ATM OVER AN ASYMMETRIC DIGITAL SUBSCRIBER LINE

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Declaration

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Abstract

This thesis investigates Asymmetric Digital Subscriber Line (ADSL) access networks and considers Asynchronous Transfer Mode (ATM) transport in the ADSL network. To analyse the ADSL access network, a simulation was developed using C++ and MATLAB. The thesis outlines the background and implementation of ADSL in the opening sections and then moves on to details about developing a simulation of the network.

The development of the simulation model outlining the development of noise sources in the network is described and modules for the network implementation are discussed. The local loop and the effects of different impairments such as bridge taps, line length, and wire gauge are also considered. The impairments in the copper loop are investigated to highlight the effect they have on the frequency response of the line. The line length and gauge increase the attenuation while bride taps introduce a non-linear phase response. The latter non-linear phase shift is very hard to compensate for. The loop characteristics are incorporated into the simulation, so as to represent different loop structures which exist in the network The overall affect of the noise sources and loop characteristics, can then be added to the simulation.

In simulating the network, specific focus is placed on discovering the possible rates at which ADSL could operate in the network. Consideration is given to both the line length and Far-End Cross Talk (FEXT). FEXT is signal noise introduced into the transmission line at the remote end by other DSL sources in the same wire bundle and is a function of the transmitter power. The simulation is then extended to include the effect of impulse noise on the ADSL physical bit error rate (BER). Impulse noise has a very high intensity and short duration. These properties of the impulse noise give rise to high BER. To offset high BER, both a coding gain from Forward Error Correction (FEC) and interleaving to distribute the errors are used. Using these methods in the simulation, improvements in network configurations are found which allow for successful ATM transport.

Results are presented for the achievable data rates, BER and the distribution of the bit errors, both under the influence of impulse noise and with no impulse noise present. The convolutional interleaver is very successful at mitigating the effect of impulse noise. The latency introduced in the interleaved stream, however, introduces problems for ABR ATM transport. The rate control mechanism for ABR tends to oscillate in high latency networks. The transportation of ATM over the ADSL network is considered. The results from the simulation at the physical level are used to interpret the effects that the ADSL link has on cell transport.

Network management is considered with ATM as the transport protocol that unifies the access network. Methods for implementing the network and end to end signalling are presented. Signalling methods for both management and call setup are presented. Consideration is also given to traffic types offered by ATM and the problems presented for them by ADSL. The specific focus here is on Available Bit Rate (ABR) and Unspecified Bit Rate (UBR) traffic types.

Early frame discard, cell discard and an AAL5 frame size limit are considered as a method for improving the throughput in the network.

From the simulations the conclusion was made that ATM offers a functional solution to the integration of the ADSL network. With the correct implementation of FEC and interleaving, the required quality of service (QoS) can be achieved. It is clearly shown that it is possible to achieve a low enough BER for ATM transport. The major limitations in the access network come from FEXT and Impulse noise. The FEXT does not pose a specific problem for ATM transport but does set the upper end data rates. Impulse noise, on the other hand, is the largest problem, as an interleaver is needed to overcome its effects. This introduces a large latency and can be problematic for specific services and flow control in ABR.

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Glossary

Abbreviations, acronyms and symbols

AAL	ATM Adaptation layer
AAL 1	The constant bit rate, synchronous ATM Adaptation Layer
AAL 5	The Most Commonly used adaptation layer can be used for constant or variable bit rate services.
ADC	Analogue to Digital Converter
ADSL	Asymmetric Digital Subscriber Line
AEX	A(S) Extension Byte
AOC	ADSL overhead control channel
ARQ	Automatic Repeat Request
AS(x)	Any one of the simplex bearer channels downstream AS1-AS3
ATM	Asynchronous Transfer Mode
ATU-C	ADSL Transceiver Unit Central Office
ATU-R	ADSL Transceiver Unit Remote End
AWGN	Additive White Gaussian Noise (Back Ground Noise)
BER	Bit Error Rate
B-ISDN	Broadband Integrated Services Digital Network
Bridge Taps	Unused twisted pairs, attached to the subscriber loop in parallel
CAP	Carrierless Amplitude Phase Modulation
CBR	Constant Bit Rate
CO	Central Office, contains line terminating equipment
CPCS	Common Part Convergence Sublayer
CPI	Common Part Indicator
CRC	Cyclic Redundancy Check CRC-8 is used
CS	Convergence Sublayer
CSA	Carrier Service Area
DAC	Digital to analogue converter
DFT	Discrete Fourier Transform
DMT	Discrete Multitone

DNS	Domain Name Server or Service
Down Stream Data	From the ATU-C to the ATU-R
DSL	Digital Subscriber Line
DSLAM	Digital Subscriber Line Access Multiplexer
DSP	Digital Signal Processing
DWMT	Discrete Wavelet Multitone
EC	Echo Cancellation
EOC	Embedded Operations Channel
FDM	Frequency Division Multiplexing
FEC	Forward Error Correction
FEXT	Far-end Cross Talk
FFT	Fast Fourier Transform
FTTC	Fibre to the Curb
FTTH	Fibre to the Home
FTTN	Fibre to the Node
HDSL	High Speed Digital Subscriber Line (Can be used as a replacement for T1 and E1 Links)
HEC	Header error control
HFC	Hybrid Fibre Coax
Ib	Indicator Bits
IDFT	Inverse Discrete Fourier Transform
IFFT	Inverse Fast Fourier Transform
ILMI	Integrated Local Management Interface
IP	Internet Protocol
ISI	Inter-Symbol Interference
Kbps	Kilobits per Second
LANE	LAN Emulation
Load Coil	Copper Coils Inserted in Series in the Loop to improve the Frequency Response
Local Loop	The Final Section of Twisted Copper to the Residential Telephone
LS(x)	Any one of the Duplex bearer channels LS0-LS2

LSA	Least Significant Bit
MAC	Medium Access Control
MPOA	Multi Protocol over ATM
MSB	Most Significant Bit
NEXT	Near-end Cross Talk
NIC	Network Interface Card
NNI	Network to Network Interface
NRT-VBR	Near Real Time Variable Bit Rate
PCR	Peak Cell Rate
PDU	Protocol Data Unit
PMD	Physical Medium dependent layer
POTS	Plane Old Telephone Service
PSD	Power Spectral Density
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
RM	Resource Management Cell for ABR flow Control
RMS.	Root mean square
RS	Reed Solomon
RT-VBR	Real Time Variable Bit Rate
SAR	Segmentation and Reassembly
SCX	Synchronisation bits (1-7)
SNMP	Simple Network Management Protocol
SNR	Signal to Noise Ratio
STM	Synchronous Transfer Mode
TC	Transmission Convergence Sublayer
TCP	Transmission Control Protocol
UNI	User Network Interface
Upstream	From the ATU-R to the ATU-C
UU	User to User
VBR	Variable Bit Rate
VC	Virtual Channel
VCI	Virtual Channel Identifier

VDSL	Very High Speed Digital Subscriber Line
VoD	Video on Demand
VP	Virtual Path
VPI	Virtual Path Identifier
WDM	Wavelength Division Multiplexing
XT	Cross Talk
XTALK	Cross Talk

Chapter 1 Introduction

1.1 Background

Asymmetric Digital Subscriber Lines (ADSL) are used to provide high-speed digital data access over existing phone lines without affecting the plain old telephone system (POTS) service. ADSL is a unique technology that not only allows the Telecommunications Companies the opportunity to sell value-added services, but also to extend the life of copper lines in the access loop. ADSL goes hand in hand with technologies such as Fibre to the Curb (FTTC) and Hybrid Fibre Coax (HFC). However ADSL is seen as a transitional technology, a forerunner to Fibre to the Home (FTTH).

ADSL offers Telecommunication Companies a good opportunity to remove the Internet traffic from the switched voice network. It is beneficial to remove the Internet traffic from the voice network because it has a completely different call holding pattern to voice calls. For example, the mean holding time for voice calls is 3 minutes and for Internet calls 16 minutes. [33]. The different call holding times, not only increase the number of blocked calls on networks, but result in problems designing and dimensioning the voice network. To offset the management problems of the voice network, the first and most likely source of traffic for the ADSL network is Internet traffic. With this move, comes the improvement in available bandwidth to the home. If a technology like ATM is used in conjunction with ADSL, it will be possible for Telecommunication companies (Telcos) to provide value added services such as Video on Demand (VoD) to residential users.

ADSL provides a method of offering broadband access to the home. It does this by using the extra analogue bandwidth available outside the POTS 4kHz, while still supporting POTS services. More specifically ADSL implements this by using the bandwidth available in the local loop from 25kHz to 1.104MHz.

This local loop bandwidth has 3 distinct channels: the first carries the old POTS services and the other two channels carry the downstream ADSL and the upstream ADSL traffic.

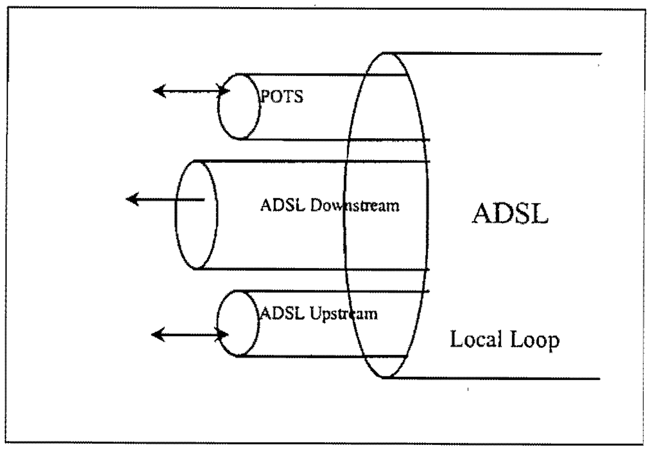

Figure 1-1 ADSL in the Local Loop

ADSL offers a basic rate of 1.536 Mbps in a simplex downstream channel and a 160 Kbps full duplex upstream channel, some control traffic flows back in the upstream bandwidth. The upstream channel carries all the traffic from the remote end to the core network, and also carries a bi-directional control channel. The downstream channel only carries the traffic to the remote end from the core network Figure 1-1. This basic rate service can be extended up to 8 Mbps downstream and 640 Kbps upstream. These data rates are, however, a function of the line length and the amount of noise present in the local loop [7].

There are two methods for implementing the up and downstream channel. The first and simplest method is to use frequency division multiplexing (FDM) where the up and down streams use different parts of the spectrum. The other option is to implement echo cancellation, which has about a 2db gain over the FDM solution. With echo cancellation the spectrum for the up and downstream overlap. Echo cancellation is implemented, because the receiver knows what signal is been transmitted at that instant and subtracts it from the received signal. The spectrum with echo cancellation is shown below in Figure 1-2.

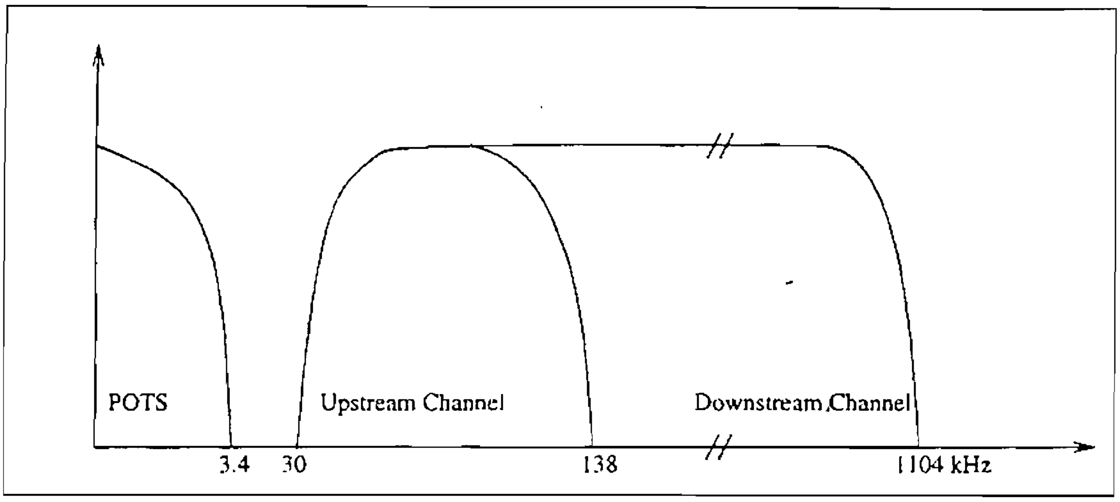

Figure 1-2 ADSL Frequency Spectrum.

Carrierless Amplitude Phase Modulation (CAP) is a variation of amplitude and phase modulation. With the onset of digital signal processing, multicarrier techniques are more feasible, such as Discrete Multitone Modulation (DMT). By using DMT instead of CAP, ADSL can achieve dynamic rate adjustment and can match the transmitted signal to the channel response. Both of these features allow ADSL full bandwidth flexibility with respect to both up and downstream data rates. The data rate may be adjusted freely and continuously. On start up or initialisation, the modem calculates the maximum possible data rate that is achievable on the specific phone line in use.

The limiting factors in the ADSL access network are the line characteristics and the noise sources present in this environment. The main noise sources are impulse noise, crosstalk, background white noise, and AM radio interference. DMT modulation offers a good solution to RF interference by eliminating that section of the spectrum from the usable bandwidth. Unlike other DSL solutions, whose data rates are limited by the near-end crosstalk (NEXT), ADSL data rates are limited by far-end crosstalk (FEXT) and impulse noise [7]. In noisy environments where the received signal is very weak, ADSL must implement a powerful Forward Error Correcting (FEC) coding scheme to achieve acceptable data rates with low Bit Error Rates (BER). In order to combat the impulse noise in the network, the modem implements a safety margin and a convolutional interleaver, which mitigates the effect of impulse noise. The convolutional interleaver is very good at eliminating errors due to impulse noise but introduces a large latency. The effect of constant noise sources such as background white noise and RF interference is offset in the bit-loading algorithm and data rate set to achieve the required BER [1].

Until now, most of the research work undertaken on DSL has focused on HDSL systems that implement CAP or DMT modulation. In past research [20] [32] [2] the focus has been on the limitations imposed by the local loop and NEXT. In ADSL, NEXT is not the major limiting factor [7]. The focus needs to be shifted to the simulation and modelling of the network using FEXT and Impulse Noise as the limiting noise sources. In the past, little or no consideration was given to impulse noise. When using ATM over an ADSL access network, impulse noise becomes a more significant problem [23]. For example, if only one ATM cell in an AAL5 frame is dropped or corrupted then the entire frame is discarded. Because of this, the level of protection offered to impulse noise and the length of the AAL5 frames must be considered.

The line characteristics of the local loop change from link to link depending on factors such as line length, changes in gauge, bridge taps, and line termination. In older loops there are also load coils, which are used to boost the frequency response of the longer loops. It is not possible to implement ADSL on these loops, as the load coils attenuate the higher frequency signal of ADSL and as a result they must be removed.

Applications require different network properties. To support the required classes of service, ADSL implements two physical streams within the physical layer of the modem. There is a Fast and Interleaved streams. The Fast stream only has FEC while the Interleaved stream has both RS-FEC and a convolutional interleaver. The ADSL link has two physical connections with different properties. The fast stream has a lower latency but is adversely affected by impulse noise. The interleave stream has a better resistance to impulse noise at the price of increased latency.

ADSL makes use of asymmetric bandwidth. ADSL is a good solution for transporting applications, which require a greater downstream speed than upstream speed. Possible applications include Internet Traffic, and video-based services, such as Video on Demand.

ATM offers the ability to fully utilise the bandwidth in the access network. The access network will require more functionality, for the newer applications that utilise greater bandwidth and guaranteed QoS. ADSL offers the first real solution to increasing the bandwidth to the home. By implementing two physical links with different properties, it can use these to offer different QoS on each channel. ATM uses these dual physical links independently, offering the QoS needed. ATM provides a uniform protocol for all services, scalability and network management all packaged together in one protocol.

The possible future applications for ADSL with their required bandwidths and protocol are listed in Table 1-1 [22] below. This clearly shows that ATM is the unifying protocol that can serve the wide range of applications that may be needed in the residential market. It is for this reason that ATM in an ADSL access network must be fully considered.

Service	Application	Downstream Bandwidth	Upstream Bandwidth	Protocol
Video	Broadcast TV	6-8 Mbps	64 Kbps	ATM (NRT-VBR)
	Video on Demand	1.5-3 Mbps	64 Kbps	ATM (NRT-VBR)
	Distance Learning	1.5-3 Mbps	64-384 Kbps	ATM (RT-VBR)
	Video Conferencing	640 Kbps	640 Kbps	ATM (RT-VBR)
Data	Internet	up to 1.5 Mbps	10% of Down	ATM (IP/MPOA)
	Remote LAN	up to 1.5 Mbps	10% of Down	ATM (MAC/LANE)

Table 1-1 ADSL Services and Protocols

1.2 Thesis Objectives

The aim of this thesis is to analyse ATM transport over ADSL, with respect to data rates, BER and required QoS. ATM is a protocol that provides the access network the required functionality with respect to network management and easy growth of the network. Provision must be made for both current legacy applications and future applications such as Video on Demand (VoD) which will require guaranteed quality of service. The primary focus is how ATM offers a solution to integrate the current Internet configuration with ISP and dialup lines into a new ADSL access network with little or no change to the ISP and the client. Point-to-Point Tunnelling Protocol (PPTP) over ATM over ADSL is a possible solution.

In order to understand ATM in the ADSL network, a simulation of the access network is developed. An analysis of the bit errors and properties of this network is made from the results obtained. The network is simulated under different network configurations of line length, amount of forward error correction (FEC) redundancy, and the varying interleaving depth.

In order to develop an accurate model for this access network it is important to model the physical characteristics of the media correctly. In doing so, accurate models of the noise sources in the network such as impulse noise, Far-End Crosstalk (FEXT) and Near-End Crosstalk (NEXT) are needed.

A method for representing the channel characteristics is also investigated, to represent lines with different structures, with regards to their length, gauge, termination, and the location of bridge taps within the local loop.

The simulation data will be used to analyse the affect the ADSL access network technology will have on the transport of ATM traffic to the home. The transportation of AAL PDU's over the access network is focused on. Different types of traffic are investigated, namely ABR, UBR, and NRT-VBR, to see how the access network affects the QoS needed for these traffic types.

Latency in the access network is considered as a result of ABR flow control. In UBR the main problem is flooding of the access point to the access network [14]. This is because the main point of congestion will be the DSLAM in the ADSL access network. UBR sources send bursts of data at the maximum line rate. A problem occurs when these sources are sending data at a rate, which is greater than the DSLAM uplink. Buffering problems result at the DSLAM and a large number of cells are dropped. It may be possible to provide a Peak Cell Rate (PCR) for UBR connections when used in an ADSL access network. This is possible with UNI 4.0 [44], At call setup, the UBR peak rate will be set by DSLAM to a rate it can accept for that users virtual channel.

Different possible network configuration with regards to cost and functionality will be analysed, focusing on the provision of ATM to the home.

1.3 Thesis Outline

The outline to the structure of the thesis is presented in this section. It follows the same progression as the research work performed. Chapter 2 outlines the modulation technique used in ADSL and the ADSL network structure. This section provides the details on the modulation scheme and the bit-loading algorithm. The ADSL transceiver units are also presented. The ADSL frame structure is introduced and its integration with the fast and interleaved streams is shown. The physical layer overhead is considered and an option for a reduced overhead mode is presented that relies on ATM cell transport.

Once the ADSL operation is understood, the characteristics of the local loop are investigated and presented in Chapter 3. This section includes the structure of the local loop and the effect these characteristics may have on the ADSL access network operation. The results for the frequency response of the loop are also presented in this section. These include the magnitude response, phase response and achievable data rates as function of loop length. The group delay is also presented for the channels and is an indication of the linearity in the phase response.

Chapter 4 outlines the noise sources present in the local loop, which may affect the ADSL modems operation. These include crosstalk, both far and near-end, additive white Gaussian noise, impulse noise, and radio interference. This section further elaborates on the characteristics of this noise and how these noise sources present in the network are modelled. A sample of the impulse noise and the power spectral density of crosstalk models used in the simulation is presented.

At this point there is a thorough presentation of all the work leading up to the development of the simulation of the ADSL access network. The simulation implementation is presented in Chapter 5. This section goes into specific details of both the operation and the actual implementation of each module used in the access network. The implementation of the Reed-Solomon FEC and interleaver and modulator are presented. This includes the algorithms used, a description of why two programming languages are used, and how they are integrated into a single functional simulation.

After defining the simulation structure and implementation, Chapter 6 presents results of the simulation undertaken. The data recorded is analysed with respect to the data rate and BER.

Chapter 7 moves towards the aspect of ATM over ADSL and is considered in conjunction with its use in an ADSL access network. This section considers the functionality needed in the ATU's and DSLAM for ATM transport. It considers the dynamic rate adjustment ADSL can implement on the link and considers this with ATM traffic. The section then presents the solution ATM offers, for end-to-end interoperability in the access network. This section ends by presenting the protocol stacks for ATM across the ADSL network.

Chapter 8 presents some background into ATM cell structure focusing on cell protection. The AAL layers are briefly introduced as aspects of them are used later in the discussion. The section then looks at cell losses in the ADSL network and presents a possibility of early frame and cell discard in an ADSL link.

Chapter 9 draws conclusions about the work undertaken, with respect to the attainable data rates, BER and the affect they have on ATM traffic.

Chapter 10 makes recommendations for future research. These include simulating Discrete Wavelet Multitone (DWMT) and to move towards simulating VDSL.

Finally, five appendices are presented to give the reader both background information and additional results. In Appendix A, a comparison is made between Carrierless Amplitude Phase (CAP) modulation and Discrete Multitone Modulation (DMT), to outline the modulation technologies that where available for ADSL. Appendix B contains extra illustrations of impulse noise from the impulse module. Appendix C goes into detail to describe and present theoretically the bit-loading algorithm used in this simulation. Appendix D presents details on the constellation diagrams and the generation of these constellations in the simulation. Appendix E presents the structure used to describe the local loop cable in the simulation. A description of the simulation variables is given and the structure of the loop is laid out, with an example.

Chapter 2 DMT Modulation and the ADSL Network

Chapter 2 describes the modulation scheme used in the ADSL access network simulation and the modem structure. Motivation on why Discrete Multitone (DMT) modulation is implemented as opposed to CAP is presented in Appendix A, the appendix outlines other options that are available, for implementing DSL. Here the implementation of DMT is presented as well as the bit-loading algorithm used in the network simulation. Following the presentation of DMT, the ADSL frame structure is presented. This also includes a discussion and description of the overhead built into the frame. The subsequent sections of the chapter detail the modem structure with specific focus on the forward error correction (FEC) and interleaving technique used.

These sections are presented in detail, as this becomes the basis of the simulation. Details on network structure, data framing and data paths are needed for the simulations. Energy levels and signal details are needed to model the physical signal correctly in the noisy channel. The implementation and structure presented here is based on the ANSI T1.413 specification [43].

2.1 DMT Modulation

DMT was first implemented by Peled and Ruiz of IBM and refined by Ruiz and Cioffi. [1]. ADSL modem implementation, predominantly utilise DMT modulation. The fundamental concept in DMT is to split the available frequency spectrum in the copper network, into sub-channels. Each of these channel sub-carriers can be individually modulated with data. DMT is optimal in that it optimises the SNR across the entire band and does not use an average SNR across the band to obtain the correct BER [1][2]. Complete channel optimisation is possible, as the frequency band is divided into sub-channels. Each sub-channel is optimised to maximise the effective data rate in individual channels. Sub-channel optimisation avoids the problem in other modulation schemes where an average channel response for the whole spectrum is used to set the data rate. Implementation of DMT modulation has only recently become viable with the onset of faster digital signal processors (DSP) and the use of Inverse Fast Fourier Transform (IFFT) to implement the modulation.

To achieve this high performance, the modulation in each sub-channel is adapted by changing the bits per symbol for each of the sub-channels. The bit rate is based on the individual channel response for each of the sub-channels. The channel response is initially found by transmitting a pseudo random sequence with an equal number of bits per sub-channel. Optimisation is based on the SNR for each of these sub-channels [7]. An illustration of the optimisation is presented in Figure 2-1 below. Here initially an equal number of bits per symbol are transmitted in all the sub-channels from the source across the channel. Based on the received signal it is possible to calculate the channel response and SNR across the band.

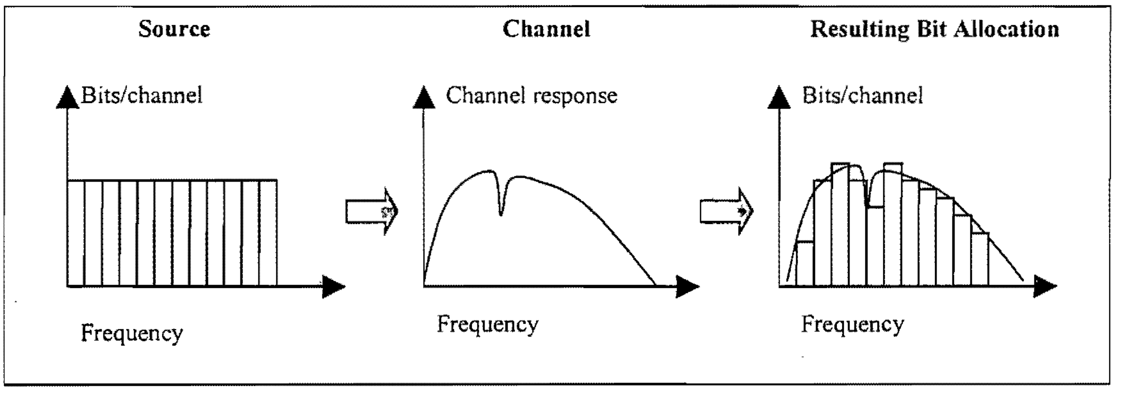

Figure 2-1 Example of DMT bit Allocation

For each of these sub-channels a Quadrature Amplitude Modulation scheme is used. A bit-loading algorithm is used to allocate the number of bits per channel based on the SNR in each of these channels. The bit loading algorithm is described in section 2.2 .

The incoming bit stream is divided into 256 parallel streams, each with the correct number of bits for the required sub-channel. These 256 streams are then mapped onto a complex number using a constellation diagram (Refer to Appendix C). In this implementation the average symbol energy is equal in all the sub-channels. The symbol energy is essential so that the transmitter energy is spread evenly across the spectrum.

The DMT modulation scheme used in ADSL divides the spectrum into 256 sub-channels spaced 4.3125 kHz apart. The ADSL spectrum ranges from 25.875 kHz to 1104 kHz. The lower 6 channels fall in the POTS and ISDN range and can't be used for ADSL the remaining 250 channels are used [43]. The same channel parameters are outlined in the ADSL Forum, ANSI and ITU specifications.

DMT modulation is implemented using a Fourier Transform. Equation 2.1.1 represents an Inverse Fast Fourier Transform (IFFT). The complex symbol X_i comes from the constellation mapping for the i^{th} sub-channel, which transforms the incoming bit sequence to a complex number [24]. The square root of N is used as a constant scalar to preserve the energy in the time domain and frequency domain. The analogue representation is shown graphically in **Figure 2-2** below [34].

$$x_n = \frac{1}{\sqrt{N}} \sum_{i=0}^{N-1} X_i e^{j2\pi in/N} \qquad 2.1.1$$

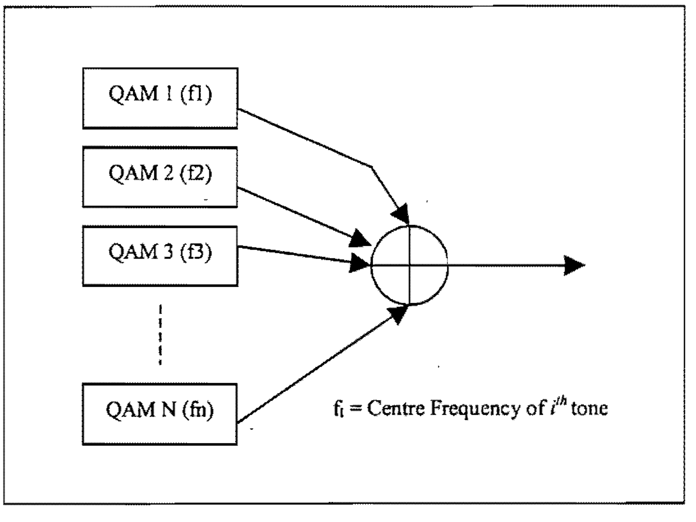

Figure 2-2 Multitone QAM System

The constellation used can have from 2 to15 bits per symbol. The Euclidean distance between the symbol points determines the resulting symbol energy. The calculation of the distance needed is dealt with in the next section 2.2 . Once the bit streams are encoded into the 256 complex points they are passed as a vector to the IFFT.

It is a requirement that for the output of the IFFT to have real values, the input vector needs to have Hermitian symmetry [43]. So to preserve the Hermitian symmetry property, the 256 point complex input vector is concatenated with vector resulting from the following equation 2.1.2:

$$\begin{gathered} Z_i = conj(Z_{512-i}) \\ for\, i = 257 to 511 \end{gathered} \qquad 2.1.2$$

The symmetrical vector is passed to the IFFT then the real data is converted to a serial data stream and passed through a DAC and Line filter. A Cyclic prefix is appended to each frame,[43] to create a guard band and to prevent inter symbol interference (ISI) in the channel between frames. For the cyclic prefix, the last 32 samples of the 512-sample DMT symbol is added to the DMT symbol before the DAC [43]. That is, the symbol length is now 544 samples long. A speed up in the DAC of 544/512 is needed to preserve the correct DMT symbol rate. The cyclic prefix is discarded in the receiver. The cyclic prefix helps to determine DMT symbol boundaries and reduce the ISI resulting from the channel.

To improve performance, Trellis coding [1][24][20] may be implemented. Additional coding is an optional extra, and is not required by the defining standards. The code outlined in the ADSL standard is Wei's 16-state 4 dimensional trellis code. In most applications Trellis coding can be expected to yield a 5dB coding gain [1].

2.2 Bit-Loading Algorithm

During the channel analysis phase, the bit-loading algorithm determines the maximum number of bits per symbol that can be carried on each of the 256 sub-channels. Once the SNR is found for all the sub-channels, the bit-loading algorithm can be used to calculate the attainable bits per symbol for each sub-channel. In the two-part process, the number of bits is firstly calculated and then the required constellation is generated for that number of bits per symbol.

The number of bits per symbol ***b*** is obtained from equation 2.2.1. The derivation of the bit-loading algorithm is outlined in Appendix C. This expression is used on each sub-channel to calculate the number of bits. Here τ is defined as the SNR gap, γ_m is the system margin and γ_c is the coding gain resulting from the FEC and Trellis coding. The SNR is calculated during channel analysis [1]. τ is the measure of the SNR in dB that is required to achieve a required BER. The system margin γ_m is the measure of the safety margin inserted for unforeseen errors in the system and is usually set to 6 dB.

$$b = \log_2\left(1+\frac{SNR}{\tau}\right) \qquad 2.2.1$$
$$\tau = 9.8+\gamma_m+\gamma_c$$

In the generation of the constellation mapping it is important to calculate the correct Euclidean distance between adjacent points. The distance ***d*** determines the average symbol energy and is governed by equation 2.2.2.

$$d^2 = \frac{6\varepsilon}{2^b-1} \qquad 2.2.2$$

The above calculation of d for a given average symbol energy applies well for even integers of b although in ADSL, odd values of b often arise. The above formula remains relatively accurate for odd values of b. For odd values of b, the Euclidean distance d is adjusted by a scaling factor in the simulation to ensure that all the sub-channels have equal average energy per symbol. The energy per symbol is illustrated in the Figure 2-3 for varying bits per symbol. The uncompensated curve is far off for odd values of b. However, after the applied correction the energy is constant for all the required values of b_i. Energy compensation is expanded on in section 5.5.2 where the simulation is outlined.

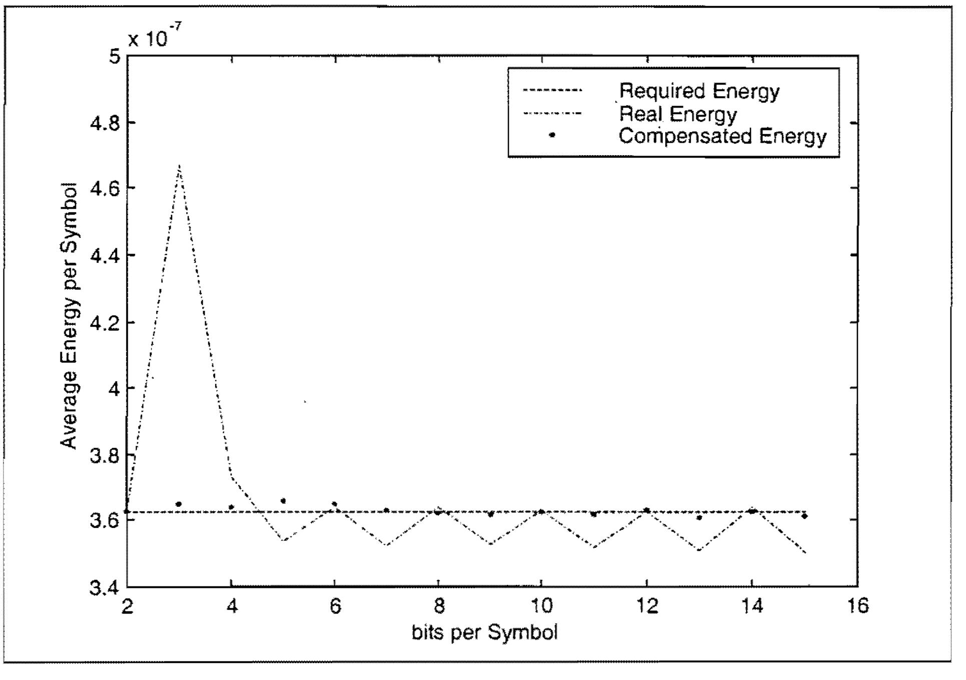

Figure 2-3 Average Energy per symbol with the TX Power Set to 100 mW

2.3 ADSL Network Structure and Reference Models

This section focuses on the network and the modem structure, both the central office and at the remote user end. Reference models for both the office and the remote end modems are now presented to aid the readers in understanding subsequent sections.

2.3.1 End to End Network Structure

There are three main components that make up the ADSL access network:

- The ADSL transceiver unit remote-end (ATU-R)
- ADSL transceiver unit central office-end (ATU-C)
- The Digital Subscriber Line Access Multiplexer (DSLAM).

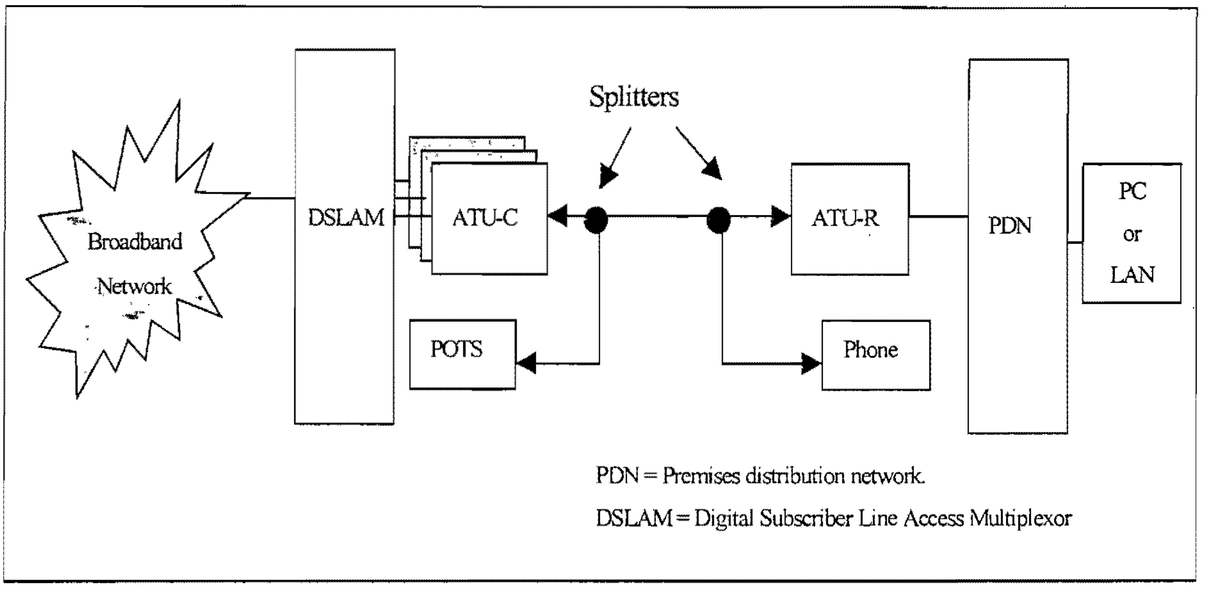

Figure 2-4 Network Structure of an, End to End ADSL Access Network

Both the DSLAM and the ATU-C are situated in the local exchange office. For each subscriber on the ADSL access network there has to be an ATU-C in the central office. The ATU-C is the modem that terminates the incoming subscriber twisted pair. Each of the ATU-Cs is connected to the DSLAM. The DSLAM serves as an access node concentrator/multiplexer for all the subscribers in the area. The DSLAM may take up different functional forms depending on the technology of the core network.

At the customer premises, the ATU-R connects the end user to the access network. The ATU-R may take on different forms depending on how the network is used. The ATU may also have other interfaces depending on the mode of operation. This may be for ATM transport or packet transport. The ATU-R may take on different forms from a plug in card in the customer PC to a standalone unit with an ATM25 or possible Ethernet interfaces.

Figure 2-4 above shows the major components and their relative positions in the access network. The ATU's form the interface to the local loop from both sides and the DSLAM forms the interface for all the subscribers to the broadband network.

2.3.2 ATU-R and ATU-C

As explained above, the ATU's are in essence the end ADSL modems one in the central office and the other at the customer premises. They are included together in this section as they are similar in functionality. This subsection outlines the structure of the ATU and its functionality as it forms the main component of the simulation presented later in Chapter 5. The structure of the ATU used complies with the ANSI [43] specification. The ETSI [51] standard, implements the same structure and interface. The only difference between the two standards is that ETSI recommends changing the PSD of the ADSL single. Allowing ISDN systems to operate on the same physical wire. Having both ADSL and ISDN sources on the same channel is not implemented in this simulation.

The main feature of the ATU, and hence an ADSL access network, is the two paths through the modem. Essentially, creating two different data paths with different properties over the same physical wire. These two paths are referred to as the Fast and Interleaved Data Streams. The ATU is shown in a more detailed functional diagram in **Figure 2-5**. The Fast Stream only has a Read Solomon Forward Error Correction (RS-FEC). The Interleaved stream has both RS-FEC and a Convolutional Interleaver. The interleaver introduces latency in the interleaved channel.

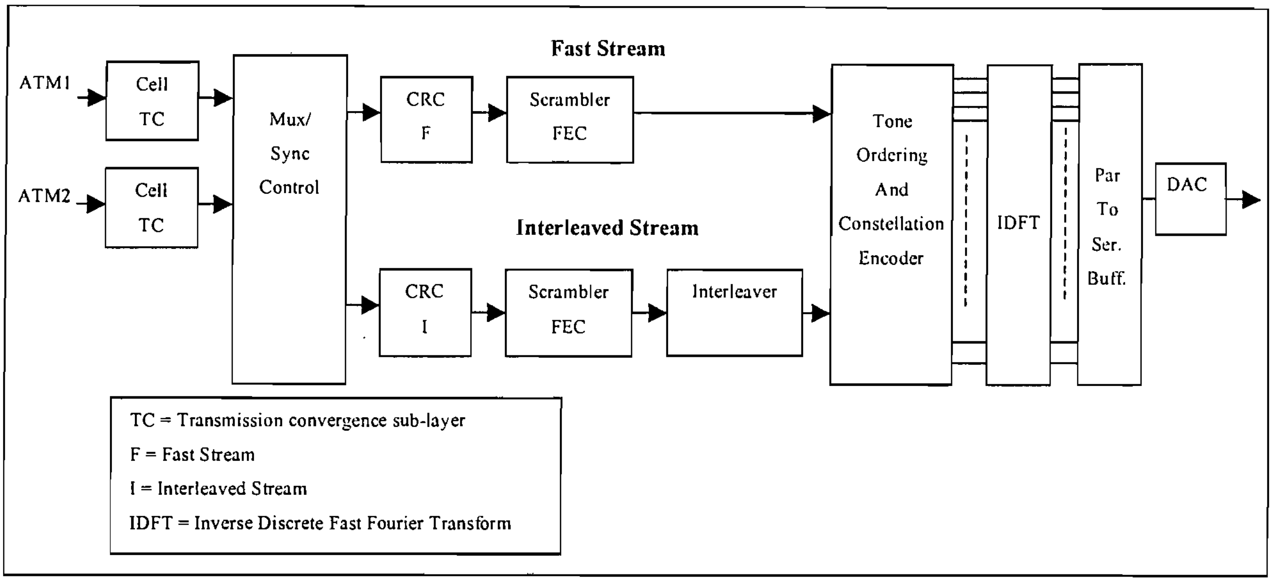

Figure 2-5 ADSL Transmitter Unit for ATM Transport

Both ANSI [43] and ESTI [51] define the same interface to the ATU C-R. At the input to the ATU-C there are 7 channels. Four downstream ASx channels AS0 - AS3 simplex channels (any one of the simplex bearer channels downstream) and three bi-directional LSx duplex channels (any one of the Duplex bearer channels LS0-LS2).

The transport protocol used in the simulation is ATM [40]. It allows for virtual channel multiplexing, this simplifies the implementation and only two channels are needed on the ATU. In the case presented here only two of the ASx channels in the downstream and two LSx channels upstream are considered. In both cases one is for the fast stream and the other for the interleaved stream.

The ATC-Receiver differs in one aspect from the ATU-Transmitter in that there are no ASx Channels as these are only used for downstream transmission. The duplex LSX channels are used. Figure 2-6 below shows all the channel interfaces to the ATU-C, whereas Figure 2-5 simplifies this interface for ATM transport.

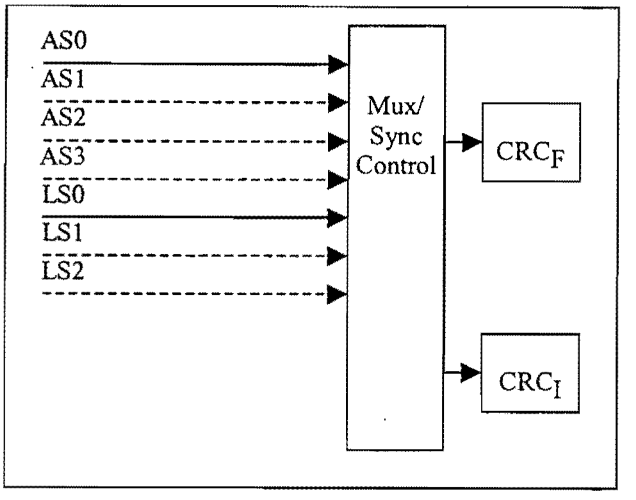

Figure 2-6 ATU-C showing all channel interfaces

Incoming data from higher layer protocols are passed to the Mux/Sync Control section. The main functions are the multiplexing of data downstream and the demultiplexing of data upstream for the different ASx and LSx channels. The synchronisation for the channel multiplexing is presented in the next section, which covers the frame structure.

The above channel separation, however, is simplified if ATM transport is implemented, as there is no need to implement all the sub-channels as multiplexing can be done via different virtual channels in ATM. Only AS0 for the fast stream and AS1 for the Interleaved stream in the downstream direction. In the upstream direction there is only a need for LS0 for the Fast stream and LS1 for the Interleaved stream. Using the above simplification makes it possible to implement ATM cell multiplexing. There is now no need to implement defined channels for time division multiplexing, as would be the case with all the ASx channels.

A functional block diagram of the ATU is presented below in Figure 2-5. The illustration is described below and deals mainly with the two unique data paths within the modem.

Both the Fast and Interleaved data streams are protected with a Cyclic Redundancy Check (CRC), A 8-bit CRC is used to check the frame integrity of the super frame. The frame structure is dealt with further in section 2.4 . A scrambler used to improve the random distribution of ones and zeros in the stream follows this. The Reed-Solomon FEC is then added to both the Fast and Interleaved streams. From this point on the 2 data paths begin to differ.

The difference between the two data paths is that the interleaved path adds a convolutional interleaver to improve the performance in the presence of impulse noise. This however introduces latency in that data path.

In both channels the configuration depends on the channel characteristics and the required service needed by the applications. Thus the FEC overhead and the interleaver depth are set at start up to obtain the correct physical characteristics for the link. The ADSL frame is then mapped bit by bit into a complex vector using the Tone Ordering Module and Constellation Encoder Module, then passed to the IDFT to generate the real time domain signal [23].

The receiver is identical to the transmitter in Figure 2-5 above except that the DAC is now an ADC and the IDFT is now replaced by a DFT.

The output stage of the transmitter is not shown in the Figure 2-5 above. The real vector resulting from the IFFT has a 32-sample Cyclic Prefix appended to it. The 512 sample vector has a last 32 samples from (X_n for n = 480 to 512) added and the new vector of 544 samples is passed to the DAC, to become a real signal to be passed through an output stage filter. The longer vector requires a speed up in the DAC so as to preserve the 0.25 ms frame period [4].

2.4 ADSL Frame Structure

The frame structure used in ADSL and the overhead associated with the ADSL frame is outlined here. The next section defines the standard method for multiplexing in an ADSL frame and describes the changes used for ATM cell transport. When considering the frame overhead, the different modes of operation have to be considered. Each mode of operation has different overhead structures associated with it. These structures become more relevant when considering the correct mode of operation for ATM cell transport. Frame structures are described in detail in [43] on page 64-66.

2.4.1 Superframe and Frame Structure

The frame structure in ADSL consists of multiple frames and a super-frame. The sub-frame corresponds to the DMT symbols. The frame structure is outlined in Figure 2-7. The super-frame consists of 68 DMT data symbols or frames, followed by a sync symbol that contains no user or overhead data. The DMT modulator module inserts the sync symbol. In order to preserve the 4000 user data symbols a second that are transmitted on the wire. This is defined in the ANSI T1.413 standard [43] and needs to be preserved. There is a required physical speed up for the sync symbol. The new symbol rate on the wire to include the sync symbol is then (69/68)*4000 symbols/sec [43].

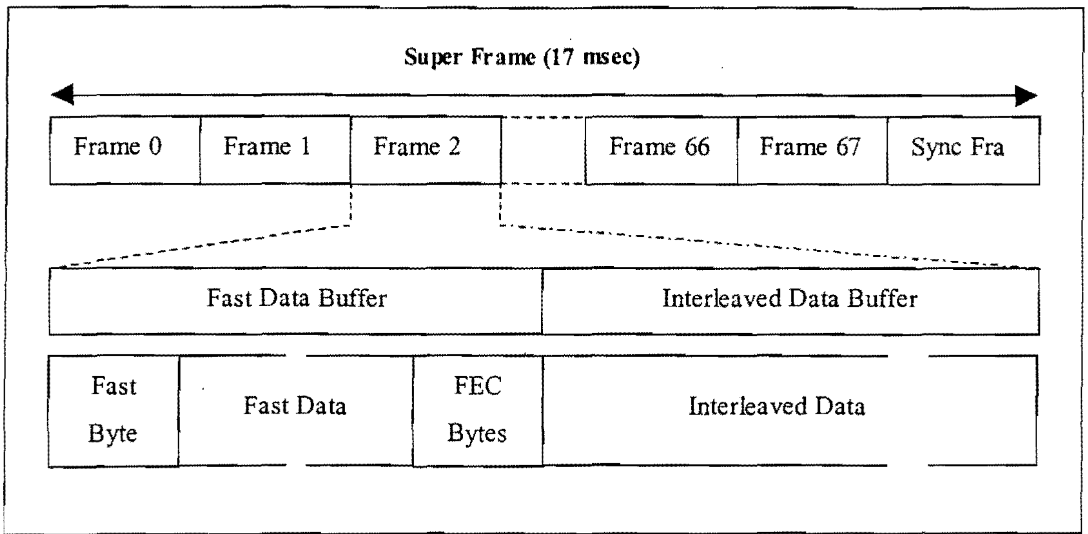

Figure 2-7 ADSL Superframe

Looking closer at the frame information, the fast byte is present in each frame but serves different functions depending on the frame position in the super frame. In frame 0, the fast byte contains the super-frame CRC. Frames 1, 34, 35, carry the indicator bits 0-23. These are assigned for Operation and Maintenance (OAM). In the other frames the fast byte carries embedded operations channel (EOC) and synchronisation control bits (SC).

Figure 2-7 shows that the super-frame is divided between the fast and interleaved data streams. This is normally the case, however, it is possible for the super-frame to only carry interleaved or fast data.

The interleaved data also has a sync byte that is used for similar functions as the fast byte in the fast stream. Frame 0 contains the 8 bit CRC for the interleaved data stream. Other frames carry SC and ADSL overhead control channels (AOC).

The number of bytes allocated to each stream varies, depending on the required data rate for the fast and interleaved streams. This may include the possibility that one of the fast or interleaved channels is not present.

2.4.2 Data Buffer at the Constellation Encoder

After the CRC, Scrambler and FEC the data buffer has the following structure outlined in Figure 2-8 for the Fast Stream below and in Figure 2-9 for the Interleaved Stream. [43]

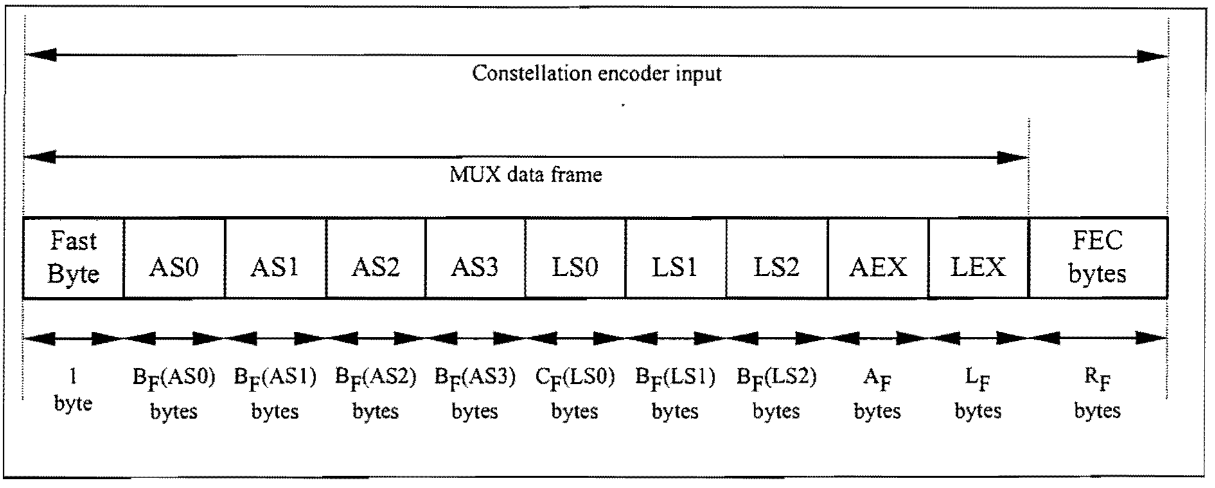

Figure 2-8 Fast Data Buffer at input of the Constellation Encoder

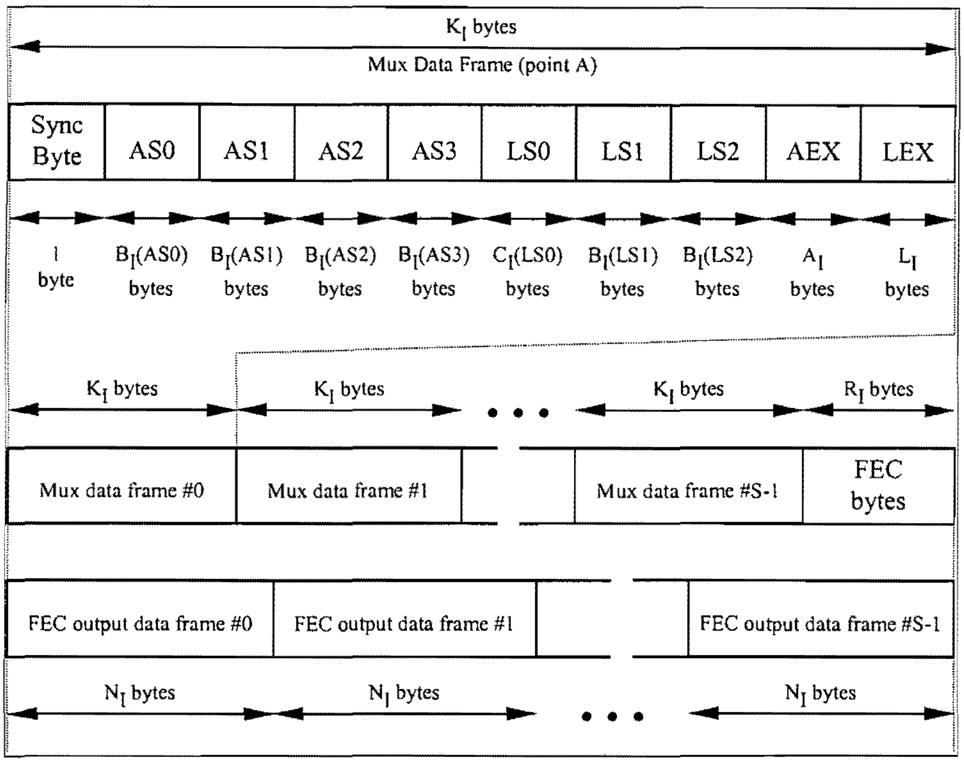

Figure 2-9 Interleaved Data Buffer

The big advantage of ATM comes when operating in a cell transport mode as there is no longer a reason to implement multiplexing on a channel basis. This results in better channel efficiency, simplification of implementation and allows the use of statistical multiplexing of traffic. To achieve this mode of operation, AS0 is assigned to the Fast Stream AS1 to the interleaved stream. Thus in the above data frames there is only a need to implement one channel, the rest of the ASx and LSx channels fall away.

In the upstream direction the data buffer is the same as in Figure 2-8 with the difference that there are no ASx Channels. The same applies for cell transport where the fast stream is assigned LS0 and interleaved stream LS1.

2.4.3 Reduced Overhead Framing

As mentioned above by implementing ATM cell based transport, there is no need to implement all the ASx and LSx channels. This opens up the provision for reduced overhead framing. In reduced overhead framing there is no more provision for synchronisation of the seven ASx and LSx channels.

In this mode consideration is only given to the sub-mode where there is separate fast and sync bytes. The other sub-mode where the fast and sync bytes are merged is not considered. The structure of this overhead is defined in Table 2-1.

Frame	Fast byte	Sync byte
0	Fast CRC	Interleaved CRC
1	ib 0-7	AOC
34	ib 8-15	AOC
35	ib 16-23	AOC
Other Frames	Eoc	AOC

Table 2-1 Overhead in ADSL Superframe in Reduced Overhead Mode

The total system overhead for this mode of operation is shown in Table 2-2 below for both up and down stream, excluding the FEC overhead and any ATM cell or AAL layer overhead.

Reduced Overhead	Down Stream	Up Stream
Fast Data Buffer		
ASx > 0 and LSx = 0	32	--
LSx > 1	--	32
Interleaved Data Buffer		
ASx > 0 and LSx = 0	32	--
LSx > 1	--	32
Total	32 kbit/s	32 kbit/s
Total Full Overhead	96-192 kbit/s	96 or 128 kbit/s

Table 2-2 ADSL system Overhead

It is clear that in using this mode the system overhead can be reduced by 83% downstream and 75% upstream. However there is now extra overhead introduced at the AAL and ATM layers.

Chapter 3 Characteristics of the local loop

ADSL utilises the local copper loop, thus offering Telecom companies the opportunity of augmenting the access bandwidth to transport value-added services over its currently installed copper network. This offers a two-fold solution to Telcos by extending the life span of copper network, which represents a large capital outlay and it offsets the need for them to provide fibre to the home to provide the required bandwidth. This, however, may be a good idea in concept, but in reality ADSL does not work on all copper loops. The performance and operation of the ADSL access network is affected to a large degree by the characteristics of the loop. The local loop characteristics are analysed and the effect they have on the network performance, is investigated in this chapter In order to do this, the loop structure and loop frequency response are presented in the following sections.

3.1 Loop Structure

The copper loop is often thought of as just a bundle of twisted pairs that run between the customer and the exchange. This, however is not the case as in reality, the local loop has a very loose and varying structure.

Most of the local loop is compatible with ADSL. If load coils are fitted to the line it is not possible to use this line for ADSL. These coils are fitted to longer lines to boost the frequency response. The number of loops falling into this category is small but this may vary, depending both on the region and the company that owns these loops.

Lines that are compatible with DSL technology are often referred to as the Carrier Service Area (CSA). The CSA is defined in the ANSI T1.413 [**43**]. The specification of the CSA is highlighted below. Unfortunately there is very little information published about the structure of the actual copper in the ground. The other standards institutes ETSI [91] and ITU-T [90] go so far as to reference the specification in ANSI T1.413. CSA is used in a broad sense to group lines, which are likely to be able to utilise ADSL.

The CSA are loops that fall into the following categories:
Loops of up to 2800m for lines of 0.4mm diameter (26 gauge)
Loops of up to 3700m for lines of 0.5mm diameter (24 gauge) [18].

Figure 3-1 below illustrates some examples of the local loop. These include gauge discontinuities and bridge taps. These same examples are used later in the actual simulation and the evaluation of the ADSL access network.

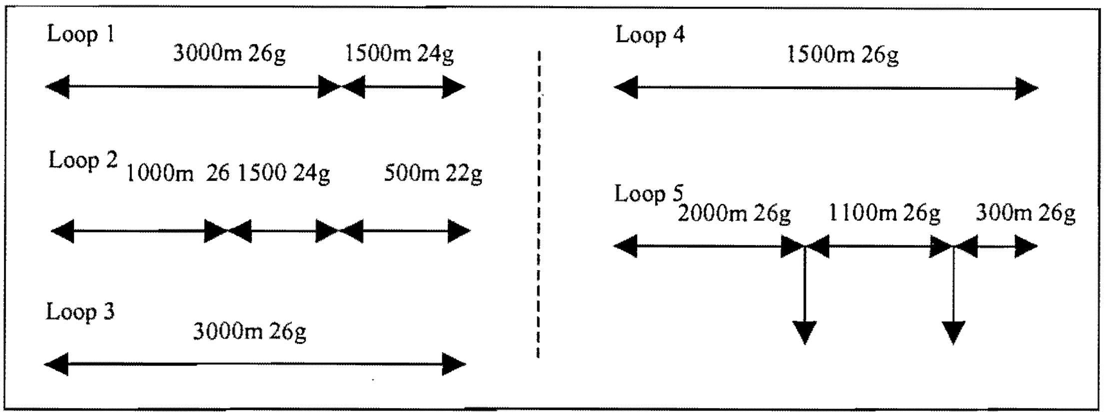

Figure 3-1 Possible Structures in the Local Loop

3.2 Frequency Response

In order to simulate an ADSL network, the local loop over which it is used needs to be included in the simulation. There are two approaches that could be followed when including the effect of the channel on the network. The first is to use a transfer function that represents the loop a function based on frequency and loop length. This, however, simplifies the network as it is not possible to represent changes in wire gauge or bridge taps. The second method, which is described below, includes the effects of bridge taps and gauge changes.

The second method requires that a two-port network represent each different section. The method described in Appendix C is the approach implemented in the thesis. This results in vectors defining the frequency response of the loop with respect to its phase and magnitude response. A program called LINEMOD was obtained from Prof. Cioffi of Stanford University to perform these simulations. David G. Messerschmitt of the Department of Electrical Engineering and Computer Science wrote the program at the University of California.

The frequency response of the channels presented in Figure 3-1 are shown below in Figure 3-2. The following data presented on the loop response was obtained by running simulations using the LINEMOD program on the loops selected to represent the CSA. These loop structures are shown in Figure 3-1 [1]. Appendix E outlines how these loops are described functionally for simulation purposes.

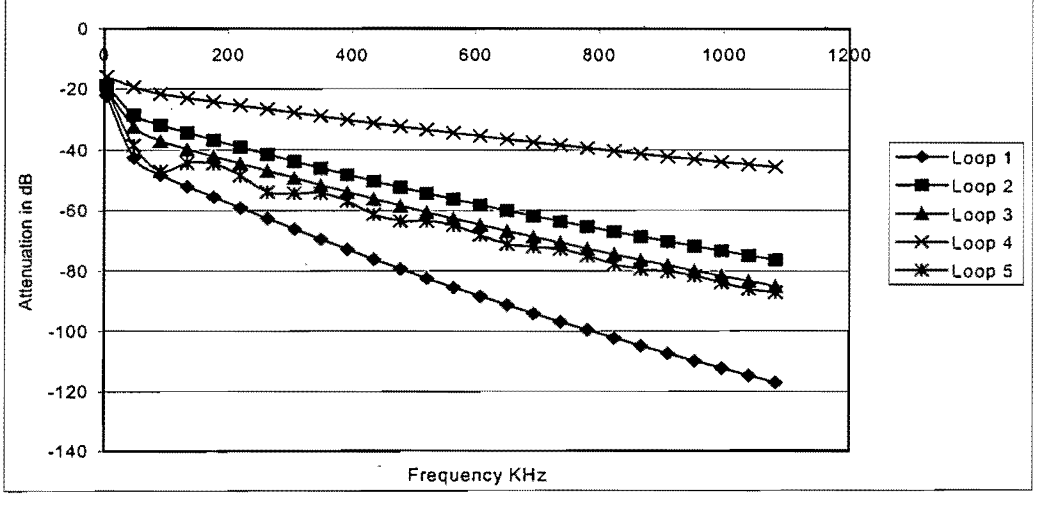

Figure 3-2 Magnitude Response of the Local Loop

It is can clear that the attenuation in the channel is both a function of frequency and line length. Channel 5 shows a different characteristic. Here there is a ringing effect in the magnitude response. Due to the presence of the bridge taps. Bridge taps create notches in the frequency response at multiples of their quarter wavelength resonant frequency. This ringing effect extends to the phase response; the other channels all have a relatively linear phase response, bridge taps result in a highly non-linear response.

Figure 3-3 and Figure 3-4 below has plots of the phase response and the group delay. The group delay is the derivative of the phase response with respect to time. Note the discontinuity introduced by the wrap around at π is removed. The group delay is constant for this channel. The channel has discontinuities at the 2π rad where the response goes negative again. This results in a jump in the group delay, which can be ignored.

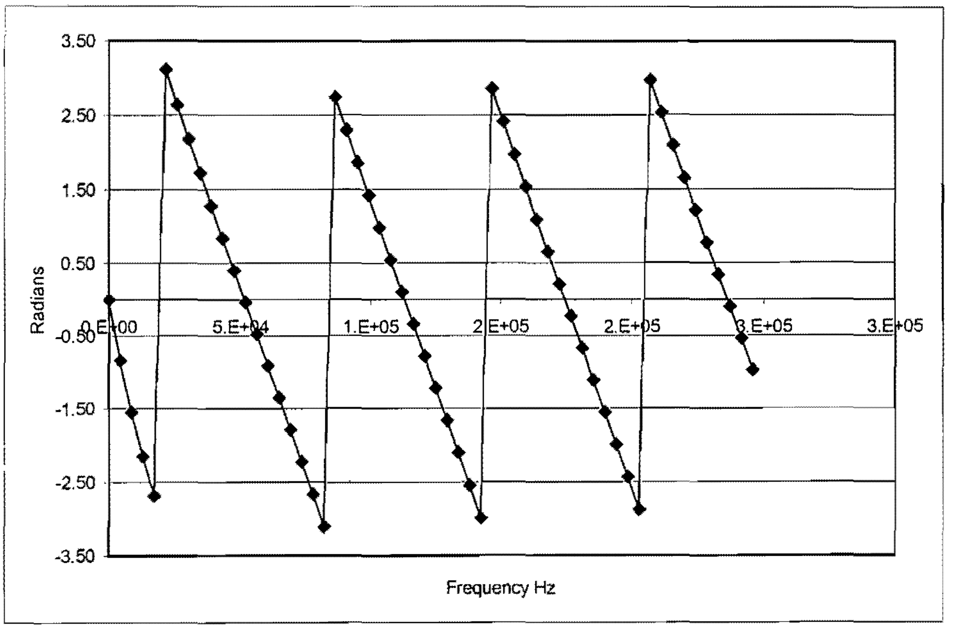

Figure 3-3 Phase Response for a Loop with no Bridge-Taps

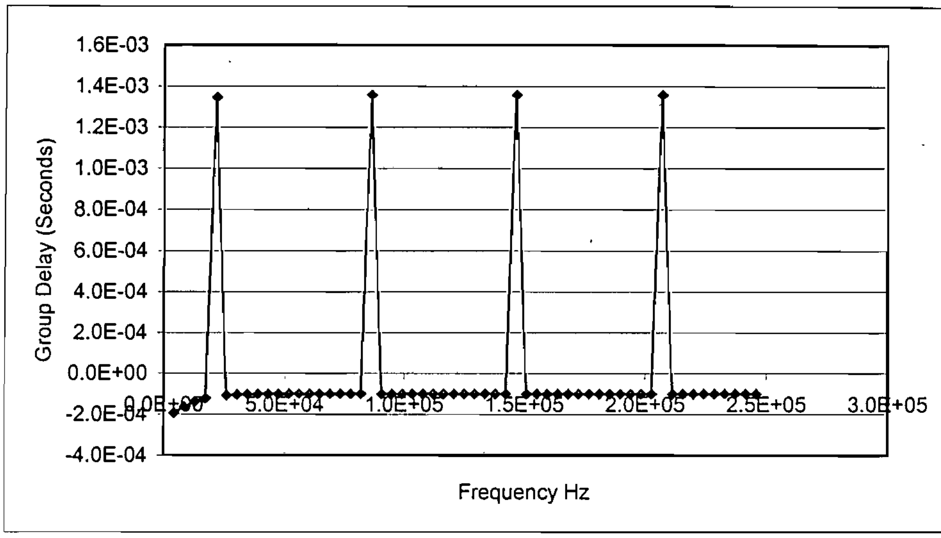

Figure 3-4 Group Delay for a Loop with no Pridge-Taps

The frequency response of the line has been presented above and this can be used to calculate the attainable data rate as a function of the line length for a specific ADSL modem using standard Power Spectral Density (PSD) transmission. This is shown below in Figure 3-5.

These results where obtained by simulating the ADSL link with the above channel response and setting up the noise sources for FEXT and AWGN. The FEC was set up for standard correctable symbols. These rates do not include the safety margin of 6 dB and the effect of impulse noise.

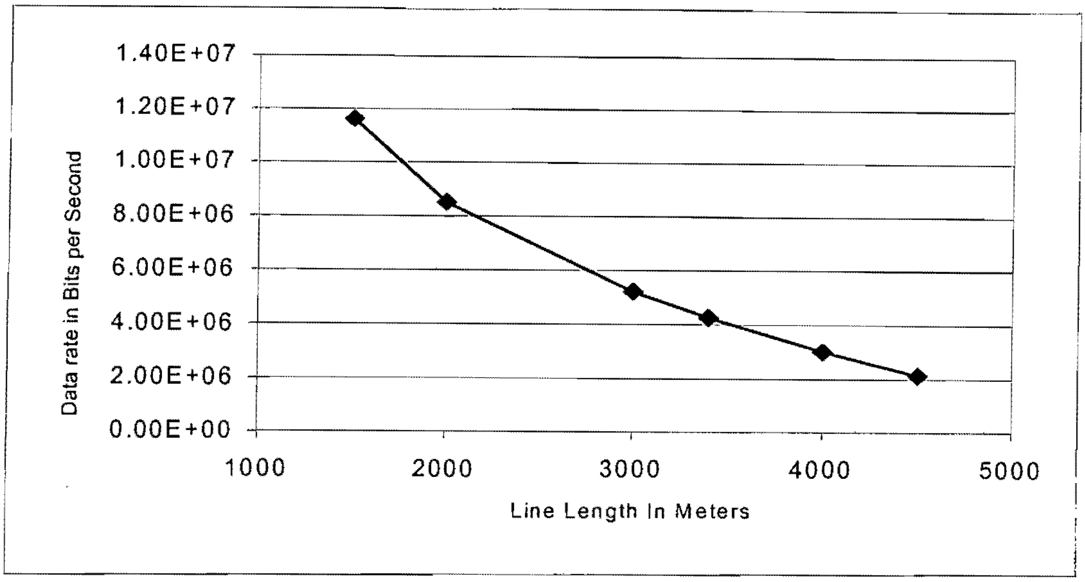

Figure 3-5 Data Rate as a Function of Line Length

The above results in Figure 3-5 show that the attainable data rate decreases as the line length increases. Lines in the local loop that are shorter than 2000m and have good noise conditions, could theoretically exceed the ADSL maximum data rate of 8 Mbps. Thus, on these lines the limiting factor is no longer the local loop but the upper bound of ADSL. A newer technology, VDSL has capabilities of data rates greater than 8 Mbps and could find an application on short cables. Similar results were found in [18] where there is a similar fall-off in the data rate with increasing cable length. However this was for a HDSL system using DMT modulation.

This data rate line length trade-off is a result of operating at higher frequencies, where the loop attenuation increases. This coupled with the fact that crosstalk interference increases at higher frequencies gives rise to this trade-off.

Chapter 4 Noise Sources in ADSL Access Networks

ADSL is an access network technology that works in a noise intensive environment. Channel noise is one of the main aspects that need to be adequately modeled if a good working model of the network is to be realised. The following noise sources are covered in this chapter: far end crosstalk (FEXT), near-end crosstalk (NEXT), background white noise, impulse noise and AM radio interference. Each section outlines the characteristics of the noise source and then defines how that source is modeled.

4.1 Background White Noise

White noise is easy to model and simple to understand. This noise results from background radiation and thermal noise in the conducting media. The noise source has a constant power spectral density. This is shown below for positive frequencies, representing half the noise bandwidth.

$$S_n(w) = \frac{\eta}{2} \qquad 4.1.1$$

As a result it is easy to model both in the time and frequency domain because the time domain is just a normal distribution and the frequency domain a flat distribution. In the time domain this can be modeled with a normal distribution, with a mean of zero and the variance set to provide the correct background noise power. This value has been set to provide Additive White Gaussian Noise (AWGN) of -140dbm/Hz for the simulation. The noise bandwidth in used in the implementation is 2.208MHz [2].

4.2 RF Interference

RF interference due to AM radio is a problem as the ADSL spectrum overlaps with it. Here the implementation of DMT in ADSL is very useful, when it comes to adding the effects of radio interference (RF). DMT modulation can reduce the number of bits in the sub-channel affected by the RF to zero. By reducing the number of bits in that sub-channel the obtainable data rate is reduced but it eliminates the affect of RF interference on the ADSL link. The reduction in RF interference only applies to ADSL implemented with DMT modulation, as in this thesis. If, however, ADSL were implemented on CAP, then this affect would have to be modelled. Basically if RF interference is added to the simulation, it requires setting the effected sub-channel bits per symbol to zero.

4.3 Near End Crosstalk (NEXT)

NEXT is not the major limiting factor in ADSL as a result of the following reasons: the asymmetric nature of the transmission in ADSL, transmit power is lower in the upstream direction and two different sections of the spectrum are used for upstream and downstream. However, there is some spill over NEXT from adjacent DSL services. These DSL services could be ISDNs, HDSLs DSLs and ADSLs. Yet in the environment concerned (mainly the residential area) it would be unlikely to find many of the other digital services in the access network. For this reason only, the NEXT model for ADSL is developed [20][29].

The Power Spectral Density (PSD) plot in Figure 4-1 below shows clearly, that the NEXT has a flat PSD over the band. To model this an extra AWGN noise source has been used with the required noise bandwidth [43].

4.4 Far End Crosstalk (FEXT)

FEXT is one of the major limiting factors in ADSL especially in the downstream direction. FEXT is not as simple to model as the previous two cases and hence requires a different approach. FEXT can be modelled by the following equation 4.1.1 [1]. The equation defines the transfer function used to generate the FEXT. Here $\boldsymbol{k}$ is the coupling constant and is a function of the number of coupling channels, and $\boldsymbol{l}$ is the line length. $\boldsymbol{H}_{\mathbf{channel}}$ is the transfer function of the channel, includes the splitters on the ADSL transceivers. These splitters include a high and low pass filter [43].

$$\begin{aligned} \left|H_{FEXT}(f)\right|^2 &= \left|H_{channel}(f)\right|^2 .k.l.f^2 \\ k &= 8\times 10^{-20}\times (n/49)^{0.6} \end{aligned} \qquad 4.4.1$$

Therefore the PSD for FEXT in an ADSL system can be described by the following equation where the PSD of the ADSL downstream signal is passed through the coupling transfer function [43]. $\boldsymbol{PSD_{adsl\text{-}distributor}}$ is the PSD of the transmitted ADSL signal on the wire. The ADSL specification [43] goes to great length to ensure that the transmitted signal has a flat PSD. Equation 4.4.2 now describes the PSD of the FEXT noise.

$$PSD_{FEXT} = PSD_{adsl-Distributor}\times\left|H_{FEXT}(f)\right|^2 \qquad 4.4.2$$

In order to simulate this correctly, the filter is set up with the required PSD, and random noise from a Gaussian distribution is passed through the filter. The reason for a Gaussian distribution is that it has a flat PSD; the resulting FEXT noise has the required PSD. Figure 4-1 shows the resulting PSD of the FEXT and NEXT noise simulated in the ADSL network.

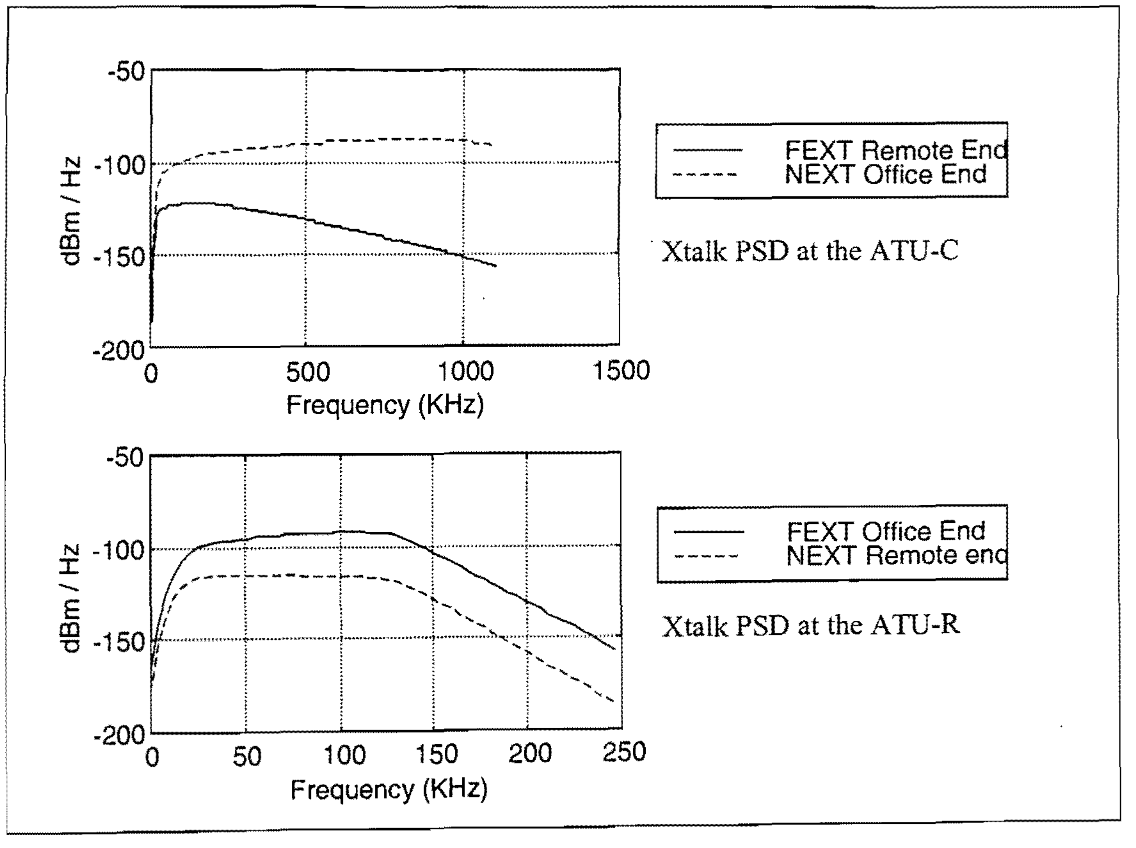

Figure 4-1 PSD for Crosstalk in an ADSL Network

4.5 Impulse Noise

Impulse noise has a random nature, with respect to its amplitude, phase and duration. In the past, little information has been gained about the characteristics of impulse noise, since it does not have a large influence on voice networks. However, impulse noise is very destructive in ADSL networks and as a result is considered here in more detail.

Impulse noise in the twisted pairs results from electrical noise from machinery and environmental disturbances such as lightning. The other main cause of impulse noise in the local loop is from switching and ringing of adjacent lines in the PSTN.

As a result of some studies[32], certain characteristics have been found for impulse noise. These are outlined as follows [32]:

- Occurs about 1-8 times per minute.
- Amplitude ranges between 5-40mV.
- Duration in the range of 30-150us.

The resulting absence of self NEXT results in a good SNR. However the long cable and high frequencies result in the received signal being severely attenuated. As a result of these weak signals, ADSL is more susceptible to impulse noise [23]. This requires impulse noise to be included if the model is to reflect the true network conditions.

DMT modulation makes use of sub-channels each modulated with QAM effectively. This results in the interesting observation that the peak voltage of the impulse noise is not that important, as compared to the PSD of the impulse noise. This was the first method used to model impulse noise effectively based on its PSD. However a lack of measured results and no known mathematical model of the PSD were found for impulse noise.

A second method was used to implement impulse noise in the model. Two test impulses supplied in the ANSI T1.413 standard [43] were looked at. Work carried out by the ADSL forum [43] suggests that these two pulses represent the most commonly occurring impulses in the access network. The testing procedure outlined in the standard required that these two impulses be applied with random amplitude and phase to the signal. The sample impulse outlined in ANSI T1.413 had to be used to model impulse noise due to the lack of real impulse noise data.

Based on this information, the two impulses where analysed and an exponential curve defined to map the envelope of the impulses. This envelope curve became the basis of the model. Equation 4.5.1 below defines the envelope function used to generate the impulse noise. At time t = offset the impulse amplitude is at a maximum and decays with time. Thus two exponentials define the impulse: a growing exponential to maximum amplitude and a decaying exponential thereafter.

The values of the constants are found from the two sample pulses defined in the ANSI T1.413. [43]. The two independent sets are used with equal probability. The decay and growth constants are found by minimising the RMS error between the envelope and the absolute value of the impulse signal. These two samples are shown below in Figure 4-2 and Figure 4-3.

Constants Used	**Sample 1**	**Sample 2**
α	0.04	0.025
β	0.08	0.08

$$
\begin{aligned}
&if\,(t < offset) \\
&Envelope = Amp\left[e^{(offset-t)\alpha}\right] \\
&else \\
&Envelope = Amp\left[e^{-(offset-t)\beta}\right]
\end{aligned}
\qquad 4.5.1
$$

It would not be valid to only use the two samples of noise in the simulation. However these two represent the most likely occurrences of impulse noise. α and β are replaced by normal distributions with a standard deviation of 2 and the mean value set to α and β respectively.

These parameters where found by experiment with the model to meet the properties outlined above and in [32]. These envelopes, when used as filters for white noise, will result in the impulse noise used in the simulation.

Normally distributed numbers are generated. This distribution has zero mean and a standard deviation of 1 and is passed through the impulse noise filter. The exponential envelope filter defined above generates the required impulse vector. This vector is then multiplied by the random sequence generated from the normal distribution and this results in a vector defining the impulse noise.

To make this simple model more realistic the decay and growth constants of the envelope are replaced with random numbers normally distributed with the mean value set to the value found from the test impulses and a specified variance. This had the result of altering the length of impulses and the exponential growth and decay of these pulses. Some resulting impulses from this model are plotted in Appendix B. Figure 4-2 below shows a sample of impulse noise from the standard with the generating envelope overlaid.

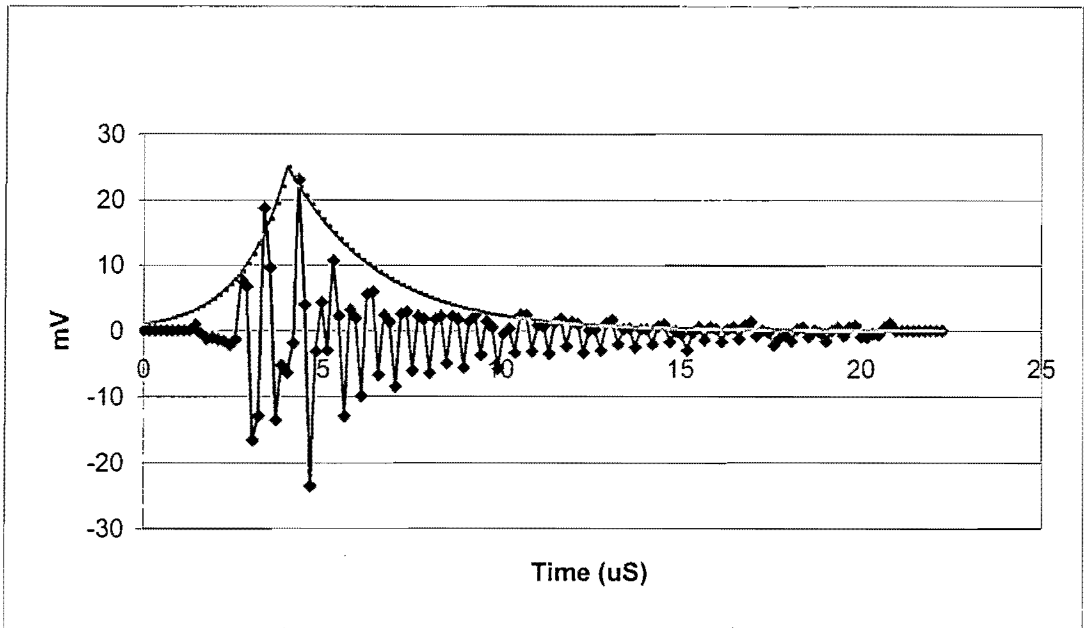

Figure 4-2 Sample 1 of Impulse Noise

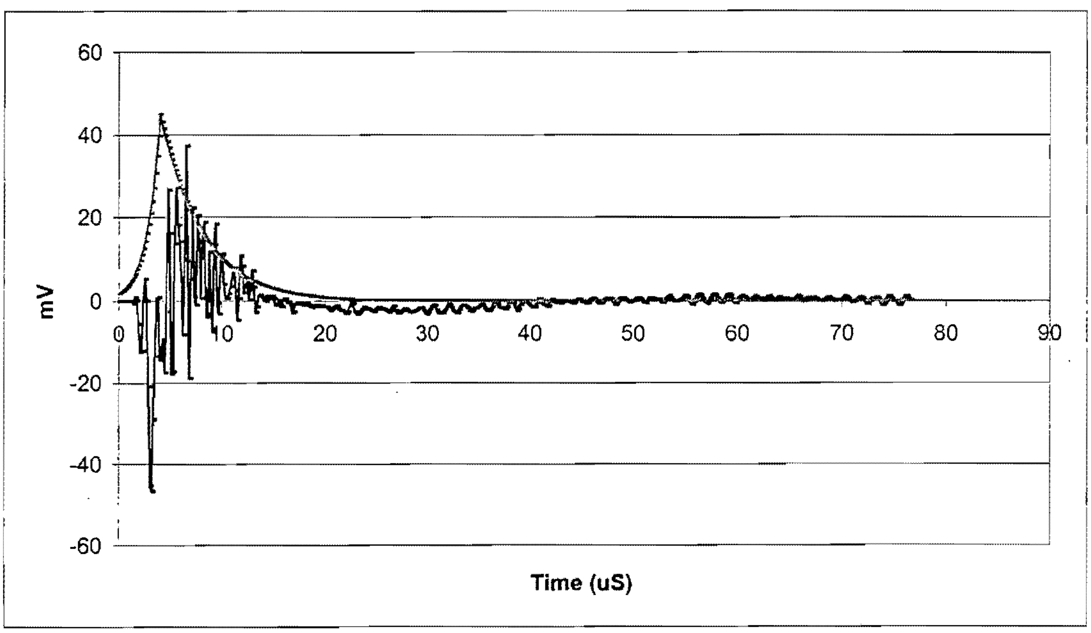

Figure 4-3 Sample 2 of Impulse Noise

The resulting distribution for the impulse noise amplitude and time duration is shown in Figure 4-4 and Figure 4-5. However neither the duration or amplitude distribution are normally distributed. This is because changing the growth alters the duration and decay constant and a non-liner exponential controls the amplitude.

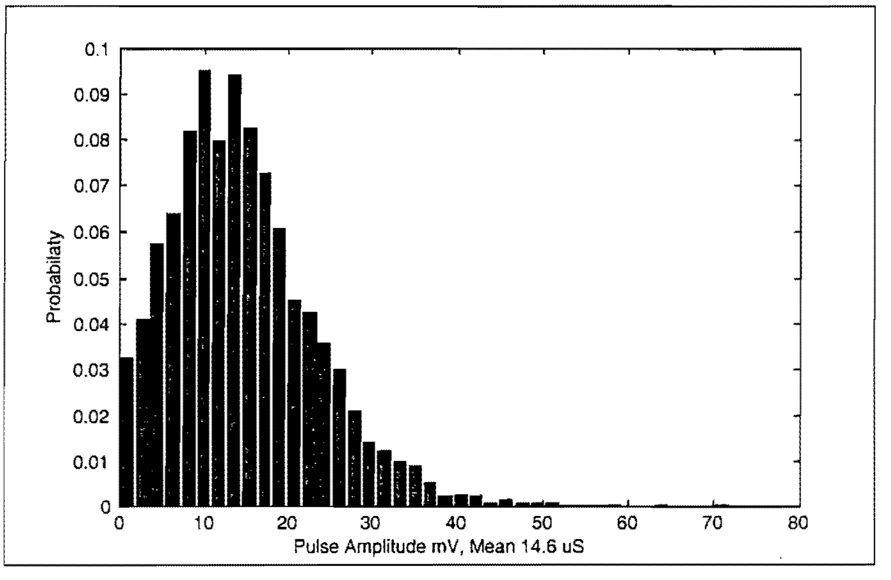

Figure 4-4 Amplitude Distribution of the Impulse Noise

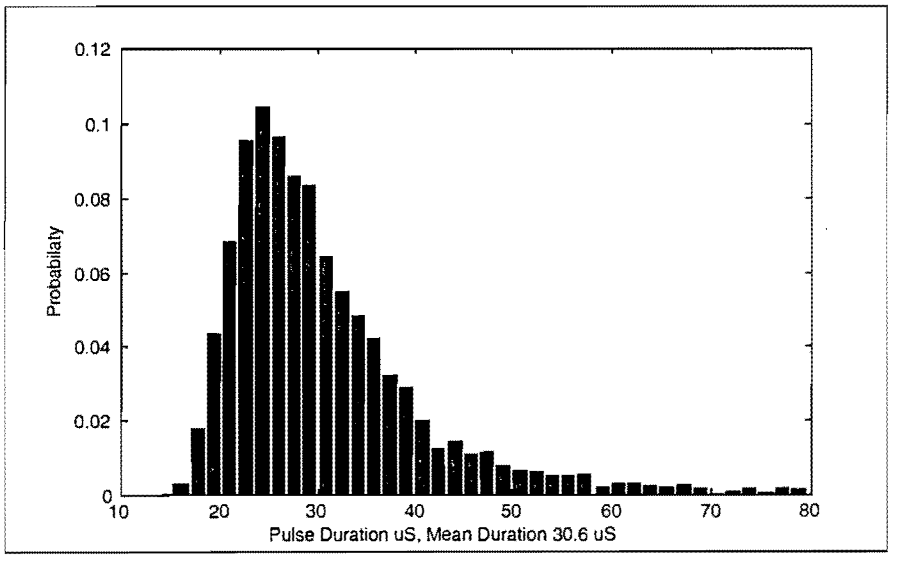

Figure 4-5 Duration Distribution of Impulse of the Noise

It is not possible to verify the distributions of the sample impulse, as there is no real line data or model available for impulse noise. The ANSI standard suggests that you test your ADSL equipment only simulating the 2 pulses defined in the standard. Sample 1 and 2 above are used in the ANSI Standard. This model tries to incorporate a real world aspect to the simulation. Varying of the constants, which define the envelope equation results in a randomly changing pulse, which meets the ANSI standards.

At this point the model can generate a sample impulse noise. However this noise is not always active in the line. The inter-arrival time of the impulse noise is important. The probability function of the inter-arrival time is based on Poisson model [**52**]. This is based on work sampling impulse noise in the German phone network [**52**]. This inter-arrival time is described by equation 4.5.2 below. There λ = mean arrival frequency. In the simulation $\lambda = 0.08$ Hz, this corresponds to 4.8 impulses a minute.

$$P(t) = \lambda^{-\lambda t} \qquad 4.5.2$$

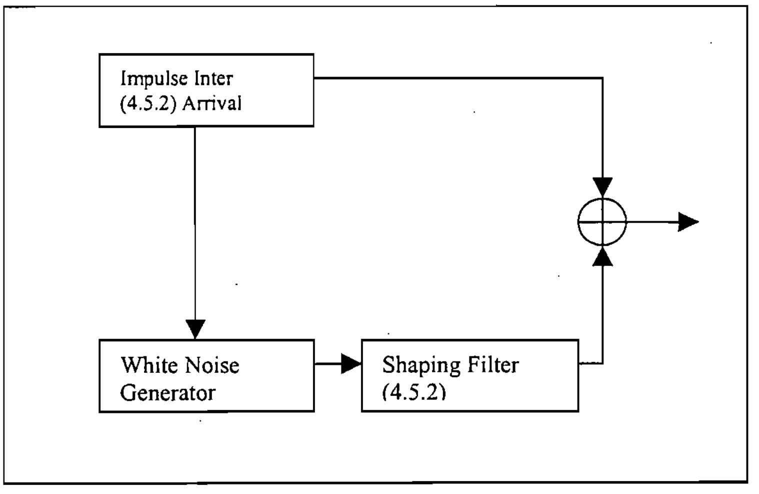

Figure 4-6 Impulse noise generation

4.6 Model of the noise injection in the simulation

Figure 4-7 below illustrates how the noise is added into the simulation. Each different noise is a separate module. The resulting received signal is affected both by the noise in the channel and the frequency response of the channel. Impulse noise differs from the others in that together with a shaping filter it also requires a module to trigger an impulse. This triggering is timed from the simulation time and triggers an impulse based on a distribution with a specified mean.

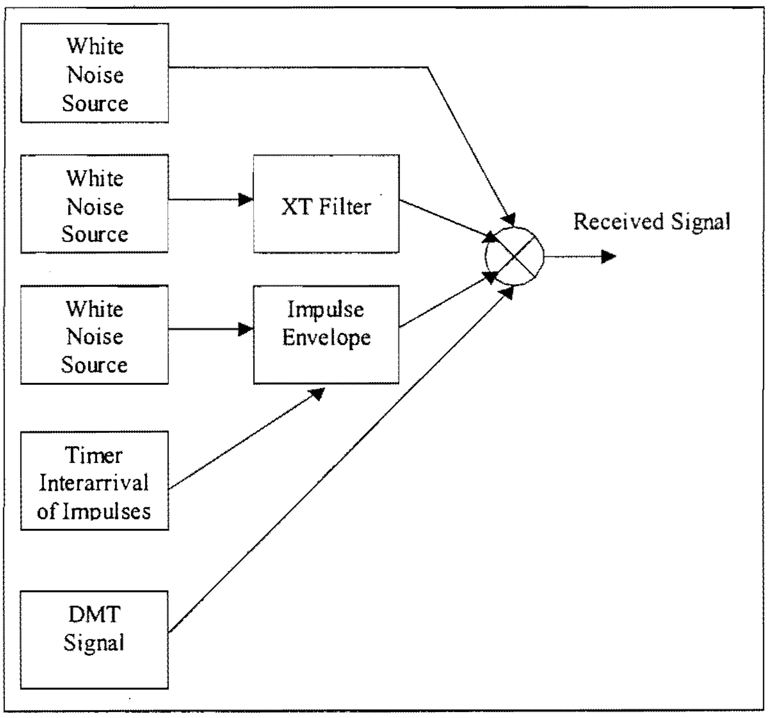

Figure 4-7 Model of the Noise Generator in the Simulator

Chapter 5 ADSL Simulator

In this chapter the outline of the simulator used in this research is presented as well as its implementation. This includes sections on system parameters, Reed-Solomon FEC, Convolutional Interleaver and DMT modulator.

5.1 System Parameters and Implementation

The simulation is written using two different packages. Matlab™ has been used to implement the modulation, noise sources and channel transfer function. C++ is used to implement the simulation control, buffering, Reed-Solomon FEC and convolutional interleaving. The C++ code is compiled using the Gnu C++ compiler for Linux. To integrate these two different implementations a Matlab to C++ converter Matcom supplied by Mathtools is used.

This approach is used to aid the implementation, as Matlab is an excellent language for describing mathematical operations and matrix implementations. On the down side the execution time of Matlab code is very slow. Using a Matlab to C++ converter made it possible to increase the execution speed and integrate it with other C++ code.

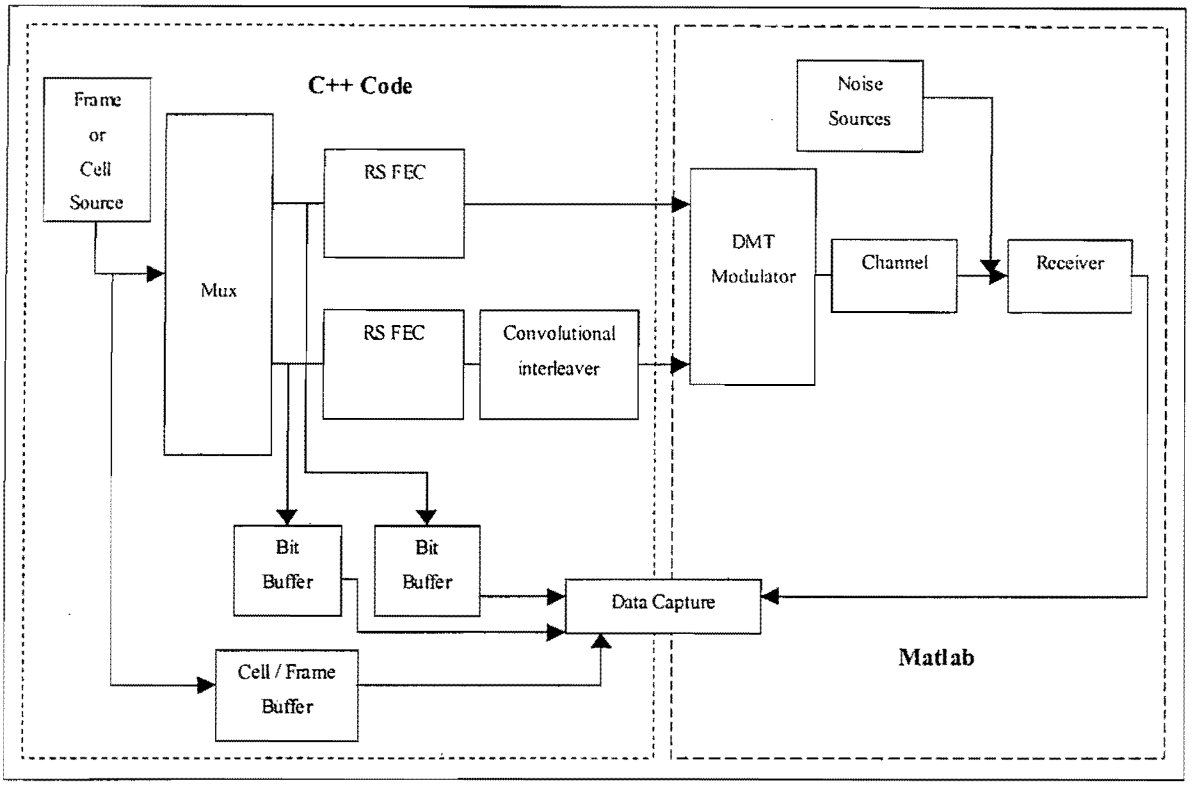

Figure 5-1 Software Simulation Model

5.2 Reed-Solomon FEC

The Reed-Solomon Codec (RS) is a very popular FEC technique for data communications. RS is also used for FEC in many other applications such as data storage and mobile computing. The basic concept in the FEC is to add redundancy to the data message so that if the message is corrupted it can still be recovered. The RS code has become very popular because of it capabilities to recover from multiple errors. This makes it very useful in an environment where impulse noise results in multiple errors.

This module implements the standard Reed-Solomon FEC. However there must be support for shortened code words. This is due to the fact that the code word can represent the entire DMT symbol or part thereof. This arises as both the fast and interleaved data streams share the 255-byte DMT symbol; each of these has its own FEC. Thus, it is unlikely that the code word would be 255 bytes long for the full 8-

bit RS coder. The Code word length is likely to vary between 50 and 150 bytes. There is support for parity bytes 0,2,4,6,8,10,12,14,16.

The incoming data is padded with zeros to 255 bytes, the FEC parity is then calculated and the resulting code word is the original data plus the parity, the zero padding is removed.

In the implementation both the coder and decoder are implemented in the same module. Yet there is a different module for both the Interleaved and Fast Data Streams. Separate modules are needed, as they may not necessarily have the same configuration at any one time.

5.3 Convolutional Interleaver

There are two options when it comes to interleaving of the above RS code. You can use a Block, Interleave / Deinterleave Matrix or a Convolutional Block Interleave /Deinterleave Matrix. Interleavers help spread the error over multiple RS frames, allowing the RS-FEC a better probability of recovering from errors.

The method used in this simulation module, is a convolutional interleaver. The specifications are for such a interleaver are outlined in the ANSI T1.413 [43] standard. To simplify the implementation both the interleaver and de-interleaver are implemented in the same module. Implementing both interleaver and de-interleaver in the same module results in an overlap of the ADSL transmitter and ADSL receiver.

Using this approach, it improves the memory usage, as one only needs to store the setup and configuration data once for both the interleaver and de-interleaver.

The interleaver can support any depth that is a power of 2, however there is a limitation with respect to the code word length. The code word needs to have an odd byte length [43]. If the length of the codeword is even, then the interleaver adds a dummy byte at the beginning of the data and this padding data is removed before transmission.

The interleaver can be defined by the following rule. Each byte of the N byte RS codeword is delayed by an amount that varies linearly with its byte index, in proportion to the interleave depth. Equation 5.3.1 describes this below.

$$Bout_i = (D-1) * index\_InByte \qquad 5.3.1$$

An example of a convolutional interleaving is shown in Figure 5-2 for (Interleaver Depth) D = 2 and (codeword length) N = 5. In example, Byte 0 is delayed 0 intervals, Byte 1 is delayed 1 interval and byte 2 is delayed 2 intervals. This leaves empty slots, which are filled by data from the previous frames. If this is the first frame the slots are set to 0.

Interleaver Input	B^j_0	B^j_1	B^j_2	B^j_3	B^j_4	B^{j+1}_0	B^{j+1}_1	B^{j+1}_2	B^{j+1}_3	B^{j+1}_4
Interleaver Output	B^j_0	B^{j-1}_3	B^j_1	B^{j-1}_4	B^j_2	B^{j+1}_0	B^j_3	B^{j+1}_1	B^j_4	B^{j+1}_2

Figure 5-2 Convolutional Interleaver

Two revolving data buffers are used to implement the convolutional interleaver. There are two buffers as each object implements an interleaver and a de-interleaver. The buffer size is allocated depending on two factors, the interleaver depth and the RS codeword length.

The interleaver effectively gives a coding gain by de-emphasising the noise over multiple frames. This gives the RS-FEC a better chance to correct from the errors introduced by the channel. The net effect is a free coding gain as there is no extra overhead. As a result there is no drop in channel efficiency, there is no extra overhead. The convolutional interleaver does introduce latency into the data stream. The latency grows linearly with the interleaver depth. The latency generated by the interleaver is defined by the equation 5.3.2 below.

$$Delay = 2 * Interleave\_Depth * 250\mu s \qquad 5.3.2$$

This can result in a maximum delay in a one-way transmission of up to 32ms if the interleaver depth is set to 64 and up to 4ms in the upstream direction the maximum interleaver depth is limited to a depth of 8. The delays above represent the maximum delay allowed by the ANSIT11.413 specification [43].

5.4 Framer

The framer is used to control the data from the Fast and Interleaved Data Streams to the DMT modulator. The Framer has to implement this framing on a bit by bit basis where the incoming frames are divided into 255 groups, each group with the correct number of bits per symbol for the appropriate sub-channel.

The module implements the above functionality by concatenating the interleaved and Fast Data stream bytes and then masking off the required bits for each sub-channel. This section is also written in C++ so as to be able to manipulate on a bit level efficiently and quickly.

The framer also implements the Tone Ordering needed in the ADSL transmitter. In Tone Ordering, the Interleaved Stream is mapped to the lower sub-channels first, while the Fast Stream is mapped on to the remaining channels. This Tone Ordering is done to limit the effect of clipping in the DAC. The reason that the interleaved data stream is mapped to the lower channels is because this is where most the clipping occurs. As the interleaved stream is more likely to recover from errors introduced by the DAC clipping, it is allocated to the lower channels. These lower channels have a better SNR [34] as a result carry a greater number of bits per symbol. These lower channels are more likely to undergo clipping in the DAC.

5.5 DMT Modulator

Channel initialisation, a component of the DMT modulator used in the simulation, is presented. This initialisation is used to obtain the channel properties. Following this a section on the constellations and gain scaling is presented. Finally, the section representing the modulator implementation used in the simulations is outlined. This entire section is developed in MATLAB and converted to C++ for integration into the simulation.

In this section there has to be a Real to Binary conversion and as this is a computationally intensive task, a large lookup table is used to improve the simulation execution speed. This comprises a large 2 dimensional array where the one row index is the real value to be converted and the return row is the required bit vector. The required numbers of bits are masked off, as there is a 15-bit resolution. This is set by the maximum number of bits per symbol supported in the ADSL system.

5.5.1 Channel Initialisation

In this section of the simulation the following procedure is followed to determine the attainable data rate. During channel initialisation the number of bits per channel are calculated using a bit-loading algorithm.

The bit-loading algorithm outlined in Appendix C is used for the allocation of the bits to each sub channel based on the SNR for that sub-channel. This channel initialisation is done in the presence of both white noise and crosstalk. However it is not possible to train the link for impulse noise, as it may not be present during training and because of its random nature.

In practise, to find the channel response, a pseudo random sequence is transmitted down the channel, The SNR and the channel response would be found, based on what should have been received and what was received. This is not implemented here, as there is prior knowledge of the channel response and the noise variance for each noise source. Therefore the data rate is calculated directly using the bit-loading algorithm.

5.5.2 Constellations and Gain Scaling

After channel initialisation, the constellations with the required number of bits per symbol for each sub channel in the simulation are generated. These are generated recursively from base constellations outlined in Appendix D. The Euclidean distance between the constellation points is defined in the channel initialisation with the bit-loading algorithm. This distance is used to define the spacing in the constellation. However in Chapter 2 it was shown that the RMS energy in the constellation varied, depending on the number of bits per symbol. As a result a gain-scaling factor is added to equalise this unwanted influence.

To implement the encoding constellation is relatively simple. A matrix lookup is generated. The bit stream passed from the framer for each sub-channel is converted to a real number as the index to lookup table. The range here depends on the number of bits per symbol. As a result the binary input stream is mapped to 256 real numbers and these are then mapped to 256 complex points.

To implement the decoding constellation is more complex as the noise introduced in the channel, moves the received complex points in both the x and y dimension. The final method used to decode is twofold. Firstly the incoming complex symbol is scaled so that it is moved entirely into the first quadrant i.e. there are no negative values. As a result the constellation is transformed into the first quadrant. Secondly if the received point falls out of the new constellation bounds, it is clipped to fall on the boundary. The real and imaginary components are mapped into integer values. These are used to index a two-dimensional lookup matrix. The matrix then returns a binary number representing the original bit stream. The integer range is from 0 to 2^b-1, where b is the number of bits per symbol for that sub channel. This mapping into an integer range is performed based on the closest valid constellation point using the Euclidean distance.

5.5.3 Final Modulation

This section is the final stage before the real time domain signal would be placed on the channel. This section uses the built in FFT and IFFT functions in Matlab® to implement the core modulation. To insure that the resulting signal from the IFFT is real and can be passed to the DAC, the input vector to the IFFT has to have Hermitian symmetry.

The input vector is mirrored with its conjugate and the resulting vector is 512 samples long and with Hermitian symmetry, guaranteeing a resulting real signal. This module adds the cyclic redundancy and sync symbol to the data.

5.6 Simplifications

Simplifications in the simulation model of the ADSL access network are:

- Perfect synchronisation between transmitter and receiver.
- High computational resolution
- Perfect knowledge of the channel transfer function in the equaliser.
- No quantisation effects for the ADC / DAC.
- Cyclic redundancy removes the intersymbol interference.
- No loss of data in the control channel for the simulation.
- The Pilot Carrier is not implemented in the DMT modulator, as perfect synchronisation is assumed above.

Chapter 6 Results

In this section results obtained from simulations of the ADSL access network as outlined in the previous sections are presented. The first results presented, are the attainable data rates in the presence of crosstalk. Following this the distribution of errors is presented in the presence of crosstalk and then in the presence of impulse noise and crosstalk. In all the results the effect of AWGN is included in the simulation.

6.1 Crosstalk Limitations

In this section the limits imposed by crosstalk are investigated. Here the network is simulated with AWGN and FEXT as the limiting factors. The network is simulated with increasing transmitter PSD to achieve maximum reach and data rate.

When considering an improvement so as to extend the reach and data rate by increasing the transmitter power, it is found that a limit was rapidly reached. As the power is increased, the increasing FEXT noise degrades the attainable data rate. These results are shown in Figure 6-1 for all five of the loops presented in section 3.1 . The results show that FEXT is the major limiting factor in ADSL access networks and that increasing the transmitter power to improve the SNR beyond a fixed point is not the solution. The solution to obtaining further increases in the data rate lie with the FEC and interleaving, which are investigated in subsequent sections.

This is for a network with 10 coupling ADSL FEXT sources in the same bundle. If the number of sources is increased this has the effect of reducing the attainable data rate. However the transmitter power has the greatest limiting effect as the coupling is proportional to the transmit power hence increasing FEXT.

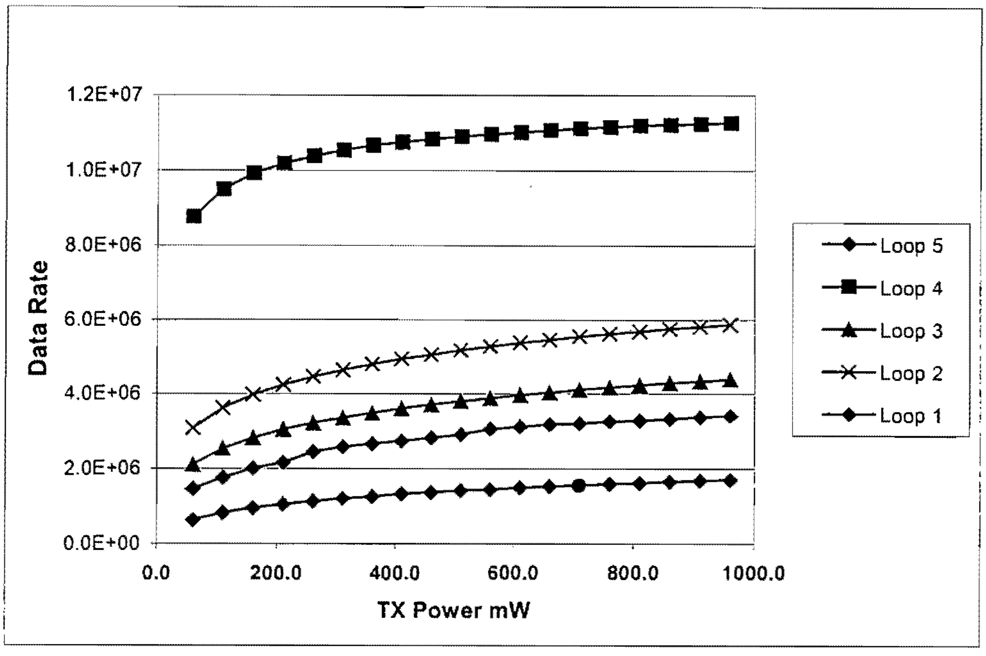

Figure 6-1 Data Rate as a function of the TX Power for 10 Coupling FEXT Channels

6.2 Bit Error Distribution in ADSL

The distribution of bit errors in the ADSL frame corresponding to the DMT symbol has been investigated. This is simulated for different network configurations with relation to FEC overhead and interleaver depth. There are two considerations with respect to noise in the network. First is to only consider the effect of constant noise sources, namely crosstalk and AWGN. The second consideration is the effect of impulse noise on the error distribution.

The reason behind this investigation is based on the fact that ATM is a protocol that should operate in a relatively noise free and low BER environment, such as fibre optic media (BER $=10^{-12}$) [15]. As a result ATM is designed to have a high throughput and for very few cells corrupted by noise in the network. However the above does not apply in an ADSL access network.

The network in which ATM was designed to operate is considered to have a high SNR and very low BER of 10^{-12} or better. These fibre optic networks offer this environment for ATM and have bit error distributions where single bit errors are the most likely errors to occur. The bit error distribution is shown in Figure 6-2 [26] [11]. This is compared later to the distributions in an ADSL system.

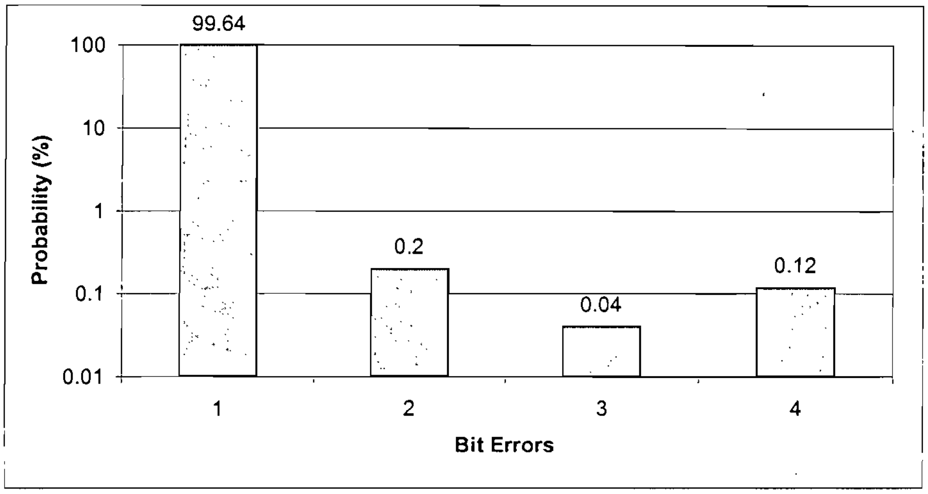

Figure 6-2 Probability Distribution of Errors over a Fibre System

6.2.1 No FEC and No Impulse Noise

In this simulation the different channels are set up under the same conditions where AWGN and FEXT are present and there is no impulse noise in the network. The aim is to find the distribution of errors in the ADSL frame for a BER of 10^{-7}. The results are shown below in Figure 6-3.

There is no correlation between ADSL errors and a fibre network where in the latter single bit errors make up 99.64 % of an errored second. Here most of the errors are single and double bit errors. However it is clear that the difference is that there is a good probability in an ADSL network for the bit errors to go as high as 7-bits per frame with a probability of 2% in an errored frame. The implication that this has on ATM cell transport is outlined in section 6.3 .

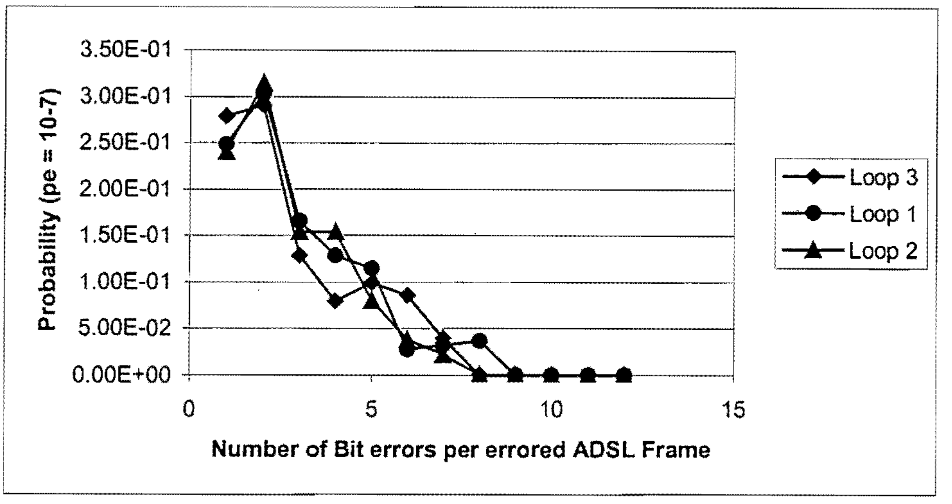

Figure 6-3 Probability Distribution of Bit Errors in an Errored ADSL Frame

The next section looks at the result of adding RS FEC to the ADSL connection to improve the data and BER rate.

6.2.2 FEC Introduced and No Impulse Noise

The following results are for an ADSL channel simulated under normal conditions with FEXT and AWGN noise present and the PSD set to the values outlined in ANSI T1.413 specification [43]. There is no impulse noise present in this simulation and hence the convolutional interleaver is set to 1. The distribution can be seen in Figure 6-4 and Figure 6-5 below. The aim of this simulation is to look at the effect that the RS-FEC has on the distribution of errors in the ADSL frame.

Looking at the data in Figure 6-4 where for a fixed FEC of 16 bytes the data rate is changed and the data is plotted for different BER from $4*10^{-3}$ to $1 *10^{-7}$, it is clear that there are no errors below 8 bit errors per frame. The worst case is where there are 8 single bit errors all in different bytes in the code word and the FEC fails. Once the FEC fails the frame will normally decode with a 0.5 BER The distribution widens as the BER increases and this can be expected because for a higher BER there must be more bit errors per frame.

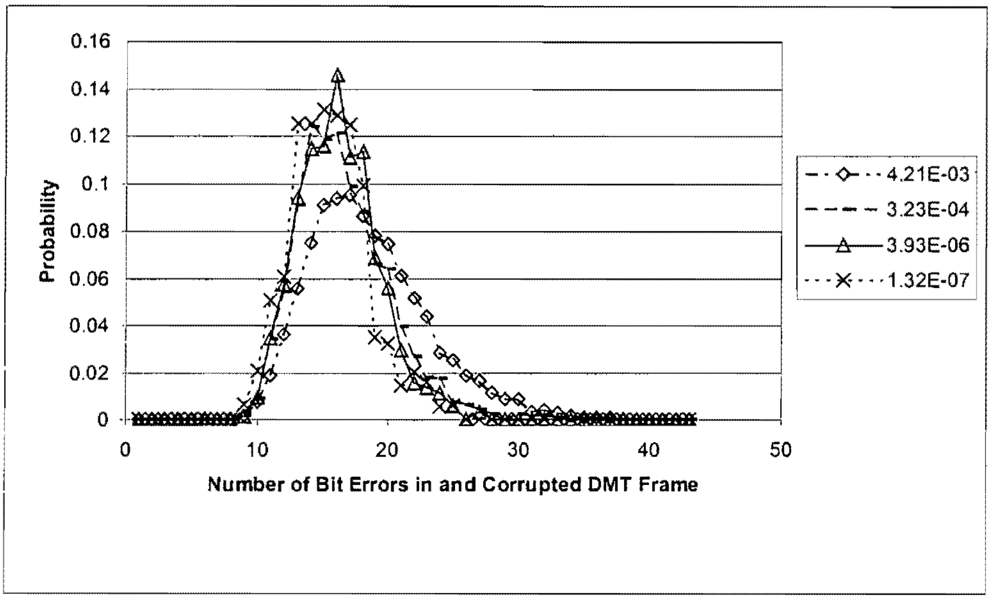

Figure 6-4 Probability Distributions of Bit errors in a Corrupted ADSL Frame. The FEC is set to 16 bytes of Overhead.

Considering the distributions for different amounts FEC overhead these results are presented below in Figure 6-5. Results are presented for FEC with 8,5,4,2 correctable symbols. The data here has the same properties as above. However the distributions are shifted along the X-axis depending on the amount of FEC parity introduced. The distribution width is governed by the BER and in ADSL this is set up for a constant 10^{-7}.

Once the FEC fails, there are more symbols with errors than are correctable symbols. In this case the data in the resulting frame will have a BER of 0.5. Thus the entire frame has to be discarded. There is no way to know what data is still valid in the frame.

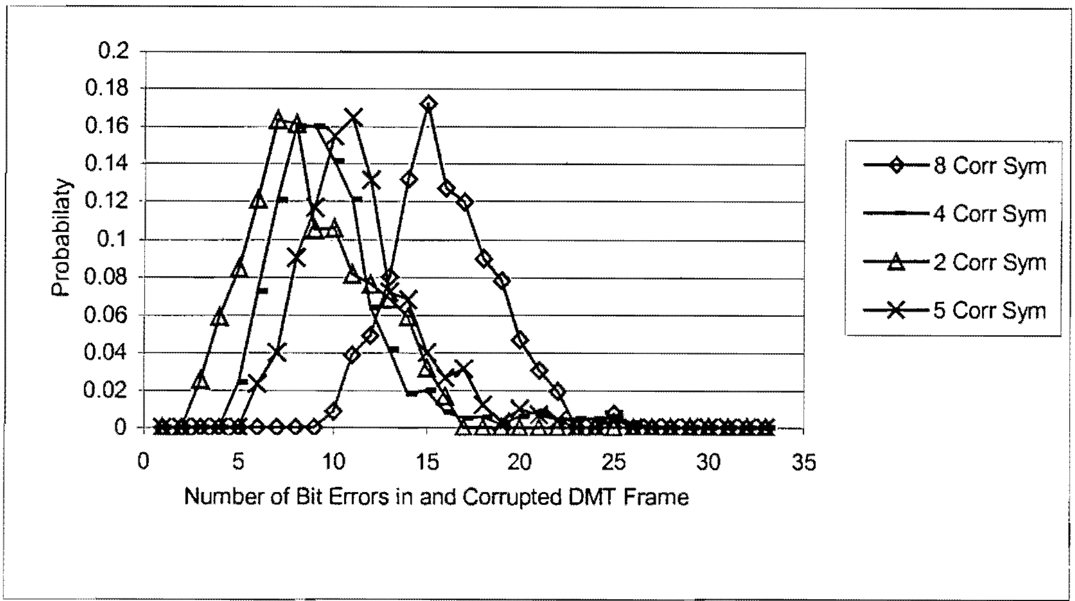

Figure 6-5 Probability Distributions of Bit errors in a Corrupted ADSL Frame for different FEC Overheads.

6.3 Effects of Impulse Noise

The effect of impulse noise on the ADSL system is important, with respect to the distribution of the errors and the attainable data rate achievable in these conditions. Impulse noise is the major limiting factor for transporting ATM traffic in the access network and the simulations carried out in this section are used to analyse the effect of impulse noise on a cell transport level.

The following results are for an ADSL channel simulated under normal conditions with FEXT and AWGN noise present and the PSD set to the values outlined in ANSI T1.413 specification [43]. Impulse noise is now added and the distributions of errors per errored frame are investigated under different network conditions for FEC and interleaver depth. The interleaver and the FEC are the major components in offsetting the effects of impulse noise. The interleaver depth and the amount of FEC overhead inserted to obtain the required BER and QoS needed for ATM transport affect these results.

The inter-arrival time and the distribution governing the inter-arrival time for impulse noise are important. However there is little knowledge as to the distribution characteristics and for this reason it is modelled in these simulations as a normal distribution. There are reasons to not use a normal distribution as some of the impulse noise is as a result of ringing and therefore will have a distribution that is affected by the characteristics of the ring. Yet with no physical data to model this from the only solution is to assume a normal distribution.

Results are presented below in Figure 6-6 and Figure 6-7, for simulations, run with differing interleaver depth and two FEC settings of 8 correctable symbols and 4 correctable symbols respectively. These graphs outline the differing nature of the bit error distributions under different configurations. However the channel is configured for an overall BER of 10^{-7} and with the 6dB margin allowed this BER improves to 10^{-13}. This all falls away with the introduction of impulse noise and BER increases radically if the correct interleaver depth is not used.

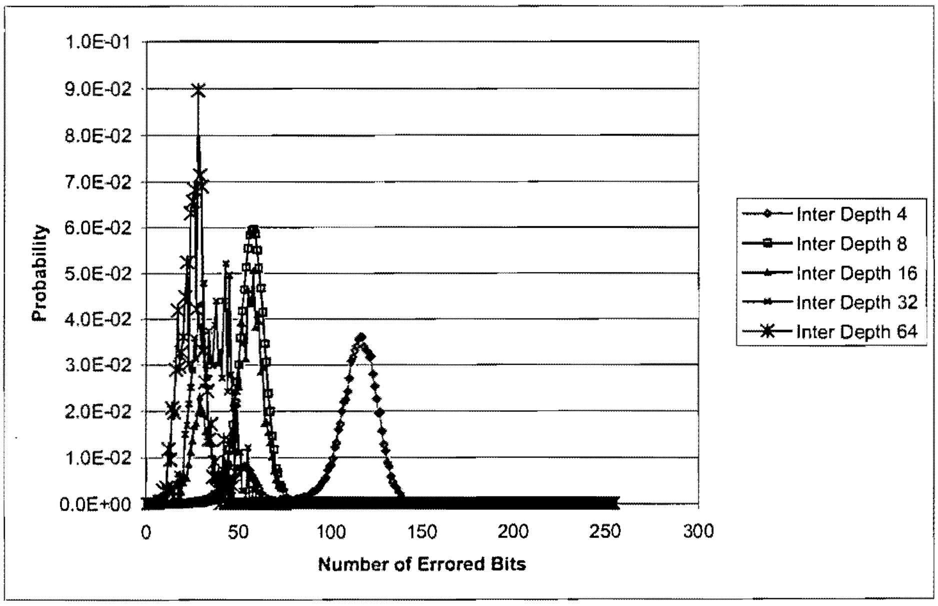

Figure 6-6 Error Distribution with FEC of 8 Correctable Symbols

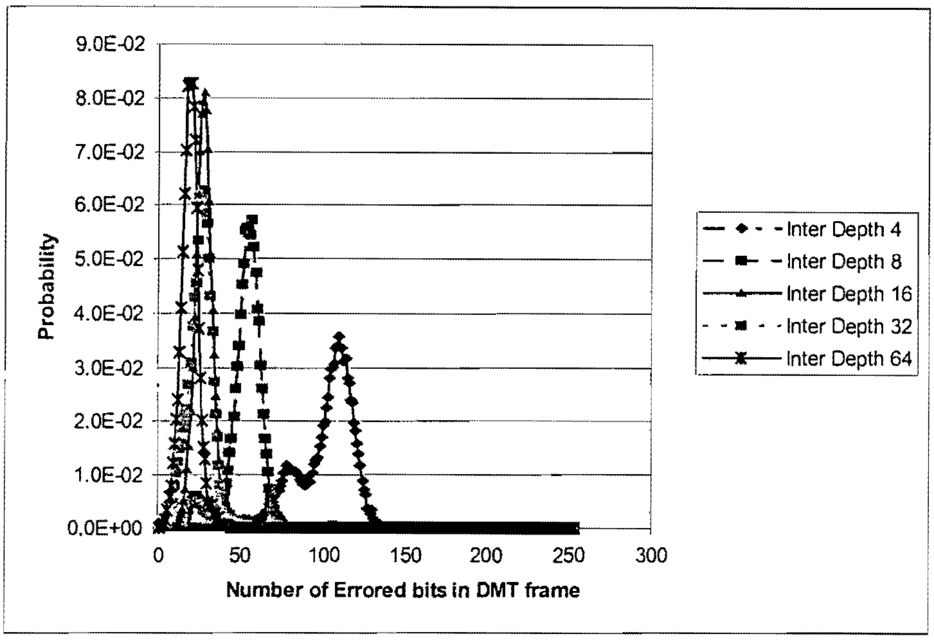

Figure 6-7 Error Distribution with FEC of 4 Correctable Symbols

The true value of the interleaver is shown in the following results. In these results the discontinuity can clearly be seen in Figure 6-8 and Figure 6-9. These results are plotted for different FEC overhead as function of the interleaver depth.

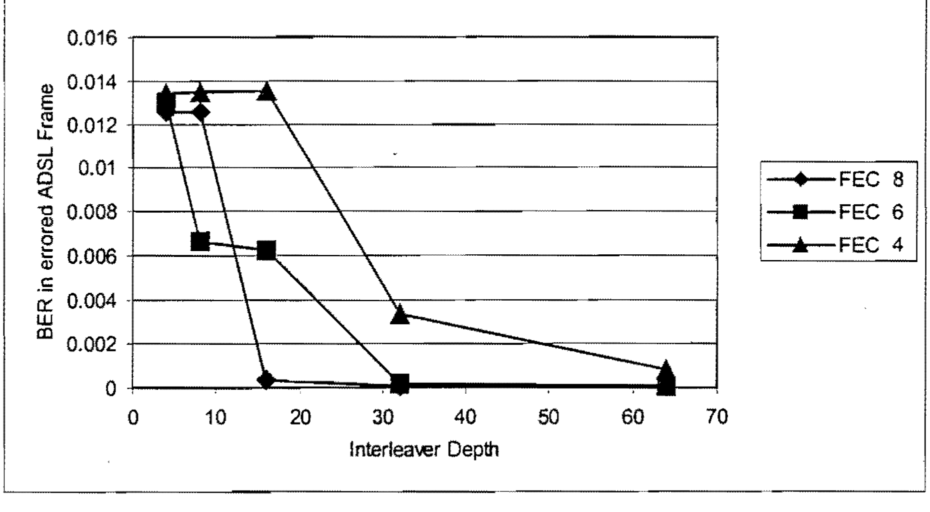

Figure 6-8 BER in an ADSL frame corrupted by Impulse Noise as a Function of the Interleaver Depth

From Figure 6-8 it is clearly shown that the BER under the effect of impulse noise in the ADSL frame drops at a specific interleaver depth for a set FEC overhead. Thus it is important to use the correct interleaver depth for the required conditions. The general trend both here and in Figure 6-9 is that with increasing interleaver depth the immunity to impulse noise is higher.

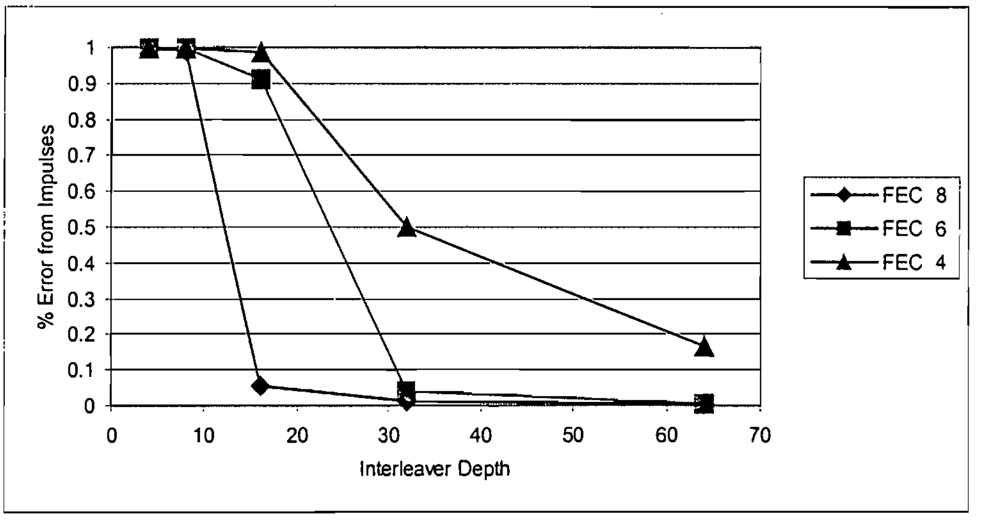

Figure 6-9 Probability of an Error Causing Impulse as a Function of the Interleaver Depth

Table 6-1 highlights additional information available about the simulation results presented above. The frame size changes as the FEC is changed for a given FEC overhead and the resulting BER. The channel efficiency changes as a function of the FEC overhead. These are highlighted in section 6.4 . The critical point where a large increase in the effective data rate is obtained is the optimal point for normal operation.

Correctable Symbols	**Frame Size**	**Data Rate (bits/Sec)**	**Effective Size**	**Effective Rate**	**Efficiency**	**% Speed Increase**
8	1256	5024000	1128	4512000	0.898	30.55 %
6	952	3808000	856	3424000	0.899	-0.003 %
4	928	3712000	864	3456000	0.931	0 %

Table 6-1 Results of Channel Efficiency and Data Rates

6.4 Channel Efficiency

ADSL uses shortened code words in the RS-FEC As a result, care must be exercised when increasing the number of parity bytes. The coding efficiency decreases rapidly as the number of parity bytes increases for shortened codes. However the ADSL frame size does not grow at the same rate and this results in a slow decrease in the channel efficiency. This is shown in Figure 6-10. This effectively results in a point being reached where adding more FEC overhead does not improve the effective data-rate.

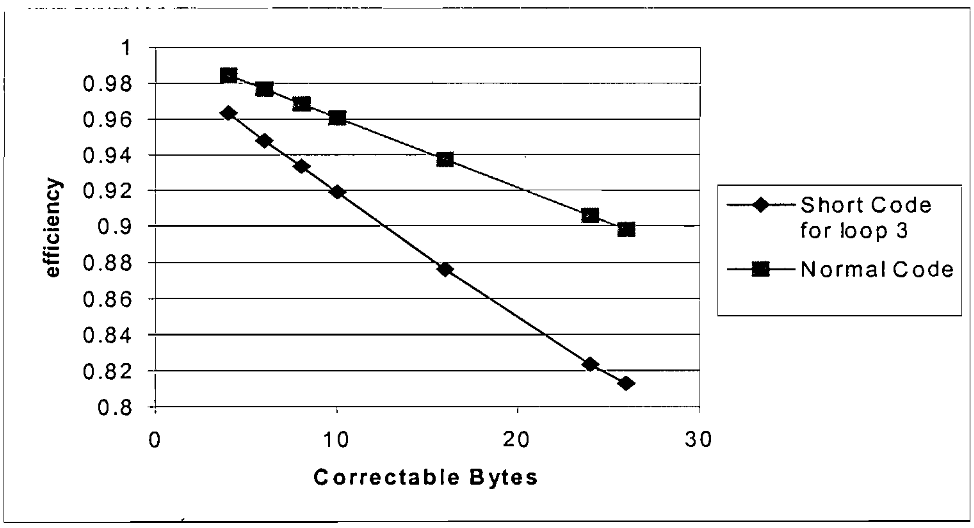

Figure 6-10 Code Word Efficiency for Shortened Codes

Chapter 7 ATM over an ADSL Access Networks

In this chapter ATM cells transported over an ADSL access network is presented. Specific focus is placed on ADSL areas that affect the transport of ATM in the ADSL network.This also extends to the functionality that the access network should offer for efficient ATM operation.

7.1 Functionality needed in the ATU and DSLAM for ATM Transport

This section deals with the implementation aspects, specific to the transport of ATM over the ADSL access network. This section concentrates on the ATM layer and the physical layer. In the physical layer the focus is on the transmission convergence sublayer.

There is also a debate over how to implement ATM over a physical layer with two data paths, each with different characteristics. This has to do with the two physical paths in the modem, which have different latencies. The solution here is to think of ADSL not as a single physical channel, but rather as two different channels, with different physical properties, sharing the same media. The handling of this property is described in the following sections.

Looking more closely at the ATM transport over ADSL, PDU's in layers higher than the ATM layer are transported transparently across the access network. The AALx layers do not have to be implemented in any of the ADSL hardware. The only time that this differs is if the ATU-R is built into a PC network interface card (NIC). In this case the AAL layer is likely to be implemented in the hardware, this is shown at the bottom of Figure 7-1. In this figure the ATU-R and the NIC are one logical unit. In this case, AAL 5 would be supported on the NIC. The reason behind implementing AAL 5 on the NIC is that it suits most applications, which may use this new access network now or in the future. AAL 5 supports: ABR, UBR and VBR traffic and these cover most applications.

If the DSLAM and the ATU-C are grouped together these can then be collectively defined as the Access Node. The Access Node forms the connection between the POTS physical layer and the core ATM network and must be able to perform the following functions:

- Routing / demultiplexing in the down stream direction. The demultiplexing here is implemented at the ATM layer. The cells are switched to the correct ATU and then to the fast or interleaved stream in the ATU based on a VPI/VCI identifier.
- Concentration of ADSL streams in the upstream direction into a single ATM stream.

The DSLAM also has to implement the transmission convergence (TC) sublayer, as well as the ATM layer. There is a TC sublayer for each stream. The TC sublayer can be placed in the ATU-C and this would remove some of the functionality from the DSLAM. The TC sublayer implements all the standard ATM functionality of cell delineation, HEC generation and idle cell insertion and deletion. This can be seen below in Figure 7-1.

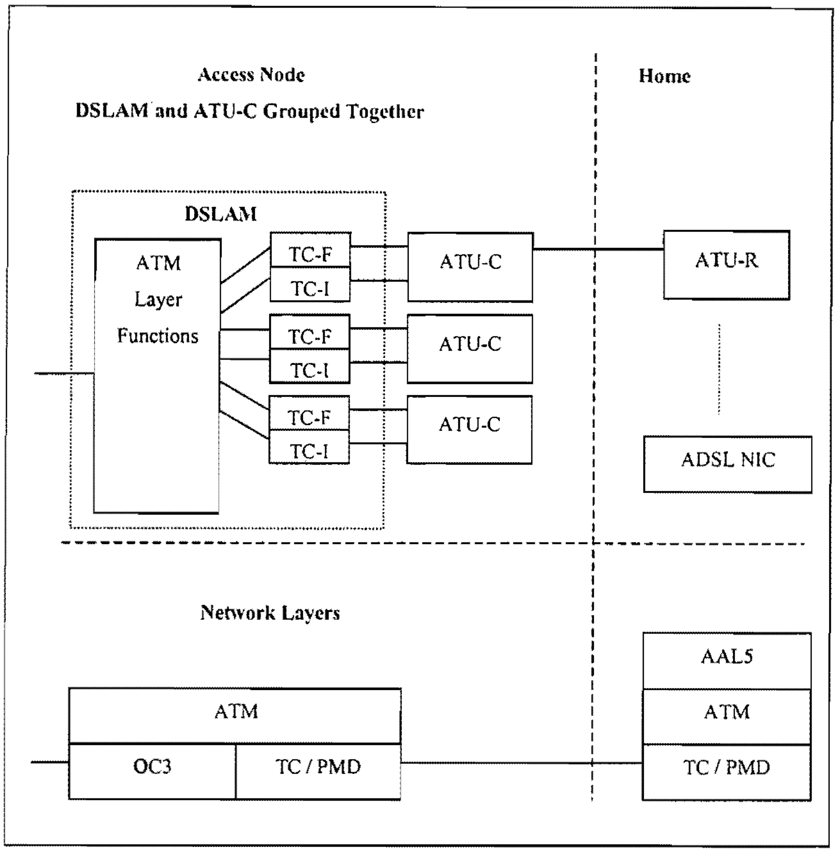

Figure 7-1 Access Node Functionality

7.2 Dynamic Rate Adjustment in ADSL

ADSL has the ability to alter not only the data rate but also the modem configuration with respect to FEC and interleaver values. To implement Dynamic Rate Adjustment the line conditions are monitored and communicating the information between the ATU-R and ATU-C using the ADSL overhead control channel (AOC).

The modems to monitor the line conditions and trade the new link configuration without affecting the user data flow. However when they signal the switch to occur on a frame boundary, data may be lost.

Dynamic rate adjustment may result in more than 10ms of user data being lost in both communication directions. The rate adjustment could possibly violate the QoS contracts negotiated in ATM call setup. As this change occurs on the physical level it is not possible to confine the cell loss to a specific VC, which is carrying non-QoS specific traffic.

This dynamic rate-adaptation in ADSL has not been eliminated from operation when the modem is configured to carry ATM traffic. However it is possible to suspend this operation. The ADSL forum has suggested that a better solution would be not to implement the rate adjustment when carrying ATM traffic or to reserve this for when there is no specific QoS traffic in the link [44].

7.3 Access Network Functionality needed for ATM Transport

If the network is to operate efficiently with a large number of ATM users on the access network, it is important to be able to handle resource management at the DSLAM where all the users access networks are connected to the core network. This basically means that the access network is capable of performing dynamic resource management at the ATM virtual connection level [44]

This makes the implementation of the access network more complex. On the other hand it is a more flexible solution to provide this functionality in the access network. As it frees up resources in the core network, there is no need for signalling VC's in the core network for each subscriber in the access network.

In order to perform the functions described above it is important for the network to have the following capabilities:

- To distinguish between cells on different VC and to perform ATM layer concentration and switching.
- Signalling.
- Perform cell level scheduling.
- Call Admission Control (CAC).
- Ability to process and negotiate ATM service classes.
- Knowledge of and capability to allocate its own resources.

7.4 ATM: A Means of Providing End-to-End Interoperability in ADSL Networks

The underlying technology that makes an end-to-end interoperable network possible is ATM, where there is ATM between the customer premises and the service provider. The service provider and the network provider are different in this discussion. This separation is not always the case and they may be one and the same. In general though the network provider sells the connection and the service provider sells an application service like Internet access and/or Video on Demand applications.

ADSL is a layer 1 protocol, then using ATM as layer 2 protocol has the following advantages:

- The network is independent of layer 3 protocols such as IP and IPX. ATM therefore provides protocol transparency.
- The scalability in bandwidth of ATM matches the rate adoption of ADSL. This allows for efficient network usage.
- ATM allows the ability to implement statistical multiplexing in an ADSL link, thereby increasing the bandwidth efficiency for the end user.
- There is already development of other DSL technologies such as Very High Speed Digital Subscriber Line (VDSL). If the underlying access technology changes in the future, there is no need to change the current network configuration.
- ATM allows the support of QoS classes in the access network. As a result ATM provides end-to-end guaranteed QoS to the end user.

7.4.1 PPP over ATM

Point to Point protocol (PPP) over ATM is an important addition to ADSL as it allows ADSL access networks to look and operate in much the same fashion as current dialup access networks. The difference now is that the connection to the ISP is via a broadband network. PPP can be supported by UBR or ABR service, as it is used for IP traffic with little or no guaranteed QoS. This section is included for completeness and is not an integral part of the work laid out in this document.

As Internet access is considered to be the main application for ADSL in the future, there needs to be a easy way to integrate the current ISP to the access network and hence to the end user. If this is to happen sooner than later then PPP over ATM is the solution [8].

PPP enables the following to be delivered over ATM:

- Authentication.
- Layer 3 configuration. Here the ISP network can assign DNS and IP address.
- Layer 3 transparency. IP and IPX are supported by PPP.

To implement the above option PPP is placed on AAL5. Thus each VC carries a PPP session. This model adds QoS to the current ISP configuration as each ATM VC carries a set PPP session and hence gives each of these PPP links an associated QoS.

7.4.2 End User Configurations

The configuration at the end user depends on what the user's needs are as well as those of the operators. However, the aspect of end configurations is examined here with a long-term solution that is both cost effective and not functionally restrictive.

With the above outline the following available configurations are looked at:

- Ethernet to ADSL Bridging/Routing modem.
- ATM/ADSL adapter in a standalone PC.
- ATM25 to ADSL modem.

Ethernet to ADSL Bridging/Routing modem has been considered a good early solution, but has many problems. In this solution the modem is an external box that terminates the ADSL line and converts ATM to Ethernet interface. The first question one should ask is why extend ATM the entire way to the home and end it in the last few feet where an Ethernet network is now bridged? By doing this there is a loss of functionality and QoS as we can no longer extend guaranteed QoS to the end user. They will have to use a MAC layer protocol and IPX or IP with no guaranteed QoS. This solution does, however, support legacy network applications well, but does not function with respect to new broadband networks. Broadband applications are seen as the biggest draw card offered by ADSL. This solution would be easy for the end user to configure, as the customer side would just look like an Ethernet network. Of course, it would be far more costly as more functionality would be required in the end modem.

ATM/ADSL adapter in a stand-alone PC is a better solution than the above because it has the added benefit that it can support native ATM all the way to the desktop. This enables applications which need guaranteed QoS to be implemented. In addition unit cost price in supplying a NIC is much cheaper than the previous solution. However there is a drawback with regards to the service provider. There is lack of control with network configuration, which is now up to the end user. However, this is more of regulatory question and is not covered here.

ATM25 to ADSL modem is the most dynamic solution. The cost would be higher than that of a stand alone ADSL NIC. This is because the user also requires an ATM25 NIC in the PC, which interfaces to the ADSL external modem. But the solution has other advantages that support it well. ATM25 to ADSL modem is more functional as a single end user or a local ATM network is supported. For the local

ATM network case a workgroup style ATM switch connecting the remote workstations to the ADSL access network would be used. This solution can also provide the services offered by the Ethernet modem described above as there can be end to end support for legacy networks using LANE, CLIP or MPOA [25].

The last two solutions offer the best functionality and cost, because ATM is been utilised end to end in the network. They do not impair the operation of the network and are able to offer all the services of the legacy Ethernet modem as a subset.

7.4.3 PVC and SVC support over ADSL

In this section PVC and SVC setup in the access network is considered. The main aspects considered are ease of setup and deployment and also long term network growth.

PVC provides the most straightforward approach to establishing an end-to-end connection in the management plane. Here the administrator will pre-configure the access network with a set of PVC for each user with the required QoS and traffic profile needed [8]. Unfortunately this becomes a growing administrative problem as the network extends. However, the only advantage here is that the DSLAM does not require extra functionality. Based on the above reasons, early deployment of ATM and ADSL will have PVC configuration.

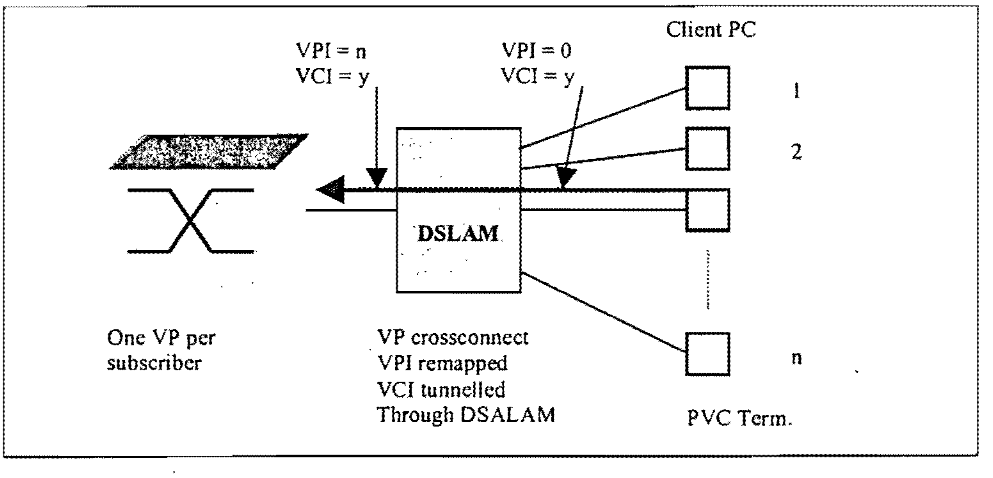

Figure 7-2 PPP over ATM using PVCs

If ADSL is to survive in a large-scale deployment then the network must support call setup in the control plane [40], thus providing SVC to the end user. There are two ways of implementing this in the access network. The first is to implement a virtual UNI, and the second is to use the DSLAM to terminate the UNI signalling.

In the first option the DSLAM is not aware of UNI signalling and there is a PVC for each end user through the DSLAM to the ATM access switch. The problem is that there is still dedicated PVC for each user to the ATM access point. This is better than the PVC option presented above, as the PVC does not exist across the whole network but only up to the access switch. This concept of virtual UNI is outlined in the UNI signalling 4.0 [8].

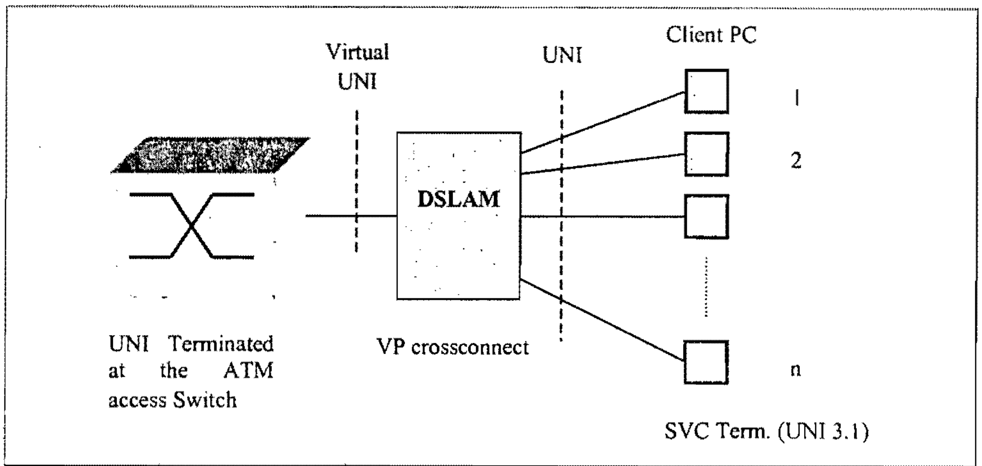

Figure 7-3 UNI Signalling Terminated at the ATM Access Switch (Virtual UNI)

The second option is the most complex and as a result may probably see a delayed deployment if implemented. By increasing the functionality of the DSLAM to be UNI aware and in an essence making it the access switch to the ATM core network the cross connect problem with a PVC per user is eliminated. This reduces the signalling channels needed at the access switch as there is only one signalling channel needed from the DSALM to the ATM access point.

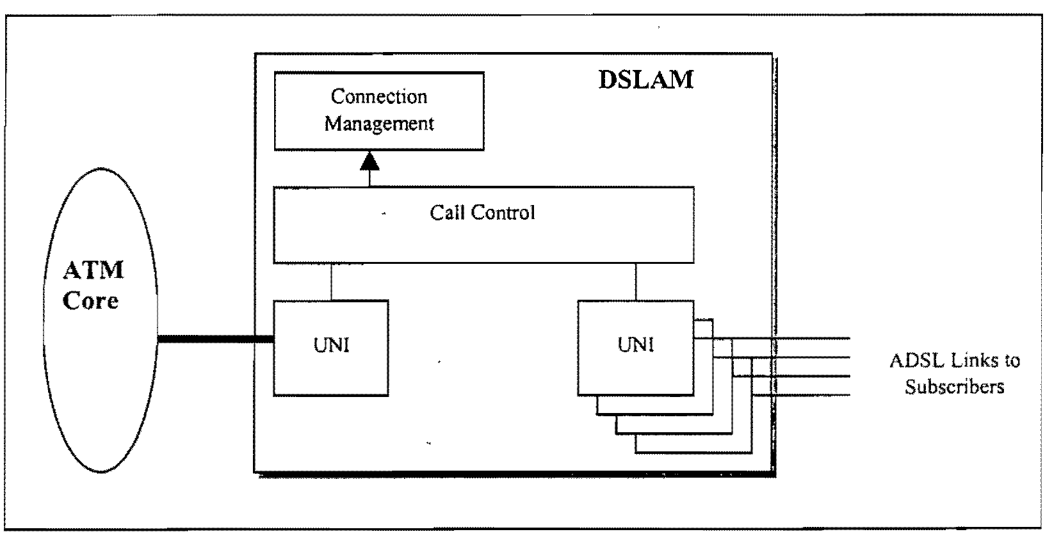

Figure 7-4 SVC Support, Signalling in the DSLAM

7.5 Flexible Data Encapsulation

As a result of ATM over ADSL, a flexible data encapsulation method will be provided that is independent of the underlying ADSL network. ATM provides an abstraction over which all services can be provided.

As a result using ATM over ADSL provides the following possible encapsulation methods:

- Ethernet frames over ATM, as defined in RFC 1483 [47]. (Encapsulates Ethernet frames in AAL5 PDUs.
- IP over ATM, using RFC 1755 [48] (Classical IP). Provides encapsulation of IP packets in AAL5 PDUs.
- PPP over ATM allows rapid deployment to fit in with current Internet Service Providers and allows remote configuration for IP and DNS.
- Native ATM provides new properties of low latency and guaranteed QoS etc. This in turn also allows new broadband applications to be developed for the access network.

By utilising ADSL with a cell based method there is far more network flexibility. Providing a better solution to the end user.

7.6 Communication Protocols Stack for ATM Across the ADSL Network

In this section the different protocol stacks that could be used in ADSL are presented. These outline the functionality ATM offers to the access network. From this it is clear that ATM over ADSL is the unifying protocol for this multi-role access network.

The protocol stacks for the different applications are shown in Figure 7-5. Clearly ATM offers the consistent link to providing the required functionality.

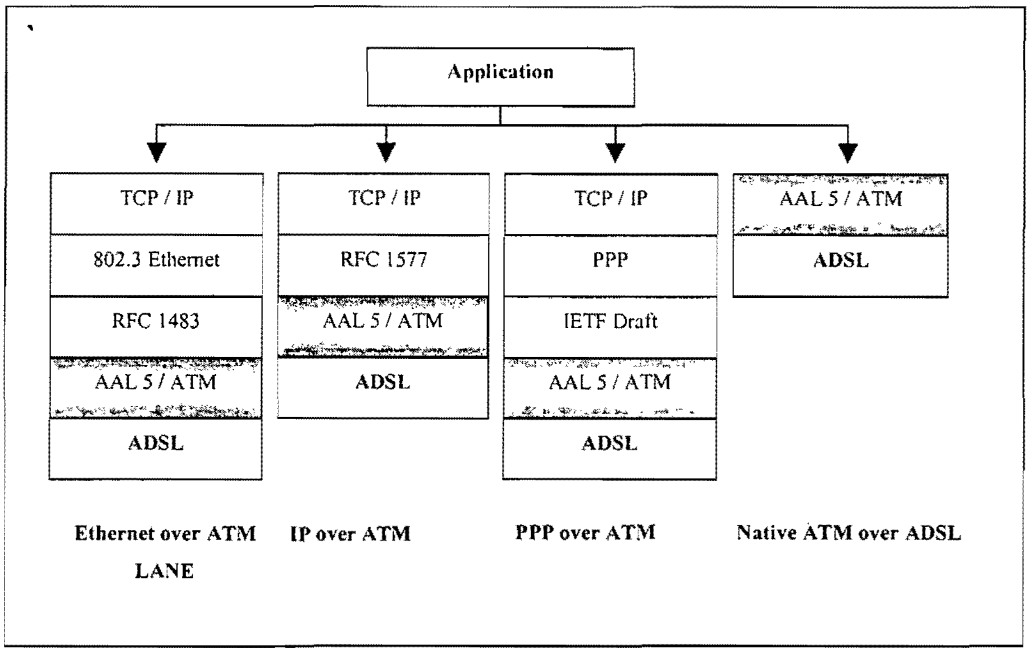

Figure 7-5 Encapsulation methods for ADSL

7.7 ABR and UBR Traffic

There are a few major traffic types to consider in ATM. Available Bit Rate (ABR) and Unspecified Bit Rate (UBR) are the most important traffic types when considering an ADSL access network network in the last mile to an IP subscriber. VBR can also be considered but the only application likely to use it is VoD. In this section specific focus is placed on ABR and UBR and the problems imposed on them by the ADSL network characteristics.

7.7.1 Available Bit Rate (ABR)

ABR traffic is generated from applications that compete for the remaining available bandwidth in the network. The only guaranteed QoS is the minimum cell rate, which can be set to zero. No other specific cell throughput guarantees are made. If ABR traffic were simply placed onto the network with no knowledge of the available bandwidth, congestion would occur and the network would then drop these cells. In order to prevent congestion, flow control is needed for ABR services. To implement flow control the source needs to be told what available bandwidth there is. A closed loop system with feedback can provide this. There are two flow control systems proposed for ABR, rate and credit based flow control. However the ATM forum has adopted rate based flow control.

The biggest problem in a control system is delay or phase lag. That is the flow control is based on old data that no longer reflects the network conditions. Latency on the round trip for ATM Resource Management (RM) cells [25] is the biggest problem. Latency is not a huge problem in the Fast Data Path, but can become a problem in the Interleaved Path.

The ADSL link is not likely to be the point of congestion. In most cases there is one user per link and very seldom will an ADSL link be connecting a LAN to the WAN. Congestion is more likely to occur in the WAN. Assuming the congestion is in the WAN, this proposes the solution that the RM cells do not travel the whole route to the client but only to the ATU-C at the DSLAM. Implementing ABR flow control in this way removes all flow control from the ADSL link.

This is feasible because the ATU-C knows the link rate, the traffic present on that link and the BER on that channel. Thus, it would be possible for the ATU-C to process the RM cell efficiently and without latency.

If the above were not implemented then the network may well operate at high cell loss ratio and oscillate. This would be aggravated in the Interleaved stream where the cell may take 30ms to traverse one way.

7.7.2 Unspecified Bit rate (UBR)

UBR differs from ABR in that it does not have a minimum data rate and it has no flow control. UBR sends data in bursts at the maximum line rate seen at that point.

This poses a totally different problem when used with ADSL. Latency is no longer the problem, but rather the ADSL physical link rate.

The ADSL line rate can vary between 1.5Mb/s and 8Mb/s but the line rates in the broadband network can be much higher. As a result there is major congestion at the egress point of the ADSL access network. The buffers in the DSLAM and ATU-C will quickly overflow and there will be a very high cell loss ratio.

A possible solution is to specify what the maximum link rate is on call setup is to the source. Currently there is no specification for this, but it has become an issue and there is now provision for this in UNI4.0. This will allow the source to limit the rate at which the UBR traffic is sent.

Chapter 8 Improving ATM Cell Delivery

Improving the delivery of complete ATM cells to the end application is the focus of this chapter. The problem is more than just an improvement to the cell throughput in the network, but rather this problem could be looked at from another angle: How to maximise the good uncorrupted data through the network? There is no point in passing data to the fixed network, if a higher layer is going to discard or drop that data if one of its cells is corrupted. In order to understand the problem presented it is necessary to look both at the cell protection offered in ATM and the higher AAL layers used in ATM.

The next section will offer a brief introduction to the protection offered to the ATM cell and then look at how data is encapsulated in the different AAL layers. Focus is also placed on the effect of high BER and the effect that bursts or impulse errors have on delivering this ATM data uncorrupted.

8.1 Cell Protection in ATM

ATM relies heavily on a good underlying channel i.e. a channel that has both a high bandwidth and a low BER. Good channel qualities allowed ATM cells to have very little protection. In the ATM cell there is a Header Error Control field (HEC). This is an 8-bit field that only corrects the header from single bit errors [11]. HEC configuration comes about again from the fibre network where most errors are single-bit errors. However it is not the case for ATM over ADSL, where the probability of multiple-bit bit errors occurring is high.

There is therefore a detrimental effect on ATM transport over ADSL. This may result in a higher probability that there will be misdirected cells due to a corrupted header and also in a loss in cell delineation. The latter can seriously affect the committed QoS offered to the end user. During cell delineation recovery, all the transported cells are dropped. This is a large problem with impulse noise.

8.2 AAL Layers

This section briefly outlines the AAL layers, specifically 1 and 5. This is presented so that those AALs can be used in a subsequent discussion on cell throughput. First AAL 1 is presented, then the popular AAL5 [15]

8.2.1 AAL 1

AAL 1 is designed for CBR services that require timing recovery as well. AAL 1 is the only other AAL that has seen large implementation and is presented here for this reason. Messages must be fed in at a constant bit rate by the application. If a cell is missing on arrival AAL 1 is able to signal this to the application.

Within the AAL layer there are two sublayers, the convergence sublayer (CS) and the segmentation and reassembly layer (SAR). The first part of AAL 1 layer is the convergence sublayer (CS), which is used to detect lost and misinserted incoming cells and also to smooth out incoming data [11].

The CS sublayer breaks up the message into 46 or 47 byte sections. The segmentation and reassembly (SAR) sublayer deals with these sections.

The SAR adds the header of 1 byte, comprising a 3-bit sequence number (SN), a 3-bit sequence number protection (SNP) and an even parity bit [25].

8.2.2 AAL 5

AAL 5 is probably the most used of all the AAL's to date and finds a place in many of the applications used in ATM. AAL 5 is also a newer AAL and replaces AAL 3/4 for many applications. The objective of AAL 5 is to offer a service with lower overhead and to have good error detection properties [25].

AAL 5's overhead is described briefly as follows. There is a 32-bit CRC checksum field for error checking in the entire message. The User to User (UU) field is 1 byte long which is not used and the Length Field is 2 bytes long and is used to indicate the length of the true payload and 1 byte Common Part Indicator (CPI) field that is not used to date. As a result AAL 5 has an 8-byte trailer and no header. The payload data is padded so that the entire AAL 5 message is a multiple of 48 bytes. The major problem here is that the payload may be anywhere between 0 to 65535 bytes, if any of these cells for a specific AAL message is dropped or corrupted the entire AAL 5 message is dropped.

8.3 Achievable ATM Cell Losses Over ADSL

In ADSL it is foreseen that ALL 5 will be the dominant AAL. The predominant traffic of the future will be ABR and variable bit-rate video for which AAL 5 cut down approach works well leaving specific functionality to higher application layers. VoD is considered as one of the major applications for ADSL. AAL5 is also suited to carrying CBR traffic for which AAL 1 was designed. The only problem is that the timing recovery will have to be built into a higher application layer. The reason that CBR traffic can be carried by AAL5 is that CBR is really a subset of VBR. For completeness AAL 1 is presented in the previous section and will also be considered here. AAL 5 is also used to carry LANE, CLIP and MPOA traffic and this together with VoD will cover most applications that would be offered to ADSL subscribers.

This section outlines the expected achievable cell throughput in an ADSL link, using the information gained in the above simulations. The first case considered is with an overall bit error rate set at the required rate. With a probability of a bit error set at 10^{-7}, then the probability of an errored cell is $0.5 * 10^{-4}$. This is extrapolated to an AAL 5 frame with full PDU. The resulting probability of errored AAL frame is 0.058.

At this point it is clear that the required delivery of ATM cells to the application is not available. On average the PDU in most networks for AAL 5 is approximately 2500 bytes. With the reduced PDU, the probability of corrupted AAL frame is $2*10^{-3}$. This improves the effective reliable throughput as now only 2.5 KB are discarded instead of 65 KB for a single corrupted cell. Thus at the application layer there is a greater throughput of data.

FEXT and AWGN do have an adverse affect on the ATM cell loss. The main reason for this is bit errors are normally distributed and occur with low probability of the order 10^{-12} once the 6dB-safety margin is included. Under these conditions the probability of an AAL 5 frame being discarded is much lower. Again considering the case of a PDU of 2500 bytes the probability is $2*10^{-8}$.

The effect of impulse noise is very destructive to ATM transport because of the distribution of the errors. The first case is where no interleaving is implemented. The BER during the impulse period approaches 0.5. Under this condition an entire ADSL frame is corrupted. This is detected by the CRC built into the ADSL super-frame. The only good side is that the errors are limited to one ADSL frame as the impulse duration is shorter than ADSL frame length. The mean impulse inter arrival time is approximately 15 seconds. Thus every 15 seconds there is complete cell delineation on every VC and all AAL 5 frames are dropped. As the number of VC's increase the number of ATM cells dropped increases linearly.

The second case is where interleaving is implemented but the interleaver depth is not long enough to offset the impulse effect. Here the BER in the ADSL frames corrupted can vary from 0.014 to 0.004. The effect changes from link to link as the ADSL frame size changes but working on an average of an ADSL frame of 1200 bits. The errors can now span 3 ADSL frames, increasing the above problem outlined above.

The third case is when the optimum combination of FEC and Interleaver for the specific link is found it is possible to improve performance dramatically. The number of impulses that cause ADSL frame errors decrease from 98% to approximately 1% that is only one in a hundred impulses will cause an error. However the bit error distribution remains constant and the effect is the same as in the first case. Yet the time between errors increases from 15 seconds to every 25 minutes. This is huge advantage as cell delineation every 15 seconds with complete AAL 5 frame discard for every VC is not acceptable.

8.4 Early Frame and Cell Discard

By transporting cells over ADSL links, there is now the possibility of implementing early frame and cell discard as well as ARQ to improve the throughput and efficiency of ADSL data links. As ADSL has a CRC on each of its super-frames it is possible to detect whether there is an error in that frame. Seeing that AAL 5 frames maybe far longer than an ADSL frame, why transport the entire AAL 5 frame if it is to be discarded? This is only made possible by the fact that in AAL 5 the last cell of a frame has its PTI bit set.

The matter of ARQ is more difficult as it would only be possible to request a new ADSL frame. The CRC will only indicate an error in the ADSL frame not in the cell corrupted. However the modem need only buffer a few complete ADSL frames and the ARQ could request this, based on a sequence number basis. This results in the problem that ATM requires cells to be delivered in order. Ultimately this would require resequencing, and introducing delay and jitter in the received cells. This would only be a problem for some traffic contracts. Best effort traffic the above scenario would be acceptable.

Chapter 9 Conclusions

As a result of the work and research carried out in the course of this thesis, the following conclusions can be drawn with respect to the future and functionality of ATM over ADSL.

ADSL offers a unique opportunity for Telcos to expand their services and market share in the near future and to be able to compete with companies offering satellite and cable Internet services. However, with this opportunity there is now a far greater demand for bandwidth and as a result the network providers will require far greater bandwidth in their core networks. To provide this increase a new broadband network is being deployed all over the world based on ATM technology.

This new core network is one of the fundamental reasons to consider ADSL and ATM hand in hand. The other reason is to be able to offer QoS and a transparency to higher layer protocols.

The simulation confirmed what was found in other work undertaken in this area: That in ADSL, FEXT was the major limiting factor, whereas in other DSL systems it was NEXT. In the simulation it is confirmed that the maximum rate was reached asymptotically as the transmitter power is increased. It was shown in section 6.1 that the data rate does not continue to improve with increased transmission power. This indicated that even though increased transmission power decreased the effect of impulse noise and AWGN, FEXT started to become the limiting factor over the data rate as the transmission power increased. In all loops the optimal transmission power was found to be 200mW.

When looking at the distribution of bit errors on an ADSL link in comparison to fibre links, which are predominantly used for ATM transport and offer low BER and low cell loss ratios it became clear that ADSL introduces problems for ATM transport. The distributions indicated that it is very likely that on an ADSL link there will be multiple errors, which is not the case in fibre links, which have 99.6% single bit errors. These multiple errors introduce problems for the ATM headers, as the HEC can only correct a single bit error. This is supported by data 6.2.2. This is the case with RC FEC. Figure 6-3 has indicated that with no FEC, 25% of the time you have single bit errors. Figure 6-4 shows the distribution of errors with FEC. The bit error will only start at the limit of the FEC, in the case used this starts at 8 bits per frame. Neither of these match the channel characteristics needed for ATM. Further more with FEC none of the frame can be used as the BER in that frame is likely to be 0.5 as the decoder cannot decode the corrupted frame.

When considering the ADSL link it is clear that impulse noise is the biggest remaining problem in the implementation of ATM over ADSL. Using interleaving, at the cost of increasing transmission delay can mitigate the effect of impulse noise. Increasing the number of parity bytes in the FEC can increase the data rate. However it is important not to go beyond the point where there is no gain in the effective data rate. Section 6.3 presents all the results for the affect of impulse noise on the BER. Figure 6-8 and Figure 6-9 highlight this. There is a significant drop in the BER and the number of error causing impulses for the correct interleaver depth. The interleaver depth needs adjusted based on FEC used. The smaller the FEC the greater the interleaver depth must be to achieve the same BER.

Channel efficiency was investigated at the physical level. The ratio of overhead to channel data is used. This is basically a function of the amount of FEC added to improve BER performance and to allow an increase in the data rate. However, seeing that in most cases a shortened RS code is used the efficiency drops off faster than the normal RS code. However there is an increase in speed that offsets the loss in channel efficiency. The biggest gain is not increasing the speed with the FEC but rather improve the BER for a given data rate. Table 6-1 shows that changing data rate as a function of the FEC overhead. Doubling the FEC from 4 to 8 correctable symbols only gives you a net data rate increase of 1.3%. In Figure 6-10 the channel efficiency of the Short and Norman code RS-FEC. The efficiency for the shortened code falls of significantly faster.

When considering the ADSL link as a cell stream instead of a bit stream, there is an increase in functionality, with regards to ease of multiplexing and there is no added overhead other than the standard ATM and AAL layer data.. However there is an increase in functionality needed in the modem to implement transmission convergence sub-layer functions needed for ATM transport. This is offset, as there is no need to implement all the ASx and LSx sub-channels present in the standard hardware. As a result using ATM eliminates the need to implement channel multiplexing in the ADSL hardware. This greatly simplifies the hardware implementation.

Using ATM over ADSL introduces two problems one for ABR and another for UBR traffic. Both these are solvable but require changes from currently employed methods. The biggest problem occurs for UBR where congestion at the ADSL link results in high cell loss ratio. There is no solution until the UNI4.0 specification is fully available and addresses mechanisms to limit the source rate for UBR not to exceed the ADSL link rate.

The big problem with ABR flow control is latency. Latency in the ADSL link causes poor performance as the ABR flow control is using old network information. This control problem is solvable but does push up the cost of the ATU-C to provide the functionality in this component to process ATM RM-Cells.

Finally ADSL will allow the process of taking the new broadband services and fibre technology closer to the home. This will be the first and most important step towards residential connectivity to the super highway.

Chapter 10 Recommendations for Future Research

ADSL is a good solution to the local loop bandwidth problems of today. However DSL technology has progressed in the last year and the prospects for VDSL are now stronger. Based on this possibility it would be beneficial to extend this study to incorporate VDSL technology. This however is not a large problem, as the architecture of the VDSL network does not differ largely from that of ADSL. In Addition VDSL uses a greater section of the spectrum and is not limited to the 1.2 MHz of bandwidth as in ADSL. This will require a check on the accuracy of the channel and noise models at higher frequency. VDSL will also use DMT modulation but with a larger number of sub-channels.

With the onset of VDSL in the network, the modelling of NEXT will become more of a factor as opposed to ADSL where FEXT is the limiting factor.

There has also been a recent development in DMT modulation using a different transform. A Wavelet Transform can replace the IFFT currently used. Thus Discrete Wavelet Multitone (DWMT) replaces DMT. DWMT is Aware's proprietary implementation [46] [45]. DWMT has advantages over DMT and may be used in the future with VDSL. The reason for this change is that it will reduce the lobe width for the sub-channels and hence reduce the inter-sub-channel interference. To implement DWMT in this simulation would require that the code, which implements the modulator, be changed to reflect the new transform.

Now that there is an understanding of what ADSL networks can offer using ATM as a transport protocol end to end a cross the network it would be fruitful to use this information when designing VoD application for the residential market. Also to study the effect ADSL will have on the core network as a result of increased traffic and to model the characteristics of this traffic once there are experimental networks in place. Traffic generated by ADSL networks may follow the same usage pattern as dialup Internet traffic, but with a larger mean value. However with the onset of

VDSL there will be a greater level of bi-directional traffic. The focus can then move away from this asymmetric bandwidth usage.

The development of the impulse noise model is based on data presented in the standards and past research to the effect impulse noise has on older digital systems. Based on this and with the co-operation of the Telcos, more measurements could be taken in the local loop to verify or improve the models implemented.

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Appendix A

Discrete Multitone (DMT) vs. Carrierless Amplitude/Phase (CAP) Modulation

This appendix will outline the differences and the advantages of DMT and CAP with relation to ADSL and the services which an ADSL access network should offer. This section is included to help the reader understand why DMT modulation is implemented in this simulation.

Probably the first and most important reason why DMT has been developed in comparison to CAP is that it is now standardised for ADSL by ANSI [43], ETSI [51] and the ITU-T [49][50]. The comparison between DMT and CAP will be described below highlighting the modulation scheme, interoperability, noise immunity, power consumption, error correction and rate adaptation.

Description of the Modulation

CAP is closely related to Quadrature Amplitude Modulation (QAM). However the orthogonal signals are generated using different techniques. CAP can be seen as a digital implementation of QAM, which uses two digital bandpass filters (Hilbert Pair) the signals are then added and fed into a DAC.

DMT on the other hand, uses what could be viewed as QAM in parallel. Here the spectrum is divided into small slots. Each one of these slots has its own carrier, which transmits a fraction of the information. An efficient way to implement this is to use a Fast Fourier Transform (FFT). This transform preserves the orthogonality between each sub-channel. This technique is very computationally intensive and has been around for some time now. Only as a result of the increased speed in DSP has it become a viable solution.

Interoperability

Interoperability has never been demonstrated between different manufactures of CAP modems. The main reason for this is that there is no standard for CAP as yet. In comparison DMT complies with the standards and has demonstrated good interoperability between manufactures.

Noise Immunity

A single carrier approach works well in a channel that is essentially "clean". All the frequencies are received the way they were transmitted. Here the channel is linear and attenuation is related to the propagated distance. However the copper loop does not have these clean properties, and this can be seen as the main downfall for CAP systems. In the local loop the channel has greater attenuation at higher frequencies and the physical affects of bridge taps create dips in the frequency response. Thus, a technique like DMT, which adapts its signal to match the channel characteristics, will achieve better results.

In addition to normal background white noise (AWGN), the local loop has impulse noise and RF interference. Inherently DMT is more robust than CAP to Impulse noise as the symbol in CAP is much shorter in comparison to the DMT symbol.

When considering RF interference from AM radio, DMT is in a good position to offset this affect because it simply reduces or places no information in the affected sub-channels. CAP is not capable of avoiding the RF interference like DMT and as a result suffers in performance.

Power Consumption and PSD

DMT modems use less power in comparison to CAP. Also the PSD of DMT is lower resulting in less cross talk coupling. Looking more closely at this DMT has transmitted power of -40dBm/Hz and CAP has transmitted Power of -34dBm/Hz. CAP has higher power consumption when compared to DMT.

A PSD mask is defined in the standards. PSD is limited ensure compatibility with other systems, by limiting the interference with other systems. However CAP implementations do not comply with this PSD mask. To represent this effect CAP cannot be deployed in the same cable as DMT. The upstream spectrum of CAP continues to 180kHz when it should stop at 140kHz and Down Stream as specified for DMT implementation

Ultimately CAP is less efficient than DMT, requiring greater bandwidth and higher signal power to obtain similar data rates. As a result of this the CAP's crosstalk renders it completely incompatible with a large number of other solutions.

Error Correction

CAP only implements FEC on the downstream direction. This complicates modem design and interoperability as the FEC is to be built in on a higher layer in the modem, and not built into a chipset solution. DMT uses error correction as a powerful technique to achieve good performance. CAP does not utilise this and as a result suffers with poor performance and interoperability issues.

Rate Adaptation

DMT is fundamentally a rate adaptive modulation scheme. In comparison CAP was inherently a fixed rate. However it has undergone advancements to achieve rate adaptation.

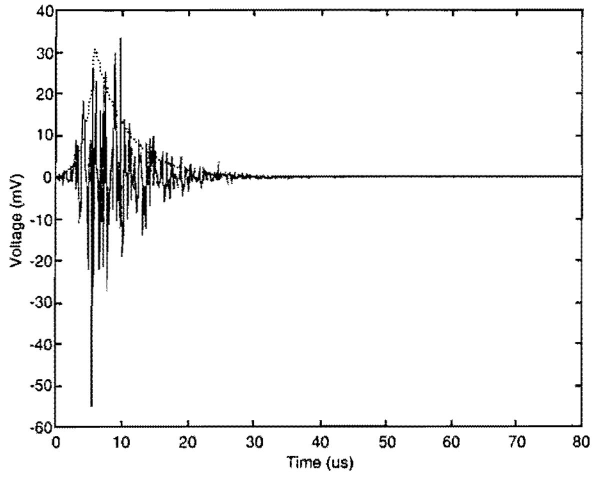
DMT has a greater degree of freedom, when compared to CAP. DMT is thus able to alter the data rate on each of the sub-channels independently. CAP is only able to alter one constellation and the bandwidth on a single carrier. DMT can step in increments of 32kbps steps, from 64kbps too greater than 8Mbps depending on line conditions. CAP can only step in 640kbps steps, steps that are 20 times greater.

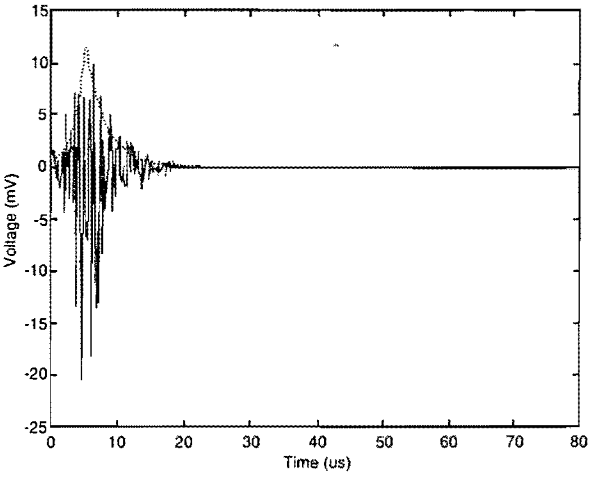
Conclusions

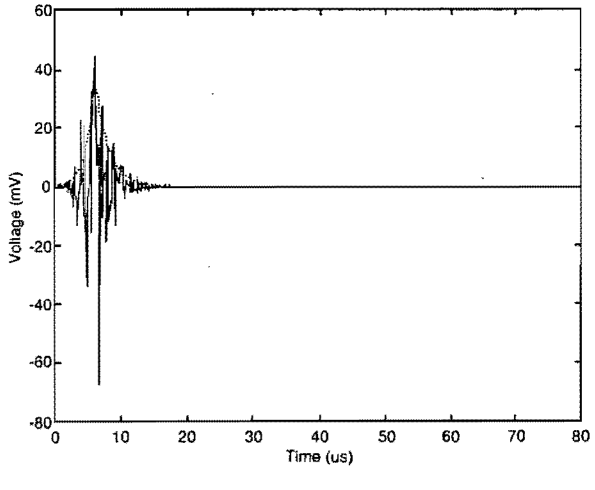
DMT is well-refined technology that can efficiently match transmition to the given channel. DMT can use the bandwidth more efficiently. DMT is more power efficient than CAP. DMT is rate adaptive. DMT has built in error correction. However DMT does suffer from latency, especially aggravated in the interleaved data stream. By supporting two physical channels an interleaved stream and a fast data stream, DMT can offer service with different latency. This feature, together with its rate adaptation, makes it well suited to transporting ATM traffic. VC requiring a specific QoS can be assigned to the correct physical channel offering the required physical characteristics needed for the traffic contract.

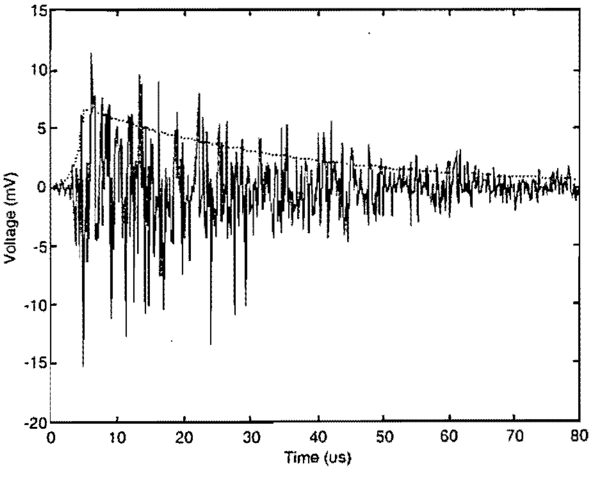
Appendix B

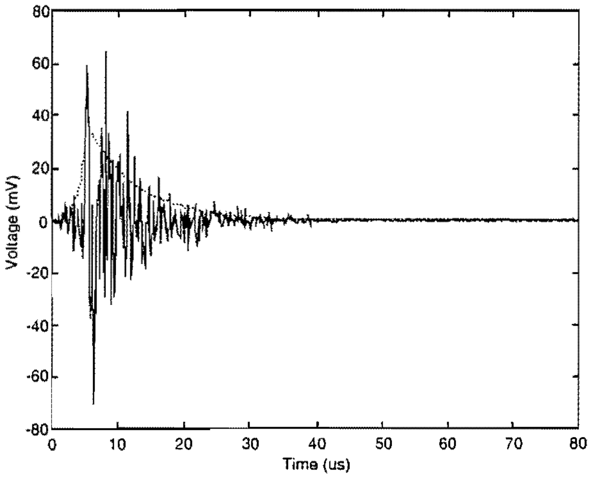
Appendix B illustrates samples of impulse noise generated from the impulse noise model used in the simulation. These represent the instantaneous noise voltage versus time as the impulse is generated. The envelope used to generate this impulse is overlaid



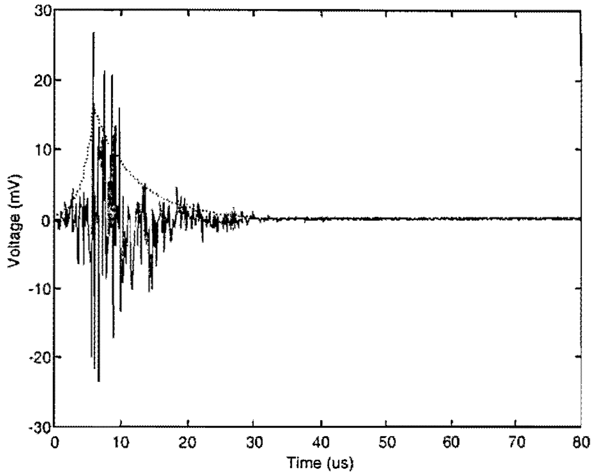








Voltage (mV)
Time (us)



Voltage (mV)
Time (us)

Appendix C

This section outlines theory used in the bit-loading algorithm. This is based on the analysis of each sub-channel as an independently modulated QAM channel. The work presented here is from a paper by Prof. Cioffi [1] A Multicarrier Primer" from Stanford University. It is included as an appendix here for completeness to help the reader to understand the derivation of the bit-loading algorithm used in the simulation.

Single carrier analysis is extended to the multicarrier system, as it is equivalent to a group of independent QAM sub-channels. In QAM modulation there is implementations with varying constellation sizes. These are usually in sizes of power of 2. This however is not the case in DMT, which uses constellation sizes between 0 and 15 bits per symbol. The important information needed is the average symbol energy for the constellation.

The average symbol energy is defined by the following equation. This is for constellations centred on the origin. This applies for even values of bits per symbol. However it is a good approximation for other constellation sizes.

$$\varepsilon = \left[\frac{2^b - 1}{6}\right] d^2 \qquad \text{C.1}$$

Here d is the Euclidean distance to the nearest other point.

If the channel gain is |H| then it is possible to define the probability of a symbol error is defined by the following equation for QAM.

$$P_e \le 4Q\left[\frac{d_{\min}}{2\sigma}\right] \qquad \text{C.2}$$

$$d^2{}_{\min} = d^2 \, |H|^2$$

For a probability of a symbol error per dimension (Pe/2) of 10^{-7} which is the case in ADSL we require the following.

$$\left[\frac{d_{\min}}{2\sigma}\right]^2 = 14.5dB + \gamma_m dB - \gamma_c dB$$
$$\gamma_m = System\ Margin \qquad \text{C.3}$$
$$\gamma_c = Coding\ Gain$$

The margin is the amount of extra performance that is needed to ensure the required performance when there are unforeseen channel impairments. If coding is present in the system then the amount is reduced by the coding gain. The typical margin used in DSL environment is 6 dB.

Now the SNR gap Γ is defined with the following equation, where σ^2 is the noise variance.

$$3\Gamma = \frac{d_{\min}^2}{4\sigma^2} \qquad \text{C.4}$$

Substituting 3 in 4 results in following result.

$$(3\Gamma)(dB) = 14.5 + \gamma_m - \gamma_c \qquad \text{C.5}$$

One can rewrite (1) as follows.

$$2^b = 1 + \frac{6\varepsilon |H|^2}{d_{\min}^2} \qquad \text{C.6}$$

Take the log of (6) and substitute (6) results in the number of bits per symbol b being defined by.

$$b = \log_2\left(1 + \frac{SNR}{\Gamma}\right)$$
$$SNR = \frac{\varepsilon\,|H|^2}{2\sigma^2} \qquad \text{C.7}$$

From this it can be seen that the SNR gap Γ is the measure of the loss with respect to the theoretical optimum performance. In practice b would be rounded to obtain the required data rate for the system.

Appendix D

Constellation diagrams used in the ADSL simulation. A more in depth description can be found in [4]. They are used to map the incoming bit stream to a complex value, which in turn is used in the DMT modulator. The mapping is based on the number of bits per symbol allocated to the relevant sub channel.

Basically, there are two forms of the constellations' even number of bits per symbol and odd values. However both have a square shape. These constellations are shown below. There is support for 2 to 15 bits per symbol. As defined by the ADSL forum and ANSI T1.413 specification [43].

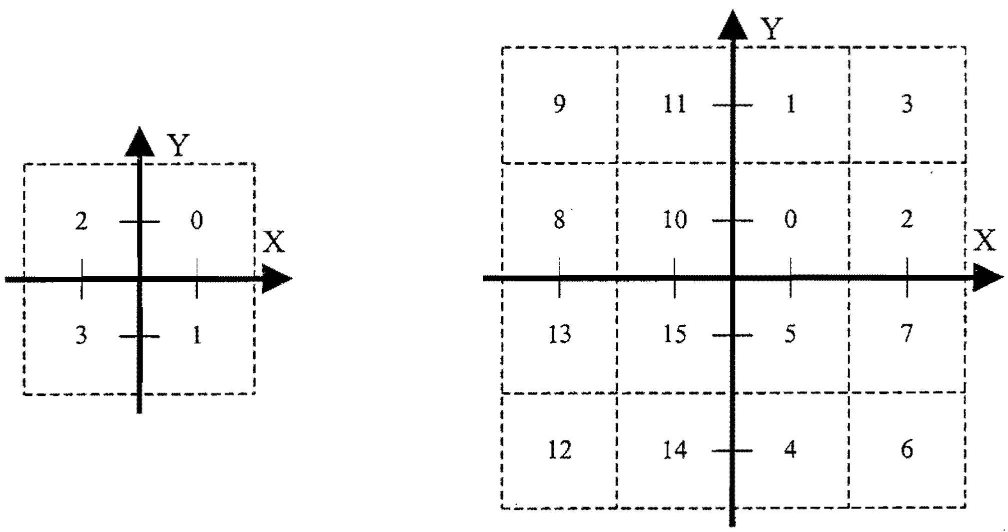


Figure D-0-1 Constellation Diagrams for 2 and 4 bits per Symbol

All possible even constellations can be generated from b = 2 and all the odd value constellations from b = 5 using the following recursive replacement.

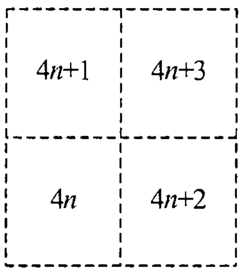


Figure D-0-2 Expansion Replacement for Constellation Diagrams

For b = 3 however there is no generating form and therefore is defined as follows.

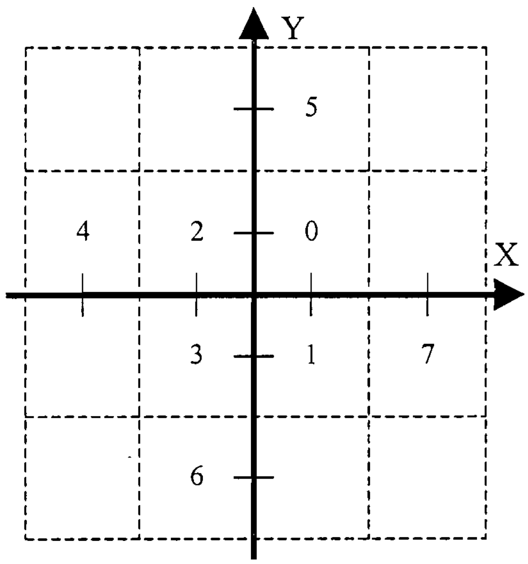


Figure D-0-3 Constellation Diagram for 3 bits per Symbol

The Constellation for 5 bits per symbol is shown below. The entire remaining odd constellations are generated from this constellation.

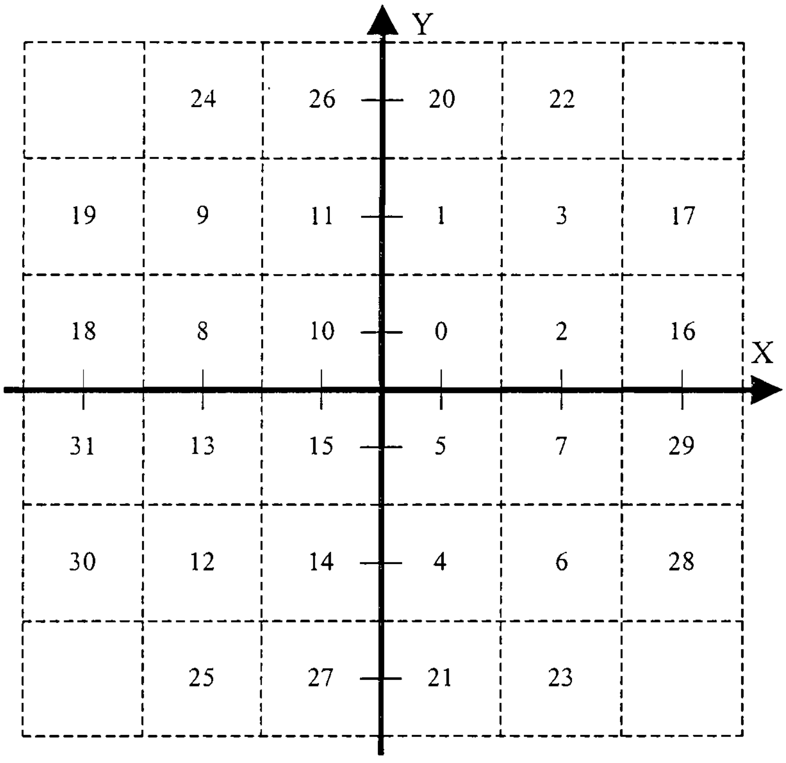


Figure D-0-4 Constellation Diagram for 5 bits per Symbol

Appendix E

This appendix outlines how the local loop line is defined for the purpose of simulation to obtain its frequency and phase response using the Line Mod program. The loop expressed here corresponds to loop 5 in Figure 3-1.

The elements of a transmission line are modelled using the ABCD parameter matrix model of a two-port. The user can add a new circuit element by simply providing a subroutine, which supplies the ABCD matrix. The input to LINEMOD is a file containing the specification of the two-ports

MEXP and NEXP are constants that are defined for the programme in the input file. T is a floating number, which controls the time axis. One usually thinks of T as the time for one bit in a data transmission system. From the following equation it is possible to calculate MEXP and NEXP.

Total time that the time response is calculated over is = T*2^MEXP.

The spacing of points in the time response is = T/(2^NEXP).

Thus MEXP should be chosen large enough to encompass the whole impulse response, and should be chosen to be larger, the narrower the bandwidth of the channel.

```
# Line 5
# MEXP NEXP T
5        4        7.24637681e-6

Series_r 100.

# First segment
Line 1.0 26
```

```
        #Bridge
        Bridge
        Line 0.5 26
        End
# 2nd segment
Line 1.5 26
        #Bridge
        Bridge
        Line 0.5 26
        End
# First segment
Line 0.3 26

Shunt_r 100.
End
```

Appendix F

This appendix outlines the inserted CD, which contains this document in electronic form and the code developed for the simulations and the third party software used.

The program code for this thesis has been developed using both C++ and MATLAB. All the simulation code has been specifically developed for a Linux based platform, as the cross compiler obtained for the MATLAB code was for Linux. However this is available for other platforms such as Win NT. The C++ Compiler is GCC for Linux and is distributed with all Unix systems.

The cross compiler is from MATCOM and a fully functional evaluation version is available on the Internet and is included on the CD. In order to use the compiler a licence number has to be downloaded from.
http://www.mathtools.com/matcom4.html

The local loop characteristics where modelled using a program LINEMOD that is also included on the CD and its works with any C++ compiler.

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