined these fields the

LI field is filled up and then a (384,374) single error correcting CRC is calculated over the information field of the parity cell.

Figure 5.8 Parity cell structure

Operation

The regeneration of discarded cells using parity cells is as shown below in Fig 5.9.

Figure 5.9 Regeneration of lost video cell

Each column of the matrix (except the last, where up to two dummy cells can be present) is first checked to determine the number of dummy cells in the column. If the number is more than one then the regeneration process is stopped. Otherwise the discarded cells are regenerated to replace the dummy cells.

Discarded cells are regenerated column by column. All received cells in one column (including the parity cells) are modulo 2 added. If no cells are missing in the column then the result is 0. If one cell is discarded in the column, the summation result is the value of the discarded cell. This regenerated cell replaces the dummy cell. The SN and CRC field of the regenerated cell is copied from the CIP field of the CLD cell in that row. Then the CRC-10 of the information field is calculated. This completes the regeneration process.

5.10 Performance Analysis

Having described the compensation scheme, we evaluate the performance obtained by using the three different encoding options, mentioned earlier. Numerical results and graphs will however be discussed in the next chapter.

In the following we assume that cells are 48 octets (384 bits) long and that two Parity/RS(56,54) cells are added to each column of M video cells.

Assuming cells in a column are lost independently, because they are interleaved, with probability p . Thus at the decoder the number of lost cells in a column is a random variable, having a binomial distribution, with parameter $M+2, i, p$.

$$\begin{aligned} P\{i \text{ lost cells in a column}\} &= b(M+2, i, p) \\ &= {}^{M+2}C_i p^i (1-p)^{M+2-i} \end{aligned}$$

The performance evaluation is presented next.

Option 1 — Only SPC code.

In this case a lost cell can be recovered if and only if it's the only lost cell in the column. The average number of lost cells in a column after decoding is given by

$$E\{L\} = \sum_{j=2}^{M+2} j b(M+2, j, p) = (M+2)p - (M+2)p(1-p)^{M+1}$$

The cell loss rate after decoding, P_{dec} is given by

$$\begin{aligned} P_{dec} &= E\{L\}/(M+2) = p - p(1-p)^{M+1} \\ &\simeq p \left[1 - e^{-(M+1)p} \right] \quad \dots 5.1 \end{aligned}$$

Option 2 — Only RS code.

The Reed-Solomon code used in our case is the RS(56,54). This code is capable of regenerating up to two lost cells in a column, assuming erasure correction, as explained earlier.

Therefore the cell loss rate after decoding, P_{dec} in this case is given by

$$\begin{aligned} P_{dec} &\simeq \sum_{i=3}^{M+2} {}^{M+2}C_i p^i (1-p)^{M+2-i} \\ &\simeq {}^{M+2}C_3 p^3 \quad \dots 5.2 \end{aligned}$$

Option 3 — Combination of SPC and RS code.

This option ensures that the most important cells in the matrix for cell reconstruction — the CLD cells, have a much lower loss probability compared to the video cells. The loss rate for all the columns except the last one remains the same as Option 1 and is given by

$$\begin{aligned} P_{dec-video} &= E\{L\}/M + 2 = p - p(1-p)^{M+1} \\ &\simeq p \left[1 - e^{-(M+1)p} \right] \quad \dots 5.3a \end{aligned}$$

The loss rate for the CLD cell column is given by

$$\begin{aligned} P_{dec-CLD} &\simeq \sum_{i=3}^{M+2} {}^{M+2}C_i p^i (1-p)^{M+2-i} \\ &\simeq {}^{M+2}C_3 p^3 \quad \dots 5.3b \end{aligned}$$

The performance evaluation, using both analytical and simulation results, for the above three options will be presented later. However, to gain an appreciation of the derivation, consider the following possibilities that might occur in Option 1). We shall perform the calculations assuming for the moment $p=10^{-9}$

1. One Video cell or one Parity cell is lost in a column — The coding scheme will always work for this possibility. The lost video cell will be first detected by the CLD cell for that row and using the parity cell for that column the cell regeneration will occur at the decoder. If only the Parity cell is lost in a column we simply use the other Parity cell present. The probability of this happening =

$$\begin{aligned} P\{1 \text{ lost cell in a column}\} &= {}^{M+2}C_1 p^1 (1-p)^{M+2-1} \\ &= 5.6 \times 10^{-8} \end{aligned}$$

2. One Video cell and one Parity cell are lost in a column — The video cell loss is detected by the CLD cell and the parity cell loss is detected by the SN-Parity field. Hence using the other parity cell present in the column, the video cell can be regenerated. Probability of occurrence =

$$\begin{aligned} {}^M C_1 p^1 (1-p)^{M-1} \times {}^2 C_1 p^1 (1-p)^1 \\ = 1.08 \times 10^{-16} \end{aligned}$$

3. 2 Video cells are lost in a column — This cannot be compensated for (except for the CLD cells column) and the compensation process is aborted for that column. Probability of occurrence =

$$\begin{aligned} {}^M C_2 p^2 (1-p)^{M-2} \\ = 1.43 \times 10^{-15} \end{aligned}$$

Thus it can be seen that the probability the coding scheme will not work is negligible and is almost 3 orders of magnitude less than the required HP cell loss rate itself.

Chapter 6

Performance evaluation of an ATM multiplexer

6.1 Introduction

The efficacy of the cell loss compensation scheme is evaluated next by considering the performance of an ATM multiplexer.

In the ATM network the bit streams of different sources are multiplexed for further transmission and switching. In the multiplexing of VBR video sources the network queues contribute only marginally to the multiplexing process and can be omitted while describing the process [41]. Therefore, no queueing theory is required and the knowledge of the bit rate distribution of the video sources is sufficient to describe this multiplexing process.

Therefore, if an abstraction is made of the queues present in the network, and the bit streams of the different sources are instantaneously added, the resulting bit stream will be the statistical sum of the individual processes. This process is referred to as statistical multiplexing [42], [43].

After getting encoded in the encoding matrix (with negligible delay), the HP cells are passed to the ATM layer and are appended there by the LP cells belonging to the same Video_PDU. The whole Video_PDU is then buffered prior to the multiplexing. This process is repeated for all the N video sources that are to be multiplexed onto the common link.

Source and Node model

The model of the statistical multiplexer used in our case is a single-queue-single server system with loss priority implemented through the push-out scheme (explained later).

H : HP Video cell
 L : LP Video cell
 O : Overhead (also HP)

Figure 6.1 Model of the ATM multiplexer

Cells are generated asynchronously from the N video encoders, stored in the queue according to the push-out procedure and served by the channel with FIFO policy (Fig 6.1). The transmission over the channel is assumed to be slotted. The multiplexer is defined by the buffer size K (cells) and the link rate C (Mbits/s). The link rate is 150 Mbits/s and the cell size is the ATM cell size of 53 octets. Hence the cell transmission time (i.e. the service time for a cell) is fixed at 2.83 μ s.

Each of the sources in our problem is identical and can be characterized by (Fig 2.2):

B_p (Mbits/s), peak bit rate (= 35.61 Mbits/s)

B_m (Mbits/s), mean bit rate (= 11.12 Mbits/s)

L (cells), mean burst length — the average no: of cells in each Video_PDU (= 5)

The sources are statistically independent, homogeneous and characterized by the same values of traffic descriptor (B_p , B_m , L)

6.2 Handling different classes of priorities

Cell loss probability as mentioned before, is one of the major performance issues in ATM networks and therefore the *loss priority mechanisms* used to decrease the loss of HP cells shall play an important role in network optimization. Several selective discarding algorithms have been proposed in order to offer different cell loss rates to different classes of cells [44]-[46]. These include primarily:

1. Partial Buffer Sharing (PBS)

A1 : Arrival rate for Class 1 (HP) cells

A2 : Arrival rate for Class 2 (LP) cells

Figure 6.2 Partial Buffer Sharing (PBS)

In this scheme (Fig 6.2) a threshold $S2$ is assigned and cells of the LP class are accepted in the buffer only if the current queue length is smaller than $S2$. As a result a

buffer of size K is partitioned into two parts: the first one of size S_2 where all cells can be stored and the second one of size $S-S_2$ where only HP cells can be stored.

Advantages and disadvantages of this scheme

The fixed buffer partition has the disadvantage of preventing LP cells from using the full buffer even when no HP cells are present. The performance of this scheme depends on the threshold S_2 , whose optimum value is a function of S and of the fraction of cells in each priority class. Better performance can be achieved when S_2 dynamically changes with the traffic in the network. Cell sequencing is easily managed by having a buffer management scheme based on FIFO policy. In this study, cell sequencing is not so much a problem as the difficulty in determining an optimal threshold; since very little is known about the variation in traffic.

2. *Push-Out scheme*

A dynamic sharing of the full buffer amongst both HP and LP cells is involved here, Fig 6.3. Any cell can be stored if buffer space is available but a LP cell can be 'pushed-out' by a HP one.

This scheme guarantees that even if LP cells are stored without restrictions, no HP cell is lost unless the buffer contains only HP cells. The selection of the cell to be discarded is controlled by the replacement strategy as shown in Fig 6.4.

A1 : Arrival rate of Class 1 (HP) cells

A2 : Arrival rate of Class 2 (LP) cells

Figure 6.3 Push-out scheme

Advantages and disadvantages of this scheme

One problem of this scheme becomes obvious from the following figure. The cell sequence integrity, if it is to be maintained, would require a complex buffer management technique.

In our case this would not pose a great deal of problem as cell integrity would be handled by the VCI of that particular cell and all we need is a resequencer at the receiver end to resequence cells belonging to a particular Video_PDU.

Figure 6.4 Replacement strategy for Push-out scheme

Of the replacement strategies shown above we adopt the LIFO technique, i.e. if there is an incoming HP cell to the queue, the last LP cell is shunted out of the queue and replaced by the new incoming HP cell. The simulation procedure can be simplified by accessing the buffer through pointers and organizing it in a FIFO list. Then the elimination of a LP cell to accommodate the new HP cell arrival would amount to a simple record elimination and not involve any stored memory shift.

In general, the push-out scheme is technically intricate to implement and PBS is currently favored for priority implementation in ATM networks. However, the push-out scheme has been shown in other applications to be comparable in performance to the PBS scheme, which is easier to implement [46].

We make use of the above fact and the flexibility offered by the push-out scheme in adopting it as the priority mechanism in our analytical and simulation work.

6.3 Queueing model of the multiplexer

We have N independent and identical sources connected as shown below, in Fig 6.5.

Source

Figure 6.5 Queueing model of the ATM multiplexer

Each source generates a variable length Video_PDU every $190.5 \mu\text{s}$. The MUX switch samples each input source in an interval of T seconds, in a multiplexer cycle (the value of T must take into account the peak bit rate of the sources and the possibility of overhead: CLD, Parity and CLD-RS cells from the encoding matrix). After collecting one cell (video or overhead) from the input it passes it onto the MUX buffer. After this it samples the next source and passes it onto the common buffer and likewise for all the N sources, until all the cells in a particular MUX cycle are sampled. The idea behind interleaving cells from different sources is driven by the following considerations

1. By interleaving, burst cell losses are reduced as lost cells are distributed over several sources. This reduces the picture quality degradation due to burst errors for any single source.

2. The gain obtained by coding in terms of an improved cell loss rate would be lost, if there was a high correlation between the cell losses for a particular video source [32]. Interleaving cells from different sources ensures that the correlation present within a source is removed.

3. When cells are interleaved, the dropped cells due to buffer overflow are more likely to belong to different sources. Therefore in the HP encoding matrix it can be assumed that the lost cells in a column follow an independent binomial distribution. This supports the assumption made in the previous chapter.

The maximum allowable buffer size, K_{max} is dictated by the maximum allowable delay - t_{max} and is given by the relation [47], [48]

$$K_{max} = \frac{C \times t_{max}}{P}$$

where P is no. of bits / cell = 53×8

For instance, a maximum allowable delay of 2.5 ms gives us a figure of $K_{max} = 884$ cell spaces.

The number of sources depends on the average transmission rate of the sources and the utilization that is desired on the transmission link.

$$N = \frac{C \times \rho}{B_m}$$

ρ = Link Utilization

C = Link capacity (150 Mbps)

B_m = Mean bit rate/source (11.12 Mbps)

We evaluate the performance of the ATM multiplexer using both analysis and simulation.

6.4 Analytical and simulation approaches for evaluating ATM multiplexer performance

Introduction

The analytic approach provides us with an elegant and simple way of determining the loss rate for different number of sources multiplexed and for different buffer sizes. The object of extreme significance here is the computational simplicity (and its consequent effect on speed) involved in this approach, making it possible to do analysis for even extremely low loss rates (e.g. $< 10^{-12}$), which is virtually impossible using computer simulation. The computational complexity and the lack of an explicit relationship between simulation parameters and results are the two major drawbacks of the simulation approach. It is therefore only used to verify the analytical results and obtain a rough understanding of the effect of simulation parameters on the performance.

The details of the approach are enumerated in the following parts of the section.

The analysis is done on a MUX cycle basis, with each cycle being $190.5 \mu\text{s}$ long. Appendix A shows the probability distribution (pd) for the length of the Video_PDU, for each video source (refer Fig 2.2). Denoting the number of cells from the i th source in a single multiplexer cycle by I_i :

$$I_i = \text{No : of cells/source/MUX cycle} \\ \text{with range } (1, 16)$$

If we multiplex N independent sources, the pd of the number of cells arriving in a multiplexer cycle becomes simply the convolution of the pd's of the individual sources [49]. Denoting the number of cells from N multiplexed sources by I_N we have

$$I_N = \sum_{i=1}^N I_i$$

$$\text{where } I_N = \text{No : of cells/MUX cycle with range } (N, 16N)$$

Since the I_i 's are identical and independent (iid) random variables, the pd of I_N can be obtained by convolving I_1 with itself N times. Fig 6.6 illustrates the pd of the multiplexed

sources, for $N=10$ (actually the envelope of the discrete distribution is shown). The pdf of I_N will be denoted by L_N .

Figure 6.6 The pf of $N=10$ multiplexed video sources

However, some of the cells arriving in each multiplexer cycle are lost due to buffer overflow and some cells are transmitted from the buffer in each cycle. Our objective here is to determine the steady state loss probability.

The input buffer of the multiplexer is of size K . The service rate for each multiplexer cycle is fixed since the cell transmission time of an ATM cell 53 bytes long on a 150 Mbit/s link is $2.83 \mu\text{s}$. Therefore, in one MUX cycle a maximum of $D = \left\lceil \frac{190.5}{2.83} \right\rceil = 67$ cells can be serviced.

The analytic approach used here was suggested by Kaye [50] and is as follows.

At the beginning of the 1st MUX cycle, we have a distribution specified by L_N coming into the head end of the ATM multiplexer. Let $BC0$ denote the distribution of

the buffer contents before the service has begun for a particular multiplexer cycle. For the 1st cycle, BC_0 is just an unit impulse, as the buffer is empty. Out of the incoming distribution a maximum of D cells can be serviced. If the distribution of the buffer contents at the end of a multiplexer cycle is denoted by BC_1 then for the first MUX cycle BC_1 is obtained by shifting L_N to the left, by D cell spaces. There will be an impulse at $BC_1(0)$ to denote the fact that if less than or equal to D cells came into the MUX at the beginning of the MUX cycle then there would be none left in the buffer at the end of the cycle.

The buffer can however hold only a maximum of K cells. Hence, if the range of BC_1 is in excess of K there will be cell losses due to buffer overflow. The complement of the cumulative distribution function of BC_1 up to K cells will give the loss probability for that MUX cycle. Let BC_2 denote the distribution obtained by truncating BC_1 to K cell spaces, i.e. distribution of the buffer contents at the end of the MUX cycle after cells in excess of K cell spaces have been dropped.

At the beginning of the 2nd MUX cycle, we once again have a distribution specified by L_N coming into the multiplexer. Therefore, BC_0 for this cycle should be represented by the sum of BC_2 (distribution at the end of the previous cycle) and L_N . Hence, BC_0 is simply the convolution of the latter two distributions. A maximum of D cells out of these is serviced and the remaining distribution is BC_1 . Once again the negative cumulative distribution of BC_1 upto K cells will give the loss probability for the 2nd MUX cycle. Having calculated the loss probability we truncate BC_1 to K cell spaces to get the distribution of BC_2 for the 2nd MUX cycle. The above process is repeated for several multiplexer cycles until, the loss probability approaches a constant value, representing the steady state. This provides us with the average loss probability.

For the initial part the priority handling scheme is neglected. Therefore, a HP cell is as likely to be dropped (in the event of buffer overflow) as a LP cell. This helps in determining the utility of the priority scheme.

Initially it is also assumed that there is no FEC overhead, but only video information, coming out of the encoders. The maximum number of sources that can be multiplexed is determined by the link utilization factor (ρ) and in order to ensure a stable system $\rho < 1$. In the case without FEC overhead, it works out that a maximum of $N=13$ video sources can be multiplexed. We therefore vary the number of sources that are multiplexed each time from $N=7$ to $N=13$. As can be seen from Fig 6.7 for $N=10$, K ranging from 25 to 100, the HP loss probability (=LP loss probability) attains a constant value within 5–10 MUX cycles.

Figure 6.7 HP/LP loss probability vs MUX cycles, $N=10$, without FEC overhead

Similar calculations were performed for other values of N , and in all cases the loss probability reaches a constant within 15–25 MUX cycles.

We next consider the case when overhead is added. The encoding matrix for HP cell loss protection is such that the FEC overhead is on an average approximately 12.5%, both for Option 1) and Option 2). We therefore add this constant overhead to I_N , to obtain I_N^* where

$$I_N^* = \text{No. of cells including overhead/MUX cycle} \\ \text{with range } (1.125N, 18N)$$

The technique adopted above is an approximation, because not only a constant percentage of overhead in each MUX cycle is assumed, but also in determining I_N^* , real number values obtained due to adding the overhead were rounded-up to the next higher integer. Following the same procedure as mentioned above, we get the variation of loss probability with respect to MUX cycles for different values of N and K with FEC overhead, excepting that now we have a distribution specified by I_N^* instead of I_N coming into the head end of the ATM multiplexer, at the beginning of each MUX cycle. Once again the HP loss probability (=LP loss probability) can be calculated. Fig 6.8 shows the variation for N=10 and K ranging from 25 to 100.

**Figure 6.8 HP/LP loss probability vs
MUX cycles, $N=10$, with FEC overhead**

Comparing Fig 6.7 and Fig 6.8, the following observations can be made:

The rate at which cells are discarded in an ATM multiplexer depends on its buffer size and the rate of cell arrival. Adding overhead cells to an arrival stream increases the traffic rate and the probability of buffer overflow; which is why for a particular value of N and K , Fig 6.8 shows a higher loss probability than Fig 6.7.

On the other hand using the overhead cells the decoder can reduce the HP cell loss probability by reconstructing some of the lost HP cells. Coding is beneficial only if the second effect is stronger than the first, resulting in a net drop in the cell loss probability. Using equations 5.1 and 5.2, we determine the net HP cell loss probability vs MUX cycles after decoding, for both Option 1) and Option 2). Fig 6.9 shows the variation

Figure 6.9 HP loss probability after decoding, (Option 1) for $N=10$

of the HP cell loss probability vs MUX cycles, after decoding for $N=10$ and Option 1). It can be seen that using FEC has reduced the HP cell loss probability from when no overhead was present.

Due to the fact that FEC is not used to reconstruct any lost LP cells and that adding the FEC overhead increases the arrival rate to the multiplexer buffer, the LP cell loss rate deteriorates compared to when there was no overhead. The quantitative effect of this will be dealt with later.

Similar graphs were obtained for other values of N (ranging from $N=7$ to $N=11$), both for Option 1) and Option 2), and in all cases except $N=11$ for Option 1), there is a significant improvement in the HP loss probability because of adding the FEC overhead, thereby confirming the utility of the coding scheme. The above calculations were also carried out starting with intermediate MUX cycles (i.e., assuming different values for the initial buffer contents) and the values of average loss probability obtained were the same.

Because of the presence of the MUX switch, consecutive cell arrivals to the common multiplexer buffer, can be considered to be uncorrelated for a single video source. Hence the simplest way to describe the cell loss behavior of an ATM multiplexer would be the long-term average cell loss rate P_L [51]. Therefore we shall henceforth concentrate on the average HP/LP loss probability.

Cell loss rate results

To find out the variation of the average loss probability with respect to the buffer size, for different values of N we perform the calculations according to the method mentioned above for 50 MUX cycles to ensure that the loss probability reaches its average value. We then have the variation of the average HP/LP loss probability with respect to the buffer size for different values of N , for the following cases: without FEC overhead— Fig 6.10, with FEC overhead— Fig 6.11, after decoding using Option 1)— Fig 6.12 and after decoding using Option 2)— Fig 6.13.

Figure 6.10 HP/LP loss probability without FEC overhead.

Figure 6.11 HP/LP loss probability with FEC overhead.

Figure 6.12 HP loss probability after decoding (Option 1)

Figure 6.13 HP loss probability after decoding (Option 2)

Figure 6.14 HP loss rate improvement factor (Option 1)

Figure 6.15 HP loss rate improvement factor (Option 2)

As mentioned earlier, adding the FEC overhead results in an increased loss rate for both HP and LP cells (compare Fig 6.10 with Fig 6.12). However at the decoder, since it is possible to reconstruct some of the lost HP cells the HP loss rate decreases overall compared to when no overhead was used (Fig 6.14 and Fig 6.15) (except for $N=11$ and Option 1)).

The LP loss rate with FEC is however worse than when no overhead was present. By reading values from Figs. 6.10 and 6.11 for a particular value of N and K , the deterioration factor in the LP loss rate can be calculated. This is reflected in Fig. 6.16 where it can be seen that there is an appreciable deterioration in the LP loss rate. Consider for example $N=10$ $K=100$, there is almost a deterioration of 4 orders of magnitude.

Figure 6.16 LP loss rate deterioration factor due to adding FEC overhead

2 OF 2

PM-1 3 1/2" x 4" PHOTOGRAPHIC MICROCOPY TARGET
NBS 1010a ANSI/ISO #2 EQUIVALENT

Confirmation by simulation

In the above figures the relationship between the buffer size and the cell loss rate is represented by straight lines in linear/logarithmic graphs. From this the cell loss rate for larger buffers can be predicted. Computer simulations are carried out for a range of small buffer sizes and linear extrapolation is then used to calculate the loss rates for larger buffer sizes. The above approach is also driven by the observations during computer simulations that when the buffer size was large, accurate results could not be obtained as there were not enough cell discard samples to get precise results.

N, Buffer Size	Probability of cell loss	
	Analytical Results	Simulation results (confidence interval=0.95)
N=10, K=5	0.403e-4	0.509e-4
K=10	0.236e-4	0.191e-4
K=15	0.831e-5	0.987e-5
K=20	0.347e-5	0.361e-5
K=25	0.227e-5	0.183e-5
N=11, K=5	0.977e-4	0.104e-3
K=10	0.640e-4	0.565e-4
K=15	0.402e-4	0.312e-4
K=20	0.167e-4	0.215e-4
K=25	0.874e-5	0.926e-5

Table 1 Comparison between analytical and simulation results - without FEC

In the simulation approach therefore, we vary K ranging from 5 to 25 cell spaces. All the simulation results were obtained using 95% confidence intervals. The only purpose of these simulations was to perform a check on the analytic values obtained. Since the loss rates are already low, a difference between the two results within the same order of magnitude was deemed to be acceptable. From Table 1 it can be seen that the results

obtained using the two approaches are within an order of magnitude, thereby verifying the analytic approach.

Observations on the utility of FEC

The following observations can be made from the above figures:

1. Even without using FEC (Fig 6.10), it is possible to achieve a very low HP loss rate (of the order of 10^{-12}) using a reasonable buffer sizing of $K=120$ (for $N=10$). The consequences of using FEC are

a) If the aim is to multiplex as many video channels as possible, for a given output link capacity, then using FEC has a disadvantage. For our particular kind of error coding scheme the maximum number of channels that could be multiplexed reduces from $N=13$ to $N=11$.

b) Taking into account the objective of maximizing N it can be observed that the gain obtained in HP loss rate by using FEC reduces substantially for increasing values of N (for $N=10$ and $K=100$, it is only an order of magnitude improvement - Fig 6.14). Hence, the FEC is limited in its utility where it is most desired.

c) There is a significant increase in the LP loss rate because of adding FEC; though in most cases it is still possible to achieve the specified LP loss rate of $<10^{-6}$.

From the above observations using FEC does not seem attractive for our problem. The cost involved is too high for only marginal improvements.

2. Using the FEC does however have the following advantages

a) For a given N and a specified loss rate, there is a substantial reduction in the buffer size required, by using the FEC. This therefore leads to a decrease in the delay involved, and holds promise for delay-sensitive video services. Consider for example $N=10$ and a HP loss rate of 10^{-12} . Without FEC (Fig 6.10), a buffer size of 120 is required whereas K reduces to 75 (Fig 6.13) using Option 2)— a reduction by almost 37 %.

b) For lower values of N there is a substantial reduction in the cell loss probability, for a specified buffer size. This is particularly useful for services where a low loss rate is essential, a requirement that may be otherwise met only by employing a very large buffer. From Fig 6.14 or Fig 6.15 it can be seen that using FEC has provided us with significant improvement in the loss rate. Consider for example, $N=9$ and a buffer size of $K=70$. There is an improvement of almost 2 orders of magnitude using Option 1) and 6 orders of magnitude using Option 2).

3. The improvement in HP loss rate increases as the buffer size increases.

4. For a given buffer size, the improvement in HP loss rate is less substantial for larger values of N , and for $N=11$ there is virtually negligible improvement using Option 2) and a definite deterioration in the loss probability using Option 1). Therefore, the coding scheme sets a bound on the number of sources that could be optimally multiplexed for a given channel capacity. Using Option 1) the optimal value is $N=10$ for $C=150$ Mbps.

5. For a specific value of N and K , the improvement in HP loss rate provided by Reed-Solomon codes is substantially superior to the one obtained using Parity cells. For example, consider $N=10$ and $K=120$. Using Option 1) the improvement in loss rate is only one order of magnitude (Fig 6.14) whereas with RS code it is almost 5.5 orders of magnitude (Fig 6.15). The reason for this is that the Reed-Solomon codes used in our case can correct up to two lost HP cells in a column of the encoding matrix, assuming erasure correction, whereas the Parity code can correct only a single HP cell loss.

Even though without any FEC it is possible to multiplex a maximum of $N=13$ video sources, the buffer requirements to achieve the required QOS of HP loss rate $<10^{-12}$, is very large (Fig 6.10) particularly for $N=11, 12$ and 13 . We therefore next investigate whether including a priority handling scheme reduces the buffer requirements for the same HP loss rate.

6.5 Evaluation of the priority scheme

HP cell loss rate

So far in the performance analysis we have neglected the priority handling scheme. Because of the nature of the push-out scheme which we implement to handle priorities, we can easily calculate the effect of the priority handling scheme on both HP and LP cell loss rates.

In the push-out scheme a HP cell will be dropped because of buffer overflow, if and only if there are no LP cells already present in the buffer, i.e., all the cells in the buffer are only HP cells. Hence the improvement in HP cell loss rate provided by the push-out scheme can be calculated by considering the HP traffic alone. Since the distribution of the HP cells generated by each video source in a MUX cycle is known, the performance was evaluated by adopting exactly the same approach as in the previous section excepting that all the multiplexed video sources are assumed to be generating only HP cells.

We first consider the effect of the push-out scheme on the HP cell loss rate *with* FEC included. Fig 6.17 and Fig 6.18 indicate the HP loss rate after decoding using push-out for Option 1) and Option 2) respectively. Comparing these with the case when only FEC was used and no priority scheme (Fig 6.12 and Fig 6.13), the improvement in HP loss rate is obtained (Fig 6.19 and Fig 6.20).

Figure 6.17 HP loss rate after FEC decoding, using push-out scheme (Option 1)

Figure 6.18 HP loss rate after FEC decoding, using push-out scheme (Option 2)

Figure 6.19 HP loss rate improvement using push-out scheme (Option 1)

Figure 6.20 HP loss rate improvement using push-out scheme (Option 2)

Both the figures (Fig 6.19 and Fig 6.20) indicate a considerable improvement in the HP loss rate, even for small buffer sizes, thereby rendering HP cells virtually error free.

However as discussed in the previous section, since FEC has several disadvantages in this study, the effect of the pushout scheme on the HP loss rate *without* any FEC. is cosidered next.

The HP loss rate without FEC overhead but using push-out is shown in Fig 6.21. Comparing this with the HP loss rate using neither FEC nor push-out (Fig 6.10) gives us the improvement in the HP loss rate, employing only the priority scheme (Fig 6.22).

The improvement in HP loss rate in this case is not as substantial as the one employing both FEC and push-out (Fig 6.19 and Fig 6.20). However, it offers the main advantage of avoiding the complexity associated with the implementation of the FEC scheme.

Figure 6.21 HP loss rate without FEC but using push-out scheme

Figure 6.22 HP loss rate improvement using only push-out scheme

LP cell loss rate

Since the push-out scheme provides an improvement in the HP cell loss rate at the expense of dropping LP cells, we next determine the effect of the push-out scheme on the LP loss rate. The LP loss rate without push-out but with accompanying FEC overhead was calculated in the previous section (Fig 6.11).

Let

$H1(L1)$ = HP(LP) cell loss probability with FEC but without push-out

$H2(L2)$ = HP(LP) cell loss probability with FEC and push-out

In both cases this is the loss rate before HP error correction.

There is a fixed ratio of 3:1 between the HP and LP cells. Therefore for every x cells transmitted from the source

Mean no. of HP cells = $(3/4) x$

Mean no. of LP cells = $(1/4) x$

Since there is 1 overhead (FEC) cell for every 6 HP video cells

Mean no. of HP cells including overhead = $(7/8) x$

Therefore the following relation holds for average losses in every x cells transmitted by the source.

Mean no. of HP cells lost, using FEC but no pushout = $(7/8) x H1$

Mean no. of LP cells lost, using FEC but no pushout = $(1/4) x L1$

Also,

Mean no. of HP cells lost, using FEC and pushout = $(7/8) x H2$

Mean no. of LP cells lost, using FEC and pushout = $(1/4) x L2$

The reduction in the number of HP cells dropped using the push-out scheme, must equal the increase in the number of LP cells dropped. Therefore,

$$\frac{7}{8} \times H1 - \frac{7}{8} \times H2 = \frac{1}{4} \times L2 - \frac{1}{4} \times L1$$

Since $H2 \ll H1$ and $H1 = L1$

$$L2 = 4.5L1$$

Hence, the increase in the LP cell loss rate due to the push-out scheme is less than half an order of magnitude. Similarly, the deterioration in the LP loss rate obtained by considering the push-out scheme but without any FEC was arrived at using the exactly the same approach as the above and is:

$$L2 = 4L1$$

Observations on the utility of the push-out priority scheme

From the above results we see that the use of push-out priority without FEC achieves a major improvement in performance without reducing the channel carrying capacity. However for the same HP loss rate it requires additional buffer space, as compared to using both push-out and FEC, and consequently larger delay.

We next consider briefly the aspect of transfer delay.

Transfer delay

One performance parameter of the ATM switching technique is the transfer delay of cells switched through the ATM network. This delay is defined as the time a cell spends in the network from its arrival to its departure (end to end transfer delay).

The transfer delay of cells running through an ATM network consists of a constant component called transmission delay, and a variable component comprising of queuing delay and the delay at the decoder due to reconstruction of missing cells (if any). The transmission delay is the same for every cell of a virtual connection and is only a few μs .

Delay due to transmission is therefore negligible and only the variable component is considered. In the following we determine the worst case delay for $N=10$ and a HP/LP loss probability of 10^{-11} . To achieve the above loss probability a buffer size of 100 cell spaces is required (from Fig 6.12), using Option 1). For this buffer size,

$$\text{Queuing delay (worst case)} = 100 \times 2.83 \mu s = 0.283 \text{ ms}$$

$$\begin{aligned} \text{Decoder delay} &= \frac{(N+1)(M+2)}{\rho} \times (\text{Cell period}) \\ &= \frac{9 \times 56}{0.7418} \times 2.83 \mu s = 1.92 \text{ ms} \end{aligned}$$

$$\begin{aligned} \text{Total delay} &= \text{Queueing delay} + \text{Decoder delay} \\ &= 0.283 \text{ ms} + 1.92 \text{ ms} = 2.203 \text{ ms} \end{aligned}$$

Chapter 7

Conclusions

Video services promise to become a major traffic component on future B-ISDN networks. The installation of optical fibers down to the subscriber local loop will lead to a proliferation of video services, particularly for home entertainment purposes. Using ATM technology to distribute these services involves the risk of cell losses due to buffer overflow, as well as random bit errors and cell delay. In this thesis we have analyzed the first two ATM impairments and suggested ways of tackling them.

The basic conclusions of the research work and the results of this thesis are summarized as follows:

- For fiber based transmission systems bit errors are generally low ($< 10^{-9}$) and can be very easily detected by including a 10 bit CRC field as part of the ATM cell payload rendering cell loss due to random bit errors negligible.
- Cell losses due to buffer overflow are a critical impairment of ATM networks. Also, when the buffer gets full, cell losses tend to occur in groups rather than randomly. Therefore, a consecutive cell loss compensation scheme to protect HP cells was used to overcome this problem. LP cell losses are detected by a simple SN check. The cell loss scheme employed three different options — 1) Using Single Parity Check codes 2) Using Reed-Solomon codes and 3) A combination of both.
- Interleaving of cells from different multiplexed sources prior to admitting them into the multiplexer buffer has the distinct advantage of removing correlation of losses for a single source. This is particularly necessary for video sources, which tend to be highly correlated. The interleaving technique mentioned in Chapter 6, can be applied to other kinds of source distributions as well, where correlation within a video source makes the analysis intractable.

- The analytic approach described in this thesis is an extremely simple and yet powerful tool for estimating cell loss rates, provided that the source distribution is known. This technique makes it possible to do analysis for very low loss rates as well.

- If from the analytic results it is observed that there exists a linear relationship between the cell loss rate and the buffer size, then the cell loss rate with a larger buffer size can be predicted from the simulation results in which small buffer sizes were assumed.

- If the objective is to multiplex as many video channels as possible into a given output communication link, then using FEC is not the solution and can prove to be disadvantageous for the following reasons

- a) The maximum number of video channels that can be multiplexed decreases because of the added overhead.

- b) The gain obtained in terms of improved HP loss rate reduces for increasing values of N .

- c) There is a significant increase in the LP loss rate.

For our particular source distribution, fortunately, the specified loss rates can be satisfied even without FEC, by employing a reasonably sized buffer for up to $N=10$. For higher values of N (up to $N=13$) to achieve the QOS with a reasonable buffer size, requires employing a priority scheme.

- Using the FEC does however provide a significant improvement in the HP cell loss rate for a specified buffer size or a substantial reduction in the buffer size for a specified loss rate. Therefore FEC is useful where loss rate reduction or delay reduction is a paramount consideration and the number of multiplexed sources is a less important criterion.

- FEC also provides an effective means of reducing the HP loss rate to negligible levels, where this is of overriding importance.

- The improvement in loss performance obtained by using Reed-Solomon codes is

far superior to that obtained by employing Parity Check codes. However, the hardware and computational complexity involved with using RS codes is considerably more as compared to Parity codes.

- The priority handling scheme used (the push-out scheme), has been shown in other applications to be comparable in performance to the PBS scheme, which is easier to implement. In our case the former technique provides significant improvement in the HP cell loss rate for both types of HP encoding schemes. Once again the RS code gives comparatively superior performance. The other attractive feature of this scheme is that the deterioration in the LP cell loss rate is negligible for the particular mix of traffic considered here.

- The improvement in the HP loss rate using only the priority scheme and no FEC is also significant though not as substantial as in the previous case. The attractive feature however is that the disadvantages of using FEC are not involved. Once again, employing the priority scheme has a very marginal effect on the LP loss rate.

Therefore, for the problem stated in this thesis, the best option is to use only the priority scheme and no FEC. The buffer requirements for various HP and LP loss rates and different values of N can be obtained using Fig 6.21 and Fig 6.11 respectively.

Appendix A Source distribution for a single video source

In this Appendix, an expression for the video source distribution is derived.

A geometric distribution has the following form [52]

$$P(X = k) = p (1 - p)^{k-1} \quad k = 1, 2, \dots \infty \quad \dots A1$$

In our case the random variable X can take on a maximum value of N , where $N=16$. We denote this by another random variable Y . Therefore,

$$Y = X \mid (X \leq N) \quad \text{where } N = 16$$

Therefore,

$$P(Y = k) = P(X = k \mid X \leq N)$$

Using Bayes result of probability

$$\begin{aligned} P(Y = k) &= \frac{P(X = k)}{P(X \leq N)} = \frac{p (1 - p)^{k-1}}{\sum_{k=1}^N p (1 - p)^{k-1}} \\ &= \frac{p(1 - p)^{k-1}}{[1 - (1 - p)^N]} \quad \dots A2 \end{aligned}$$

Since, the mean of the above truncated distribution is 5, the exact distribution is derived as follows.

$$\text{Mean} = S = \sum_{k=1}^N k p_k = \sum_{k=1}^N k \frac{p (1 - p)^{k-1}}{[1 - (1 - p)^N]}$$

Algebraic manipulation finally simplifies the above relation to

$$S = \frac{1}{(1-q)} - \frac{Nq^N}{(1-q^N)} \quad \text{where } p + q = 1$$

Equating the above mean to 5,

$$11q^{17} - 12q^{16} + 5q - 4 = 0 \quad \dots A9$$

Solving the above equation, the values obtained are

$$q = 0.8284$$

$$p = 0.1716$$

Substituting the value of p in equation A gives

$$\begin{aligned} P(Y = k) &= \frac{p(1-p)^{k-1}}{[1 - (1 - 0.1716)^{16}]} \\ &= (1.051) p(1-p)^{k-1} \quad k = 1, 2, \dots, 16 \quad \dots A4 \\ &\quad \text{and } p = 0.1716 \end{aligned}$$

Appendix B Generation of geometrically (truncated) distributed random numbers

The truncated geometric distribution which describes the source characteristics, is not a built in random generating function provided by the simulation package. Since the exact source distribution is known (Appendix A), the geometrically (truncated) distributed random numbers can be generated using the inverse transform technique [53].

The source distribution is given by equation A4, which is again shown here, for convenience

$$p(x) = (1.051) p (1 - p)^{x-1} \quad x = 1, 2, \dots, 16$$

$$\text{and } p = 0.1716$$

The cdf (cumulative distribution function) is

$$F(x) = \sum_{j=1}^x (1.051) p (1 - p)^{j-1}$$

$$= (1.051) [1 - (1 - p)^x] \quad \text{for } x = 1, 2, \dots, 16 \quad \dots B1$$

Using the inverse transform technique, a truncated geometric random variable (rv) X will assume the value x whenever

$$F(x - 1) < R \leq F(x)$$

$$\therefore (1.051) [1 - (1 - p)^{x-1}] < R \leq (1.051) [1 - (1 - p)^x]$$

$$\text{or} \quad (1 - p)^x \leq 1 - \frac{R}{(1.051)} < (1 - p)^{x-1}$$

$$x \ln(1 - p) \leq \ln\left(1 - \frac{R}{(1.051)}\right) < (x - 1) \ln(1 - p)$$

$$\text{Thus, } \frac{\ln\left(1 - \frac{R}{(1.051)}\right)}{\ln(1 - p)} \leq x < \frac{\ln\left(1 - \frac{R}{(1.051)}\right)}{\ln(1 - p)} + 1 \quad \dots B2$$

Thus, $X = x$ for that value of x satisfying inequality B2, or, in brief, using the round-up function

$$X = \left\lceil \frac{\ln\left(1 - \frac{R}{(1.051)}\right)}{\ln(1 - p)} \right\rceil \quad \dots B3$$

References

- [1] Special Issue on Packet Speech and Video, *IEEE J. Select. Areas Commun.*, vol. 7, June 1989.
- [2] Special Issue on Broadband Networks, *IEEE Network Mag.*, vol. 3, Jan. 1989.
- [3] S. Dixit and J.B. Nardone, "A variable bit rate layered DCT video coder for packet switched (ATM) networks," *Proc. ICASSP'90*, vol. 37, p. M8.9, 1990.
- [4] R. Ballart and Y. Ching, "SONET: Now its the standard optical network," *IEEE Commun. Mag.*, vol. 29, Mar. 1989.
- [5] B. Maglaris, et al., "Performance models of statistical multiplexing in packet video communications," *IEEE Trans. Commun.*, vol. 36, pp. 834-844, July 1988.
- [6] N. Ohta, M. Nomura, and T. Fujii, "Variable rate video coding using motion-compensated DCT for asynchronous transfer mode networks," *Proc. IEEE ICC'88*, June 1988.
- [7] F. Kishino et al., "Variable bit-rate coding of video signals for ATM networks," *IEEE J. Select. Areas Commun.*, vol. 7, pp. 752-760, June 1989.
- [8] W. Verbiest, L. Pinnoo, and B. Voeten, "The impact of the ATM concept on video coding," *IEEE J. Select. Areas Commun.*, vol. 6, pp. 1623-1632, Dec. 1988.
- [9] D.G. Morrison, "Variable bit rate video coding for ATM networks," *British Telecom Tech. J.*, pp. 70-80, July 1980.
- [10] B. Hagglund, *Private communication*, July 1991.
- [11] Y. Zhang et al., "Variable bit-rate video transmission in the Broadband ISDN environment," *Proc. IEEE*, vol. 79, pp. 214-222, Feb. 1991.
- [12] R. Clarke, *Transform Coding of Images*. New York: Academic, 1985.
- [13] Y. Yasuda, H. Yasuda, N. Ohta, and F. Kishino, "Packet video transmission through ATM network," *Proc. GLOBECOM'89*, pp. 25.1.1-25.1.5, 1989.
- [14] H.C. Reeve and J.S. Lim, "Reductions of blocking effects in image coding," *Opt. Eng.*, vol. 23, pp. 34-37, Jan./Feb. 1985.
- [15] A. Netravali and J. Stuller, "Motion-compensated transform coding," *Bell Syst. Tech. J.*, vol. 58, pp. 1703-1718, Sept. 1979.

- [16] M. Vetterli, "Multi-dimensional subband coding: Some theory and algorithms," *Signal Processing*, vol. 6, Feb. 1984.
- [17] J. Woods and S. O'Neil, "Subband coding of images," *IEEE Trans. Acoust., Speech, Signal Processing*, vol. 34, pp. 1278–1288, Oct. 1986.
- [18] P. Westerink, "Subband coding of images using vector quantization," *IEEE Trans. Commun.*, vol. 36, pp. 713–719, June 1988.
- [19] G. Karlsson and M. Vetterli, "Packet video and its integration into the network architecture," *IEEE J. Select. Areas Commun.*, vol. 7, June 1989.
- [20] Y. Linde, A. Buzo, and R. Gray, "An algorithm for vector quantizer design," *IEEE Trans. Commun.*, vol. 28, Jan. 1980.
- [21] CCITT Study Group XVIII, "ATM Adaptation Layer specifications — I.363," Geneva, Jan. 1990.
- [22] S.E. Minzer, "Broadband ISDN and Asynchronous Transfer Mode (ATM)," *IEEE Commun. Mag.*, pp. 17–24, Sept. 1989.
- [23] CCIR Study Groups Documents: Doc. IWP CMTT/3–93, "ATM Adaptation Layer functions for video and audio signal transmission with constant bit rates in B-ISDN," Feb. 1990.
- [24] S. Dravida and V.R. Saksena, "Error performance of IEEE 802.6 Metropolitan Area Networks," *Proc. INFOCOM '90*, pp. 105–115, 1990.
- [25] K.A. Tse et al., "DS-1 facility performance —Optical fiber as a long haul technology," *Proc. ICC '87*, pp. 646–649, 1987.
- [26] R. Damodaran and S. Dravida, "Error detection and correction options for data services in B-ISDN," *Proc. GLOBECOM '90*, pp. 900.5.1–900.5.7, 1990.
- [27] Y. Sato and K. Sato, "Virtual path and link capacity design for ATM networks," *IEEE J. Select. Areas Commun.* vol. 9, 1991.
- [28] Y. Sato and K. Sato, "Statistical multiplexing characteristics of multimedia traffic in high speed burst transport system," *Trans. IEICE*, vol. J72–B-1, pp. 1112–1119, 1989.
- [29] R. Kishimoto, Y. Okumura and K. Irie, "HDTV transmission system design in a SDH based STM network," *Proc. ICC '91*, pp. 491–495, 1991.
- [30] N. Shacham, "Packet recovery and error correction in high-speed wide-area networks," *Proc. MILCOM'89*, pp. 29.5.1–29.5.5, 1989.

- [31] N. Shacham and P. McKinney, "Packet recovery in high-speed networks using coding and buffer management," *Proc. INFOCOM '90*, pp. 124–131, 1990.
- [32] T. Kitami and I. Tokizawa, "Cell loss compensation schemes in an asynchronous broadband ISDN," *Proc. INFOCOM '90*, pp. 116–123, 1990.
- [33] H. Ohta and T. Kitami, "A technique to detect and compensate consecutive cell loss in ATM networks," *Proc. INFOCOM '91*, pp. 781–790, 1991.
- [34] H. Ohta and T. Kitami, "Simulation study of the cell discard process and the effect of cell loss compensation in ATM networks," *Trans. IEICE*, vol. E73, pp. 1704–1711, 1990.
- [35] CCIR Study Groups Documents: Doc. IWP CMTT/3–23, "Sequence Number structure and functions for TV transmission in B-ISDN," April 1991.
- [36] D. Lee and K. Tzou, "Hierarchical DCT coding of HDTV for ATM networks," *Proc. ICASSP'90*, vol. 37, p. M8.8, 1990.
- [37] K. Irie and R. Kishimoto, "Adaptive sub-band coder design for HDTV signal transmission," *Proc. ICASSP'90*, pp. 2117–2120, 1990.
- [38] J. Suzuki and N. Ohta, "Variable rate coding scheme for audio signals with subband and embedded coding techniques," *Proc. ICASSP'89*, vol. 14, p. S4.18, 1989.
- [39] G. Clark and J.B. Cain, *Error-Correcting Coding for Digital Communications*. Plenum Press, New York, 1981.
- [40] S. Lin and D. Costello, *Error Correcting Coding — Fundamental and Applications*. New Jersey: Prentice-Hall, 1983.
- [41] B. Maglaris et al., "Performance analysis of statistical multiplexing for packet video sources," *Proc. GLOBECOM'87*, pp. 1890–1899, 1987.
- [42] L. Dittmann and S.B. Jacobsen, "Statistical multiplexing of identical bursty sources in an ATM network," *Proc. GLOBECOM'88*, pp. 39.6.1–39.6.5, 1988.
- [43] M. Hirano and N. Watanabe, "Characteristics of a cell multiplexer for bursty ATM traffic," *Proc. ICC'89*, pp. 13.2.1–13.2.5, 1989.
- [44] G. Hebuterne and A. Gravey, "A space priority queueing mechanism for multiplexing ATM channels," *Comp. Networks and ISDN Systems*, vol. 20, pp. 37–43, 1990.
- [45] K. Rothermel, "Priority mechanism in ATM networks," *Proc. GLOBECOM'90*, pp. 505.1.1–505.1.5, 1990.

- [46] A. Gravey, P. Boyer and G. Hebuterne, "Tagging versus strict rate enforcement in ATM networks," *Proc. GLOBECOM'91*, pp. 271-275, 1991.
- [47] B. Jabbari and F. Yegenoglu , "An upper bound for cell loss probability of bursty sources in broadband packet networks," *Proc. ICC '91*, vol. 2, pp. 699-702, 1991.
- [48] A. Chandan, "Performance evaluation of an ATM multiplexer," *Master's Thesis*, Dept. of Electrical Engg., University of Ottawa, 1990
- [49] A. Papoulis, *Probability, Random Variables and Stochastic Processes*. New York: McGraw-Hill, 1984.
- [50] A.R. Kaye, *Private communication*, March 1992.
- [51] G.Woodruff et al., "Control of ATM statistical multiplexing performance," *Comp. Networks and ISDN Systems*, vol. 20, pp. 351-360, Dec. 1990.
- [52] A. Leon-Garcia, *Probability and Random Processes for Electrical Engineering*. New York: Addison-Wesley, 1989.
- [53] J. Banks, *Discrete-event system simulation*. New Jersey: Prentice-Hall, 1984.

END

15-1 2-93

FIN