NCS TIB 99-1

NATIONAL COMMUNICATIONS SYSTEM

TECHNICAL INFORMATION BULLETIN 99-1

ASYNCHRONOUS TRANSFER MODE (ATM) OVER ASYMMETRIC DIGITAL SUBSCRIBER LINE (ADSL) SYSTEMS

JANUARY 1999

OFFICE OF THE MANAGER
NATIONAL COMMUNICATIONS SYSTEM
701 SOUTH COURT HOUSE ROAD
ARLINGTON, VA 22204-2198

NCS TECHNICAL INFORMATION BULLETIN 99-1

ASYNCHRONOUS TRANSFER MODE (ATM) OVER ASYMMETRIC DIGITAL SUBSCRIBER LINE (ADSL) SYSTEMS

JANUARY 1999

PROJECT OFFICER:

DALE BARR, Jr.
Senior Electronics Engineer
Technology and Standards Division

APPROVED FOR PUBLICATION:

PETER FONASH
Chief, Technology
and Standards Division

FOREWORD

Among the responsibilities assigned to the Office of the Manager, National Communications System (NCS), is the management of the Federal Telecommunications Standards Program. Under this program, the NCS, with the assistance of the Federal Telecommunications Standards Committee, identifies, develops, and coordinates proposed Federal Standards which either contribute to the interoperability of functionally similar Federal telecommunications systems or to the achievement of a compatible and efficient interface between computer and telecommunications systems. In developing and coordinating these standards, a considerable amount of effort is expended in initiating and pursuing joint standards development efforts with appropriate technical committees of the International Organization for Standardization, the International Telecommunications Union-Telecommunications Standardization Sector, and the American National Standards Institute. This Technical Information Bulletin presents an overview of an effort which is contributing to the development of compatible Federal and national standards in the area of Asynchronous Transfer Mode Standardization. It has been prepared to inform interested Federal and industry activities. Any comments, inputs or statements of requirements which could assist in the advancement of this work are welcome and should be addressed to:

Office of the Manager
National Communications System
Attn: N6
701 S. Court House Road
Arlington, VA 22204-2198

This document was prepared under contract to the

Office of the Manager
National Communications System

Contract #DCA100-95-C-0126

by

SETA Corporation
6862 Elm Street
McLean, VA 22101

(703) 821-8178
Fax: (703) 821-8274

ABSTRACT

Asymmetric Digital Subscriber Line (ADSL), a new modem technology currently being deployed, promises to provide enhanced and affordable access to the Internet, live video, and a wide variety of other multimedia broadband services over existing copper twisted-pair wirelines. Because of its capability to accommodate a broad range of traffic from diverse sources, support high data rates, and provide gains in bandwidth efficiency, Asynchronous Transfer Mode (ATM) is gaining broad acceptance as the preferred transport mode for broadband terrestrial communications. The key to successful transport of ATM traffic over ADSL is an end-to-end architecture that with protocol transparency ensures preservation of both the rate adaptive characteristics of the ADSL access medium, and the quality of service (QOS) guarantees of ATM. The ability of ATM over ADSL to provide seamless broadband access for remote users appears to have significant implications for planners and providers of national security and emergency preparedness (NS/EP) services. This report presents the results of an examination from a NS/EP perspective of selected technical interface/system considerations associated with the use of ADSL technology to provide access to the public switched telephone network (PSTN) for ATM traffic.

TABLE OF CONTENTS

LIST OF FIGURES

LIST OF TABLES

EXECUTIVE SUMMARY

PURPOSE

This report presents the results of an examination from a national security and emergency preparedness (NS/EP) perspective of selected technical interface/system considerations associated with the use of Asymmetric Digital Subscriber Line (ADSL) technology to provide access to the public switched telephone network (PSTN) for Asynchronous Transfer Mode (ATM) traffic.

BACKGROUND

The rapid growth of the Internet and the World Wide Web (WWW), and the widespread deployment of personal computers have greatly accelerated the need for higher bit rates, faster local loop connections to the PSTN, and more responsive access to bandwidth on demand. While the picture is not altogether clear at this time, it is certain that new capabilities which are resulting from the current deployments will have significant implications for planners and providers of NS/EP services——e.g., the ability to provide real-time video from a disaster site for remote damage assessment or medical diagnosis and treatment. Recent advances in digital signal processing (DSP) have resulted in significant increases in usable bandwidth over traditional wirelines used for PSTN access. However, in many respects, the universal transmission of data on such lines still has significant limitations. Specific limitations concern achievable data rates, the quality of transmissions, and the availability of shared access.

The ongoing deployment of Integrated Services Digital Network (ISDN) capabilities allows the transmission of voice and data over a single high-speed connection. However, ISDN is still basically a dialup, circuit-switched technology which tends to waste bandwidth on access wirelines used for Internet connections by utilizing the entire bandwidth even when no data is flowing over the connection. ISDN service is generally more costly than regular analog telephone lines, and although widely deployed, is still not yet universally available—especially in remote areas. ADSL, a near-term alternative to ISDN, is a modem technology which uses advanced DSP, an analog coding algorithm, and existing copper twisted-pair telephone wirelines to move digital information between two points. It is specifically designed to exploit the one-way nature of most multimedia communications where large quantities of information flow toward the user and only a small quantity of interactive control information is returned. ATM is generally accepted as the leading transport mode for broadband terrestrial communications. Because of its flexible architecture, NS/EP Internet users and service providers are expected to be significant users of ATM technology. Consequently, it appears that issues with potential impact on the transport of ATM over access networks based on ADSL should be an area of special interest to the NS/EP community.

ADSL TECHNOLOGY OVERVIEW

Depending upon line length and loop conditions, the copper wirelines used in the local telephone loop to carry voice traffic between subscriber locations and the telephone company's central office are also capable of carrying megabits of data. However, since in telephone conversations 64 kilobits per second (Kbps) are sufficient to accurately reproduce the human voice, voice traffic only utilizes four kilohertz (kHz) of the available bandwidth. Theoretically, with the proper modulation scheme and coding, the unused bandwidth could be used to support high-speed digital connections to the PSTN for Internet access and other interactive multimedia applications. Because ADSL uses the same copper wirelines as normal telephone service on a non-interference basis, subscribers are able to continue using their existing telephones while also maintaining digital connections. The ADSL modem simultaneously creates a high-speed downstream——from the network to the subscriber——digital simplex channel, a medium-speed digital duplex channel, a baseband analog duplex or plain old telephone service (POTS) channel, and an ADSL overhead channel (AOC) for framing, error control, operations and maintenance. Full-duplex connections permit simultaneous communications in both directions. Simplex connections permit one-way service from the network to the customer without the capability to change directions. All sub-channel rates are programmable in any combination of multiples of 32 Kbps. The maximum net data rate depends on the characteristics of the loop on which the system is deployed. The high-speed simplex downstream channel transmission rate ranges from 1.5 megabits per second (Mbps) to 6.1 Mbps, while the transmission rate for the medium-speed duplex channel ranges from 16 Kbps to 640 Kbps. Historically, services beyond POTS have been "asymmetric" in nature. That is more information is transferred in one direction than the other. ADSL is "asymmetric" in that its two-way bandwidth is not evenly divided between the upstream and downstream directions. This asymmetric nature of the divided bandwidth makes ADSL especially suited for Internet and selected multimedia-intensive WWW and client/server applications that are mostly transmitting data downstream. Upstream requests and responses from users to the network are small in comparison and require little upstream bandwidth.

ADSL Modem

The ADSL modem is the key element which makes ADSL operations possible. In a typical ADSL installation, a single twisted pair of telephone wires is used to connect the ADSL Transceiver Unit-Remote (ATU-R) at the subscriber's location to the ATU-Central (ATU-C) at the telephone company central office. The installed modems create a high-bandwidth downstream channel, a smaller upstream channel and a basic POTS channel for voice. The POTS channel is split off from the digital modem by filters for routing of the POTS signal in the circuit-switched network, thus guaranteeing uninterrupted POTS in the event of modem failure. The data signals are routed into the telephone company's high-speed packet-switched network. Simultaneously, ADSL conveys the channels and line overhead for framing, error control, operations, and maintenance.

Multiple Channel Techniques

To create multiple channels, ADSL modems divide the available bandwidth of the wireline by use of one of two techniques: frequency division multiplexing (FDM) or echo cancellation. FDM assigns one band for upstream data and another band for downstream data. Echo cancellation assigns the upstream band to overlap the downstream band and separates the two by means of adaptively forming a replica of the echo signal arriving at a receiver from its local transmitter and subtracting it from the signal at the input to the receiver. Echo cancellation uses bandwidth more efficiently, but is more complex and costly to implement. With either technique, ADSL splits off a 4 kHz region for POTS at the lower end of the band. Because voice and data traffic are transported on logically separate paths, the data can be routed around the local switch, thus providing load relief without massive hardware and software upgrades.

ADSL Modulation

The discrete multi-tone (DMT) modulation block is the heart of the ADSL modem transmitter. Before entering the DMT modulation block, the payload to be transmitted consists of a buffer of data from the fast channel and a buffer of data from the interleaved channel when both channels are implemented. Modulation converts input bits into waveforms to be sent over the channel. Demodulation in the receiver maps received waveforms back to the bits that generated the waveform—that is it recreates the bits at the demodulator output that were present at the modulator input. With digital systems, the number of bits or hertz that can be supported with an encoding scheme is an important consideration. Encoding is the generation of character signals to represent quantized samples. A frequently heard complaint of digital systems is that they can be wasteful of bandwidth when compared with an analog system. For example, an analog voice channel is nominally 4 kHz, whereas a digital voice channel is 64 kHz——assuming one bit per hertz of bandwidth——a 16-to-1 difference in the required bandwidth. Because of the perceived wastefulness, techniques that involve the use of more efficient line codes to permit shared use of loop bandwidth for both analog and digital data are receiving increased emphasis. ADSL line coding involves encoding a digital signal into an analog signal that represents the original signal for transmission over an analog channel. Since the PSTN loop is both bandwidth limited (kHz) and power limited, line coding schemes generally focus on manipulating the analog carrier signal.

An analog signal has at least three attributes with relevance to modulation: amplitude, frequency, and phase. Modulation involves manipulating one or more of these attributes to increase the number of bits/hertz in a transmission line. Vendor efforts to date have resulted in two main competing line codes or signal modulation algorithms to conserve bandwidth in ADSL implementations: Carrierless Amplitude/Phase (CAP) modulation and DMT modulation. The first modulation algorithm, CAP, is well understood and chipsets for the process are readily available. However, CAP is only offered by AT&T Paradyne at this time. DMT, the second modulation algorithm used in ADSL implementations has been ratified by ANSI as the coding standard for ADSL. DMT is considerably more complex and costly than CAP. Despite ratification as the coding standard, modems utilizing both CAP and DMT are still being tested to further examine employment costs and performance attributes. Based on the results of testing to date, it appears that both encoding algorithms have so far elicited a positive response for performance.

ATM OVER ADSL: TRAFFIC MANAGEMENT CONSIDERATIONS

ATM——a broadband, low delay, packet-like switching and multiplexing technique——segments voice, data, multimedia, and video into fixed-size cells for transparent transmission. It is asynchronous in that the recurrence of cells from an individual user is not necessarily periodic. ATM traffic patterns are generally unpredictable. While a great deal is understood about the pattern of regular voice traffic, the pattern for bursty traffic such as video, data, or compressed voice, is less well understood. Recent research studies indicate that such traffic will not ”smooth out” over an extended period of time—smooth traffic behaves like random traffic that has been filtered. For normal voice traffic, the filtering or limiting of peakiness is accomplished by call blockage during the busy hour. For bursty traffic, there will always be some possibility that shared link bandwidth will be temporarily exceeded and cells will have to be dropped. To preclude this possibility requires considerable buffer support at every contention point where ATM terminals compete for access to the supporting telecommunications medium. Efficient use of the supporting telecommunications media requires high utilization——operating at or near capacity. Contention——a condition arising when two or more data stations attempt to transmit at the same time using a shared channel——impedes high utilization and degrades quality of service (QoS). The challenge for the supporting telecommunications medium is to maximize utilization without negatively impacting QoS. The key to meeting this challenge is traffic management.

The principal role of traffic management is to protect the network and the end-system from congestion. With ATM over ADSL, several mechanisms are used in appropriate combinations to monitor and control the flow of traffic and congestion within the ADSL local loop. Because ATM is connection-oriented, the transport of ATM cells over ADSL requires the establishment of an end-to-end virtual channel/circuit (VC) prior to the transmission of user data. A VC is a pre-established communications channel. ATM transport over ADSL is supported by mapping the ATM data stream into predefined ADSL sub-channels. Data applications using ATM over ADSL access loops will require low cell loss rates to avoid throughput collapse due to packet retransmission. Since many data applications cannot predict their own traffic parameters, it is difficult for a call-level traffic control scheme to optimize both low cell loss and the efficient use of supporting telecommunications media resources.

With ADSL systems as with any access network where two or more users share a common medium on a dynamic basis, the risk of collision is always present. The arrival pattern for ATM cells at a particular channel is irregular. However, with statistical multiplexing, cells are multiplexed according to when they arrive. This provides traffic sources with a great deal of flexibility since they are now free to generate cells as necessary and do not need to generate data at a constant bit rate. Because of the sophisticated cell scheduling provided by statistical multiplexing, the potential bandwidth starvation problem associated with the monopolization of transmission bandwidth by higher priority traffic is avoided. With respect to traffic management, only the physical medium dependent (PMD) and the transmission convergence (TC) sublayers of the physical layer, and the ATM layer of the Broadband-ISDN (B-ISDN)/ATM reference model are relevant. However, because of the impact certain ADSL network features have on the ability of the supporting ADSL network to fulfill ATM QoS guarantees, selected network features must receive special consideration. Specific features include: (1) channel processing capabilities, (2) the adequacy of forward error correction (FEC) capabilities, and (3) the ability of the ADSL system to perform rate adaption.

CHANNEL PROCESSING CAPABILITIES

According to the draft ANSI standard [3], the processing of ATM data for transport over ADSL can roughly be divided into three logical functions performed by the ADSL modem: (1) a channel processor function, (2) a TC sublayer function, and (3) a PMD function. In examining the three logical functions described above, there appears to be some duplication in the functions of the ADSL system modem and two of the three layers of the B-ISDN/ATM reference model. It appears that this duplication could result in reduced overall throughput because of an unnecessary increase in overhead and implementation complexity. Under certain scenarios, it appears that the potential reduction in transmission capacity available for bearer payloads could have significant impact on real-time, high data rate NS/EP applications like real-time video——especially during periods of heavy traffic or degraded communications conditions.

FORWARD ERROR CORRECTION

Even with the combined forward error correction (FEC) cell header and transmission oriented capabilities inherent in ATM over ADSL, it appears that FEC may not be sufficiently effective in certain environments because of the length of the continuous blocks of ATM data transmitted over ADSL. FEC codes can typically correct only eight bytes in a transmitted data block. Data blocks containing more than eight bytes in error are generally discarded. To reduce the possibility of cell discard, ADSL modems adopt an interleaving process to reorder the byte stream so that bursts of errors are distributed evenly among FEC code words. Interleaving spreads the effect of impulse noise interference over many widely separated bits with the aim of reducing the number of byte errors in a data block to a number correctable by the FEC code. However, interleaving adds delay to the end-to-end transport of data and also requires memory buffers at both the transmitter and receiver. Consequently, re-ordering the byte stream implies buffering many code words, and the introduction of delay proportional to the depth of the interleaver. For high-speed ATM cell transport, the process of interleaving may result in intermittent congestion at the ADSL buffer. In such instances, ATM cells may be discarded due to overloading of the buffer, or late arrival of the cells at the buffer because of excessive cell delay variation (CDV). As a consequence of congestion, interleaving itself could initiate buffer overflows resulting in the loss, discard or delay of cells.

RATE ADAPTION

The wide variations in characteristics of existing copper wirelines have the potential to allow non-linearities in copper-based ADSL networks. Such non-linearities frequently result in greater attenuation at higher frequencies and other impairments introduced by equipment or line conditions. To compensate for non-linearities in the local loop, each ADSL modem receiver must synchronize to the other's transmit signal during the initialization sequence. This enables the receiver to then calculate the distortion effects of the channel and compensate accordingly. Based on the calculations, a determination is then made of the loop's ability to support transmission of the requested configuration's aggregate data rate across the given loop. Rate adaption refers to the ADSL modem's capability to continually monitor the condition of the line and change its data rate in response to changing line conditions. The higher the signal-to-noise ratio (SNR), the higher the data rate that can be accommodated with a given QoS. Successful transmission of ATM over ADSL requires a reliable mechanism for adapting the data flow rate to the transmission medium at start-up, upon initialization following line re-connection, and dynamically with significant changes in line conditions.

Modem rate adaption can be static or dynamic. Typically, with static rate adaption SNR is measured at startup and a predetermined margin is subtracted to allow for possible additional noise or signal attenuation. Based on the adjusted SNR——with the margin subtracted——a maximum reliable data rate is then selected. The higher the margin, the lower the data rate, but the more adaptable the system is to changes in line conditions. Dynamic rate adaptation refers to a change in data rate during a steady state while user data is being transmitted. With dynamic rate adaption, the modem continually monitors the SNR. If the margin falls, the data rate is dynamically reduced until the margin returns to its original value. Correspondingly, if the margin increases, the data rate could likewise be increased to bring the margin back to the original level. Dynamic rate adaption includes not only rate modification, but also rate reapportioning between the fast and interleaved paths of the ADSL modem. Reapportioning could result in the corruption of several symbols, and purging the interleaver for the interleaved paths. Purging results in a loss of bits equal to the interleaver depth, and therefore an effective interruption of service duration equal to the ratio of the interleaved depth (D) to the bit rate (R). The maximum duration of this interruption is yet to be determined by the ANSI T1E1.4 Committee and the ATM Forum, but it is estimated to be 100 or 125 milliseconds (ms). In the fast path some of the symbols are also corrupted. The exact handling of this corrupted data during these periods prior to swap, has not been specified by the ADSL Forum. However, the rate adaptation process at the ADSL modem may impact the QoS of ATM users, especially for service contracts that are time sensitive or require low error rate. One approach being considered is to fill the interleaver and the deinterleaver with idle cells and transmit idle cells in those ADSL frames which may be corrupted.

Dynamic rate adaptation seems the rate adaptation mode of most interest to NS/EP users of critical applications requiring constant bit rate service——e.g., video or data applications with rigorous timing control and performance parameters. Dynamic rate adaption is not currently supported by the T1.413 standard, but is being discussed for future revisions. In the interim, to ensure uninterrupted service for high priority traffic during emergencies, NS/EP users of ATM over ADSL might consider negotiating a significantly higher than normal margin value with a corresponding reduction in data rate thus precluding the need for a rate change.

SECTION 1.0

INTRODUCTION

1.1 PURPOSE

This report presents the results of an examination from a national security and emergency preparedness (NS/EP) perspective of selected technical interface/system considerations associated with the use of Asymmetric Digital Subscriber Line (ADSL) technology to provide access to the public switched telephone network (PSTN) for Asynchronous Transfer Mode (ATM) traffic.

1.2 SCOPE

This report contains (1) a brief overview of ADSL technology, and (2) an examination from an NS/EP perspective of some of the key technical interface/system considerations related to the use of ADSL technology to transport ATM traffic in the local loop.

1.3 BACKGROUND

New capabilities provided by the rapid growth of the Internet and the World Wide Web (WWW), and the widespread deployment of personal computers have greatly accelerated the need for higher bit rates, faster local loop connections to the PSTN, and more responsive access to bandwidth on demand. While the picture is not altogether clear at this time, it is certain that new capabilities which are resulting from the current deployments will have significant implications for planners and providers of NS/EP services——e.g., the ability to provide real-time video from a disaster site for remote damage assessment or medical diagnosis and treatment. Recent advances in digital signal processing (DSP)——techniques for digitizing an analog signal and processing it in the digital domain, then changing it back into the analog domain——have resulted in significant increases in usable bandwidth over traditional wirelines used for PSTN access. However, in many respects, the universal transmission of data on such lines still has significant limitations. Specific limitations concern achievable data rates, the quality of transmissions, and the availability of shared access. While the trunks and switches of the public telephone network have largely evolved from an all analog environment to a virtually all-digital environment with major capacity and performance improvements, many of the local loops leading from the telephone company's central office to NS/EP user locations remain essentially analog. Notable exceptions are those user areas where Integrated Services Digital Network (ISDN) capabilities have been substantially deployed. An ISDN line is a digital telephone service that allows the transmission of voice and data over a single high-speed connection. In the United States, ISDN basic rate interface (BRI) service offers throughput rates of up to 128 kilobits per second (Kbps) when two bearer channels are used. However, ISDN is still basically a dialup, circuit-switched technology which tends to waste bandwidth on access wirelines used for Internet connections by utilizing the entire bandwidth even when no data is flowing over the connection. Additionally, ISDN service is generally more costly than a regular analog telephone line and although widely deployed, is still not yet universally available—especially in remote areas.

ADSL, a near-term alternative to ISDN, is a modem technology which uses advanced DSP, an analog coding algorithm, and existing copper twisted-pair telephone wirelines to move digital information between two points. It is specifically designed to exploit the one-way nature of most multimedia communications where large quantities of information flow toward the user and only a small quantity of interactive control information is returned. The current focus of the ADSL Forum and the American National Standards (ANSI) Technical Committee Working Group T1E1.4 is on new PSTN access systems that embody ADSL to deliver megabit services. The identification and resolution of technical interface and system parameter issues to ensure transparent delivery of application-based services and quality of service (QoS) capabilities across such systems are receiving special attention. Specific issues involve transmission techniques, user interfaces, and interface functionality for transmission services providing access to Web and client/server applications. Traffic management——particularly channel processing, error correction and rate adaption——is a key area of special interest. ATM is generally accepted as the leading transport mode for broadband terrestrial communications. Because of its flexible architecture, NS/EP Internet users and service providers are expected to be significant users of ATM technology. Consequently, it appears that issues with potential impact on the transport of ATM over access networks based on ADSL should be areas of special interest to the NS/EP community.

1.4 ORGANIZATION

This document is further divided into the following subsequent sections:

- Section 2.0, *ADSL Technology Overview;* Provides a brief overview of ADSL technology.
- Section 3.0, *ATM Over ADSL: Traffic Management Considerations*; Examines from an NS/EP perspective selected traffic management interface/system considerations related to the transport of ATM over ADSL.

1.5 REVISIONS

This document will be updated as directed by the Technology and Standards Division (N6), Office of the Manager, National Communications System (OMNCS). Comments and recommendations which may assist in the advancement of this effort are solicited and should be forwarded to:

Office of the Manager
National Communications System
Attn: Technology and Standards Division (N6)
701 S. Court House Road
Arlington, VA 22204-2198

SECTION 2.0

ADSL TECHNOLOGY OVERVIEW

Depending upon line length and loop conditions, the copper wirelines used in the local telephone loop to carry voice traffic between subscriber locations and the telephone company's central office are also capable of carrying megabits of data. However, since in telephone conversations 64 Kbps are sufficient to accurately reproduce the human voice, voice traffic only utilizes four kilohertz (kHz) of the available bandwidth. Theoretically, with the proper modulation scheme and coding, the unused bandwidth could be used to support high-speed digital connections to the PSTN for Internet access and other interactive multimedia applications. ADSL is one of several digital subscriber line (DSL) modem technologies used to transform ordinary twisted copper pair telephone lines into high-speed digital lines for PSTN/Internet access. Other variations of DSL technology include High data rate DSL (HDSL) which employs modems for twisted pair access at T1 or E1 speeds, and Very high data rate DSL (VDSL) for twisted pair access at even higher data rates. Both HDSL and VDSL are derivatives of ADSL. Although there may be some overlap in capabilities, there are significant differences in effective range and operating characteristics associated with each variation. While none of the DSL modem technologies represent an end-all solution for transport in the local loop, it appears that ADSL, the focus of this document, offers the most promise for support of the high bandwidth services and mode of operation currently envisioned for the Internet and WWW. Table 2-1, DSL Technologies and Capabilities, provides a brief summary of the general capabilities of the above DSL technologies.

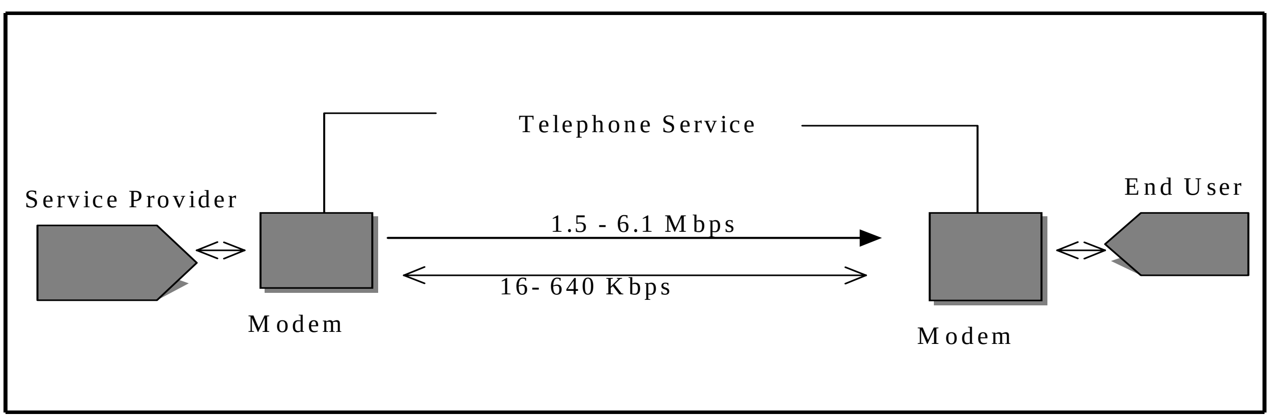
Table 2-1. DSL Technologies and Capabilities

Technology	Data Rate	Supported Applications
HDSL	1.544 Mbps 2.048 Mbps	T1/E1 service, WAN/LAN access, server access
ADSL	1.5-6.1 Mbps 16-640 Kbps	Internet access, video on demand, remote LAN access, interactive multimedia
VDSL	13-52 Mbps 1.5-2.3 Mbps	Same as ADSL, plus High Definition Television (HDTV)

Because ADSL uses the same copper wirelines as normal telephone service on a non-interference basis, subscribers are able to continue using their existing telephones while also maintaining digital connections. The ADSL modem simultaneously creates a high-speed downstream——from the network to the subscriber——digital simplex channel, a medium-speed digital duplex channel, a baseband analog duplex or plain old telephone service (POTS) channel, and an ADSL overhead channel (AOC) for framing, error control, operations, and maintenance. Full-duplex connections permit simultaneous communications in both directions. Simplex connections permit one-way service from the network to the customer with no capability for changing directions. All sub-channel rates are programmable in any combination of multiples of 32 Kbps. The maximum net data rate depends on the characteristics of the loop on which the system is deployed. The high-speed simplex downstream channel transmission rate ranges from 1.5 megabits per second (Mbps) to 6.1 Mbps, while the transmission rate for the medium-speed duplex channel ranges from 16 Kbps to 640 Kbps. Historically, services beyond POTS have been "asymmetric" in nature. That is, more information is transferred in one direction than the other. ADSL is "asymmetric" in that its two-way bandwidth is not evenly divided between the upstream and downstream directions. However, ADSL offers both asymmetric and symmetric (synchronous) modes of operation.

As illustrated in Figure 2-1, ADSL Channel Rates, most of the two-way bandwidth is devoted to the downstream direction. Only a small portion of bandwidth is devoted to the upstream direction——from the subscriber to the network. Each channel can be sub-multiplexed to form multiple, lower rate channels. The actual downstream data rates depend on a number of factors including the length of the copper line, its wire gauge, and the presence or absence of bridged taps——sections of unterminated twisted-pair cables connected in parallel across the cable——and-cross-coupled interference. These factors may introduce impedance unbalances or undesired signals in the channel thereby affecting its effectiveness when used for high-speed data. The asymmetric nature of the divided bandwidth makes ADSL especially suited for Internet and selected multimedia-intensive Web and client/server applications, since most Internet and multimedia-intensive client/server data are transmitted downstream. Upstream requests and responses to the network from users are small in comparison and require little upstream bandwidth.

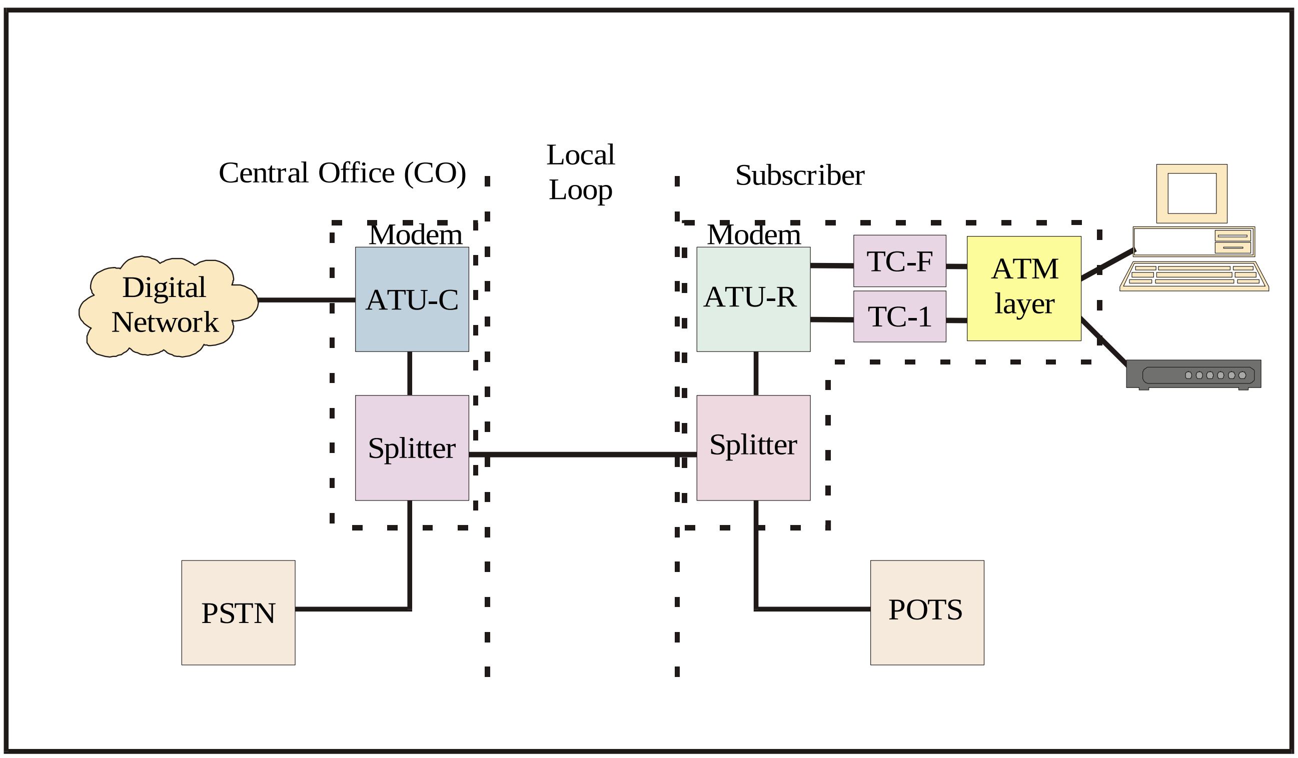
Figure 2-1. ADSL Channel Rates



2.1. ADSL MODEM

The ADSL modem is the key element which makes ADSL operations possible. As depicted in Figure 2-2, ADSL System Reference Model, in a typical ADSL installation, a single twisted pair of telephone wires is used to connect the ADSL Transceiver Unit-Remote (ATU-R) at the subscriber's location to the ATU-Central (ATU-C) at the telephone company central office. The installed modems create a high-bandwidth downstream channel, a smaller upstream channel, and a basic telephone service (POTS) channel for voice. The POTS channel is split off from the digital modem by filters for routing of the POTS signal in the circuit-switched network, thus guaranteeing uninterrupted POTS in the event the modems fails. The data signals are routed into the telephone company's high-speed packet-switched network. Simultaneously, ADSL conveys the channels and line overhead for framing, error control, operations, and maintenance. The mode of operation——asymmetric or symmetric——are application options. The ATU-R and ATU-C may be configured for either type of transport, e.g., synchronous transfer mode (STM) bit synchronization transport or ATM cell transport.

Figure 2-2. ADSL System Reference Model



At the subscriber's location, service modules (set-top boxes, routers, PC interface devices, data terminals) attached to the premises wiring scheme interconnect subscriber premises equipment to the local loop via an ATU-R. The ATU-R is connected to the local loop via a splitter. Splitters are filters that separate voice and data traffic by separating the high frequency ADSL signals from the low frequency POTS signals. At the network end of the loop, the circuit terminates in another voice-data splitter. The network end splitter connects to an ATU-C, which in turn is connected to, or integrated within, an access node——a concentration point for broadband and narrowband data. The access node provides access to a digital network. This configuration allows TV signals, interactive video, Internet access, and a wide variety of other data types to share access to the ADSL-equipped local loop.

The ATU-R functional block shown in Figure 2-2 (1) terminates and originates the ADSL signal entering and exiting the subscriber's premises via twisted pair cable; (2) handles the interface with the subscriber's terminal equipment; and (3) performs operation, administration, and maintenance (OAM) functions. In configurations where the subscriber's terminal equipment is integrated within the functional block, the interfacing subfunction is not necessary and may be eliminated. Two separate paths——a fast transmission convergence (TC-F) path and an interleaved transmission convergence (TC-I) path——are provided between the ATU-R and ATM layer functions to provide simultaneous transport of multiple data channels in a particular direction with dual latency, if needed. Latency is the inherent delay induced by a network element in handling a frame from the time it is received on an input port of the functional block to the time it is transmitted out on an output port. Cyclic redundancy check (CRC) generation, scrambling, and forward error correction (FEC) coding are independent on each path.

The primary difference between the two paths is that the interleaved path contains an interleaving function in the transmitter and a de-interleaving function in the receiver, whereas the fast path does not. Performance of the interleaving/de-interleaving functions introduces delay in the interleaved path that is not present in the fast path. Consequently, bits transmitted on the fast path pass through the transmitter and receiver with less delay than bits transmitted via the interleaved path. ADSL has the option of operating in a single latency mode with low delay——fast or interleaved——in which all data is allocated to one path, or in an optional dual latency mode with greater delay, in which data is allocated to both paths. However, although provisions for two paths are provided, according to the ADSL reference model it is only mandatory to implement a single path.

2.2 MULTIPLE CHANNEL TECHNIQUES

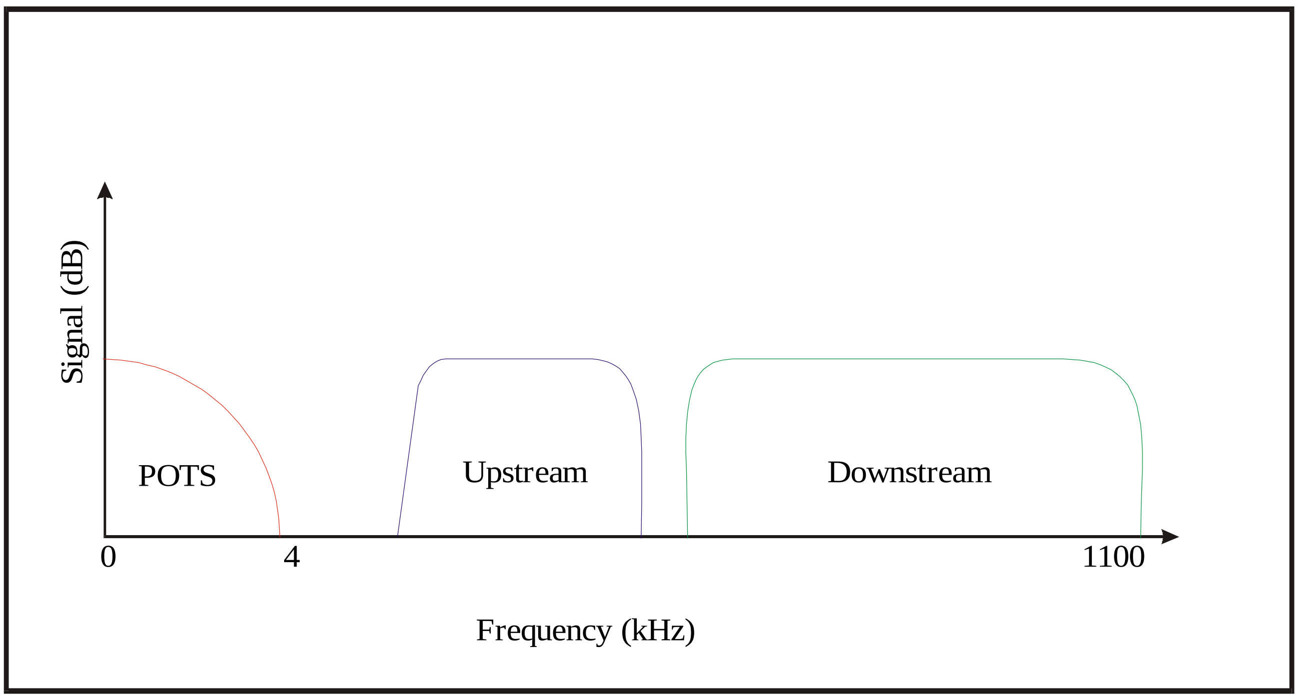
To create multiple channels, ADSL modems divide the available bandwidth of the wireline by use of one of two techniques: frequency division multiplexing (FDM) or echo cancellation. FDM assigns one band for upstream data and another band for downstream data. Echo cancellation assigns the upstream band to overlap the downstream band and separates the two by means of adaptively forming a replica of the echo signal arriving at a receiver from its local transmitter and subtracting it from the signal at the input to the receiver. Echo cancellation uses bandwidth more efficiently, but is more complex and costly to implement. With either technique, ADSL splits off a 4 kHz region for POTS at the lower end of the band. Because voice and data traffic are transported on logically separate paths, the data can be routed around the local switch, thus providing load relief without massive hardware and software upgrades. A more detailed discussion of FDM and echo cancellation techniques is provided below.

2.2.1 Frequency Division Multiplexing

With FDM two or more simultaneous, continuous channels are derived from a transmission medium connecting two points by assigning separate portions of the available frequency spectrum to each of the individual channels. With ADSL, FDM assigns one band for upstream data and another band for downstream data. The upstream and downstream paths are then further divided into one or more high-speed channels and a corresponding number of (one or more) low-speed channels by time division multiplexing (TDM). TDM is a method of multiplexing whereby a common transmission path is shared by a number of channels on a cyclical basis by enabling each channel to use the path exclusively for a short time slot. In this way a circuit capable of a relatively high information transfer rate is divided to provide a number of lower speed channels. Figure 2-3, FDM Bandwidth Division, illustrates the FDM bandwidth division discussed above.

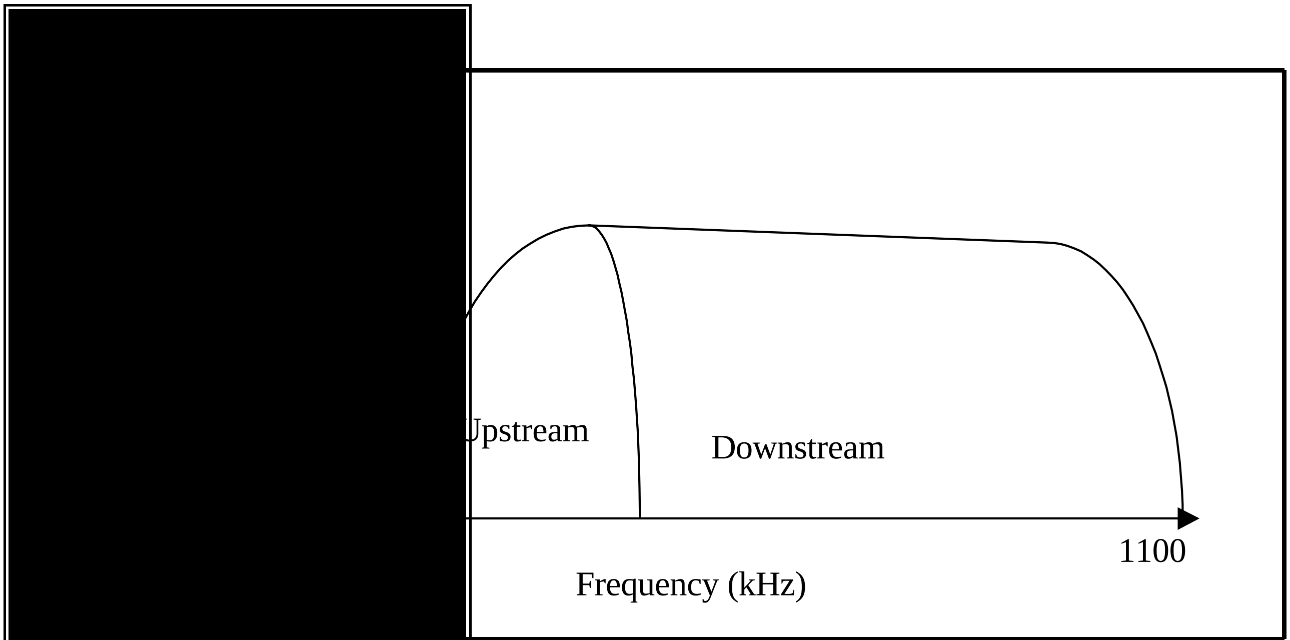
Figure 2-3. FDM Bandwidth Division

2.2.2 Echo Cancellation



Echo cancellation is a form of echo suppression used on circuits with long transmission times to attenuate the direction of transmission which is not active so that any echoes which are fed into a circuit do not return to the starting point and cause confusion. With ADSL, echo cancellation assigns an upstream frequency to overlap the downstream frequency band, and separates the two by constructing a signal that closely approximates the echo component and subtracts it from the locally transmitted signal to stop a received signal from being transmitted back to its origin. Allowing the upstream and downstream signals to be superimposed or overlapped so as to basically share the same bandwidth/frequency, effectively increases the overall available ADSL bandwidth. Figure 2-4, Echo Cancellation Bandwidth Division, illustrates the bandwidth separation achieved by the employment of echo cancellation.

Figure 2-4. Echo Cancellation Bandwidth Division



2.3 ADSL MODULATION

Vendor efforts to date have resulted in two main competing line codes or signal modulation algorithms to conserve bandwidth in ADSL implementations: Carrierless Amplitude/Phase (CAP) modulation and Discrete Multi-Tone (DMT) modulation. The first modulation algorithm, CAP, is a variation of quadrature amplitude modulation (QAM) coding. QAM is a high-density modulation scheme used by most modems today to increase the amount of information that can be carried within a given bandwidth. CAP is well understood and chipsets for the process are readily available. However, CAP is only offered by AT&T Paradyne at this time. The second modulation algorithm, DMT, has been ratified by ANSI as the coding standard for ADSL. Due to computational complexity and the required number and size of DSP devices, program memory, and temporary memory, DMT is considerably more complex and costly than CAP. Consequently, despite ratification of DMT as the ADSL coding standard, modems utilizing both CAP and DMT are still being tested to further examine implementation costs and performance attributes over a wide range of conditions. Based on the results of testing to date, a present view of the two schemes seems to indicate comparable performance with DMT having some advantages in digital processing, but requiring greater cost in analog circuitry [33]. A further discussion of each technique is provided below.

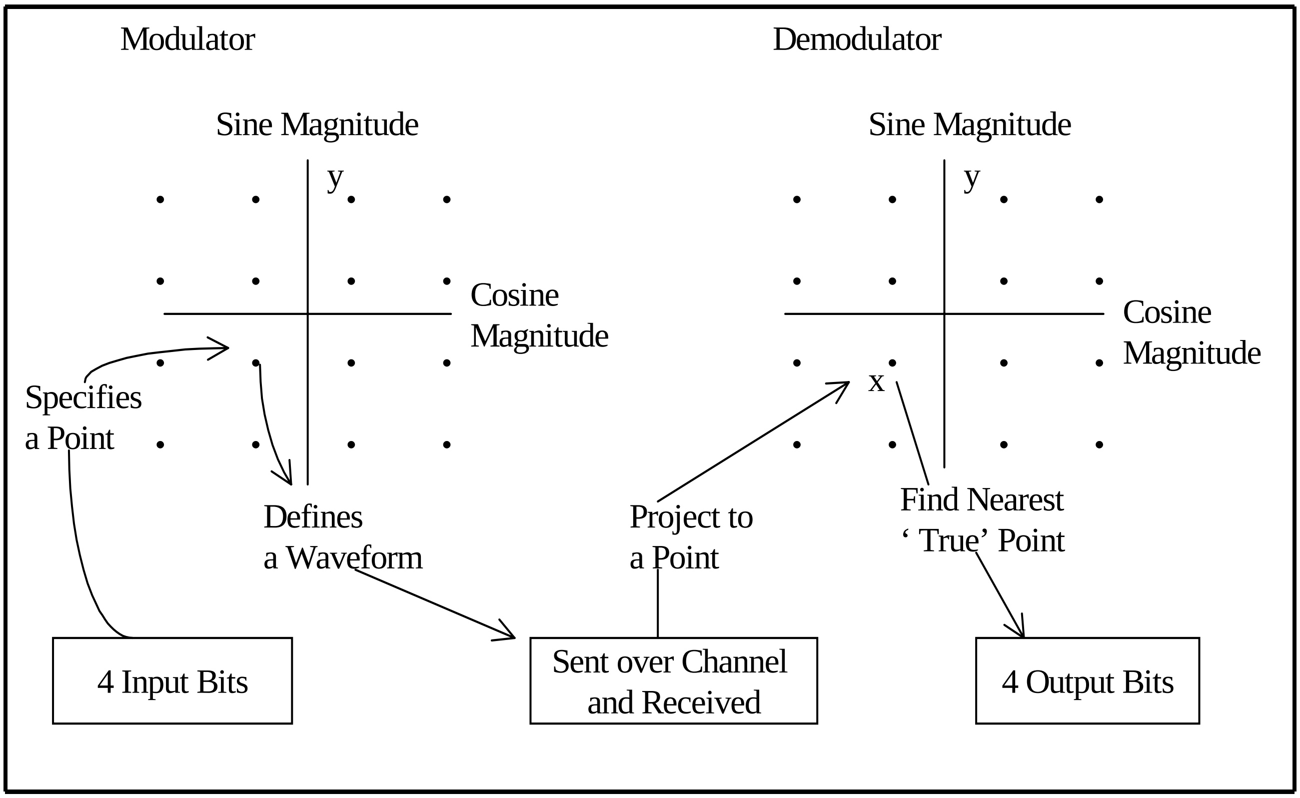
The DMT modulation block is the heart of the ADSL modem transmitter. In typical DMT systems all processing is confined to a single block. Before entering the DMT modulation block, the payload to be transmitted consists of a buffer of data from the fast channel and a buffer of data from the interleaved channel when both channels are implemented. Modulation converts input bits into waveforms to be sent over the channel. Demodulation in the receiver maps received waveforms back to the bits that generated the waveform—that is, recreates the bits at the demodulator output that were present at the modulator input. With digital systems, the number of bits or hertz that can be supported with an encoding scheme is an important consideration. Encoding is the generation of character signals to represent quantized samples. A frequently heard complaint of digital systems is that they can be wasteful of bandwidth when compared with an analog system. For example, an analog voice channel is nominally 4 kHz, whereas a digital voice channel is 64 kHz——assuming one bit per hertz of bandwidth——a 16-to-1 difference in the required bandwidth. Because of the perceived wastefulness, various techniques involving the use of more efficient line codes to permit shared use of loop bandwidth for both analog and digital data are receiving increased emphasis. Line coding is the process of encoding a digital signal into an analog signal that represents and conveys the information over an analog channel. Since the PSTN loop is both bandwidth limited and power limited, line coding schemes generally focus on manipulating the analog carrier signal.

An analog signal has at least three attributes with relevance to modulation: amplitude, frequency, and phase. Modulation involves manipulating one or more of these attributes to increase the number of bits/hertz in a transmission line. One technique that is becoming widely used is bit packing——packing more into a hertz of bandwidth——using some form of QAM coding. QAM utilizes a sine wave and a cosine wave with the same frequency component to convey information. The waves are sent simultaneously over a single channel with the sign and magnitude of the amplitude of each wave manipulated to convey the information being sent. In the simplest modulation case, a single signal element represents one bit. However, with complex coding techniques such as QAM, single signal elements may represent multiple bits, thereby increasing the effective data rate. 64 QAM is the most commonly used scheme. However, 4 QAM, also known as quaternary phase shift keying (QPSK)——a compression technique used in modems and in wireless networks——and 256 QAM are also used. Figure 2-5, Typical 16-QAM Constellation, shows a typical 16-QAM system. This example is sometimes called 16-QAM because the constellation has 16 points. In this example, the four bits of information to be transported are mapped to 1 of 16 points on the QAM constellation. For the four bits of information, 16 points allow a unique point for any combination of bits. The x and y components of the point to which the bits are mapped specify the amplitude of the cosine wave and the sine wave to be sent over the channel. Both the transmitter and receiver know the predetermined method of mapping between the bits and a point.

After cosine and sine waves are sent over the channel, the receiver recovers and estimates the amplitude of each wave. These magnitudes are projected on a constellation identical to that used at the transmitter. Noise and distortion in the channel and less than optimum performance by the transmitter and receiver may prevent an exact projection of the point onto the receiver's constellation. However, the receiver selects the closest point to the projected point as the point the transmitter is most likely to have used to generate the QAM symbol. Then in the opposite direction of that used by the transmitter, the receiver maps that point into four bits using the same mapping method employed in the transmitter. If too much noise is present at the receiver, an inaccurate point could be projected on the constellation, resulting in an error in estimating the recovered symbol.

Figure 2-5. Typical 16-QAM Constellation

2.3.1 CAP Modulation



CAP is a single-carrier modulation technique that uses three frequency ranges for ADSL operation. Similar to a QAM modulator, CAP uses a constellation to encode bits at the transmitter and decode bits at the receiver. With CAP, incoming data modulates a single carrier that is then transmitted down a telephone wireline. The carrier itself is suppressed before transmission and reconstructed at the receiver——therefore, the term ”carrierless.” CAP handles speeds of up to 1.5 Mbps. With ADSL, CAP supports the three ADSL channels—POTS, downstream data, and upstream data channels—by splitting the frequency spectrum. Voice occupies the 0-4 kHz frequency band. The remainder of the frequency spectrum is used by the upstream channel and the high-speed downstream channel. A guard band is used to separate the voiceband POTS from the ADSL frequencies. CAP treats the data channels as one big pipe through which to send as many bits as possible.

2.3.2 DMT Modulation

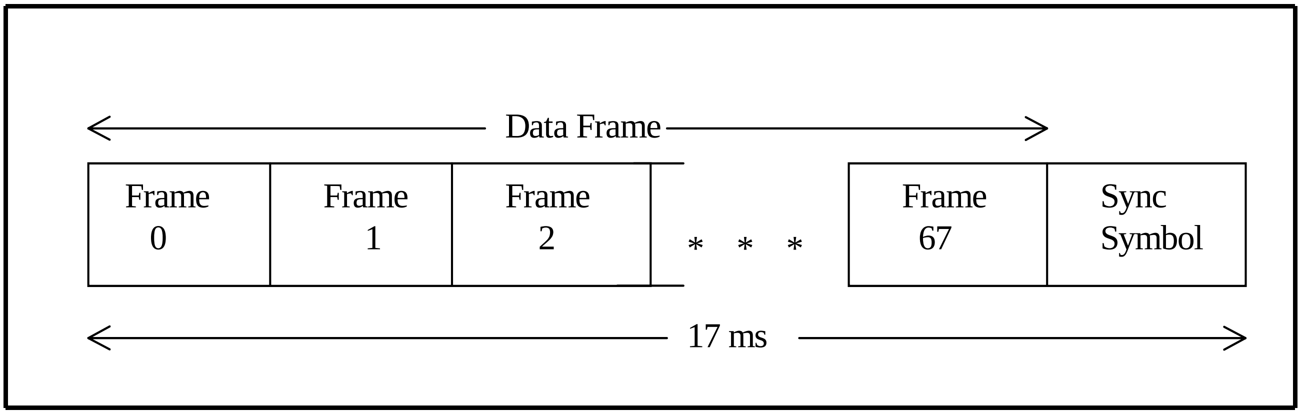
DMT is a multi-carrier modulation technique based on the fact that it is possible to carry digital signals on multiple carriers whose spectra overlap, and to separate the signals without interchannel interference. DMT is newer and faster than CAP. It handles speeds of up to 6 Mbps. Its performance enhancing capabilities lie in its ability to work with a large number of sub-carriers rather than a single carrier. DMT expands on several of the features of QAM by employing multiple constellation encoders. Each encoder receives a set of bits that are coded using a constellation encoder. As with QAM and CAP, the output value from the constellation encoder are the amplitudes of cosine and sine waves. However, a different sine and cosine frequency is used for each constellation encoder. All the sine and cosine waves are then summed together and sent over the channel.

DMT divides the 1.1 MHz data transmission channel into 256 4-kHz sub-channels and then selects the best ones on which to send data——hence the term "multi-tone." Each sub-channel is independently modulated using QAM and has its own carrier. The actual bit rate for each sub-channel depends on line and noise conditions. The signal-to-noise ratio (SNR)——the ratio of the amplitude of the desired signal to the amplitude of noise signals at a given point in time—is continuously monitored by the DMT system and the system dynamically adjusts each channel accordingly. To accomplish the required adjustments, the system varies the bit densities on each channel to overcome noise and interference that may be present in sections of that spectrum. Certain sub-channels can be left unused due to external interference. Because of its ability to maximize throughput on good channels and minimize throughput on channels with heavy interference, DMT proponents argue that on noisy lines DMT is better than CAP.

2.4 FRAMING

Understanding the framing structure of ADSL is key to understanding how the different logical channels are combined or separated when transmitted on a single ADSL physical channel. ADSL uses a superframe structure composed of 68 ADSL data frames which are encoded and modulated into DMT symbols. The superframe structure is 17 milliseconds (ms) long. As shown in Figure 2-6, ADSL Superframe Structure, the DMT symbols are followed by a synchronization symbol inserted by the modulator to establish superframe boundaries. The synchronization symbol carries no user or overhead bit-level data. The ADSL modem synchronizes the data channels to the ADSL DMT frame rate and multiplexes them into separated fast and interleaved (to be discussed later) data buffers. A CRC, scrambling to randomize the data, and FEC coding is then applied to the contents of each buffer separately and the data from the interleaved buffer is subsequently passed through an interleaving function prior to being combined with data from the fast buffer. The two data streams are then tone ordered and combined into a data symbol as input to an encoder. Tone ordering is the process of extracting bits and assigning them to tones to minimize errors. In the encoder, the extracted bits are encoded into complex values for input to the DMT modulation block.

Figure 2-6. ADSL Superframe Structure



SECTION 3.0

ATM OVER ADSL: TRAFFIC MANAGEMENT CONSIDERATIONS

ATM——a broadband, low delay, packet-like switching and multiplexing technique——segments voice, data, multimedia, and video into fixed-size cells for transparent transmission. It is asynchronous in that the recurrence of cells from an individual user is not necessarily periodic. ATM traffic patterns are generally unpredictable. While a great deal is understood about the pattern of regular voice traffic, the pattern for bursty traffic such as video, data, or compressed voice is less understood. Recent research studies indicate that such traffic will not "smooth out" over an extended period of time—smooth traffic behaves like random traffic that has been filtered. For normal voice traffic, the filtering or limiting of peakiness is accomplished by call blockage during the busy hour. For bursty traffic, there will always be some possibility that shared link bandwidth will be temporarily exceeded and cells will have to be dropped. To preclude this possibility requires considerable buffer support at every contention point where ATM terminals compete for access to the supporting telecommunications medium. Efficient use of the supporting telecommunications media requires high utilization——operating at or near capacity. Contention——a condition arising when two or more data stations attempt to transmit at the same time using a shared channel——impedes high utilization and degrades QoS. The challenge for the supporting telecommunications medium is to maximize utilization without negatively impacting QoS. The key to meeting this challenge is traffic management.

The principal role of traffic management is to protect the network and the end-system from congestion. With ATM over ADSL, several mechanisms are used in appropriate combinations to monitor and control the flow of traffic and congestion within the ADSL local loop. Because ATM is connection-oriented, the transport of ATM cells over ADSL requires the establishment of an end-to-end virtual channel/circuit (VC) prior to the transmission of user data. A VC is a pre-established communications channel. Channelization of different ATM payloads——the allocation to channels of the total data rate available for user data in any one direction——is embedded within the ATM data stream using different VCs and/or virtual paths (VPs). VPs are logical associations or bundles of VCs. ATM transport over ADSL is supported by mapping the ATM data stream into predefined ADSL sub-channels. Data applications using ATM over ADSL access loops require low cell loss rates to avoid throughput collapse due to packet retransmission. Since many data applications cannot predict their own traffic parameters, it is difficult for a call-level traffic control scheme to optimize both low cell loss and the efficient use of supporting telecommunications media resources. Sophisticated cell scheduling is required to avoid the bandwidth starvation problem associated with higher priority traffic monopolizing the port's transmission bandwidth.

With ADSL systems as with any access network where two or more users share a common medium on a dynamic basis, the risk of collision is always present. With conventional multiplexing, e.g., TDM or FDM, channels are multiplexed by assigning each channel a fixed time-slot in a periodic frame structure (TDM), or a separate portion of the frequency spectrum (FDM). With ATM, statistical multiplexing allows multiple connections to share the same output-port bandwidth. The arrival pattern for ATM cells at a particular channel is irregular. However, with statistical multiplexing, cells are multiplexed according to when they arrive. This provides traffic sources with a great deal of flexibility since they are now free to generate cells as necessary and do not need to generate data at a constant bit rate. Because of the sophisticated cell scheduling provided by statistical multiplexing, the potential bandwidth starvation problem associated with the monopolization of transmission bandwidth by higher priority traffic is avoided. A form of statistical multiplexing is already in use on transatlantic telephone links to increase link capacity. The technique is referred to as Time Assigned Speech Interpolation (TASI) where speech packets are not generated during silence intervals when a person is not talking.

With respect to traffic management, only the physical medium dependent (PMD) and the transmission convergence (TC) sublayers of the physical layer, and the ATM layer of the Broadband-ISDN (B-ISDN)/ATM reference model are relevant. The PMD sublayer defines the parameters at the lowest level of the physical layer, such as speed of the bits on the media. The TC sublayer transforms the flow of cells into a steady flow of bits and bytes for transmission over the physical medium. On transmit, the TC sublayer maps the cells to the frame format, generates the header error control (HEC), and sends idle cells when the ATM layer has none to send. On reception, the TC sublayer delineates individual cells in the received bit stream, and uses the HEC to detect and correct received errors. The ATM layer is completely independent of the physical medium. The functions of the ATM layer include cell header generation and extraction. In the transmit direction, the cell header generation function receives a cell information field from a higher layer and with the exception of the HEC sequence, generates an appropriate cell header. In the receive direction, the cell header extraction function removes the ATM cell header and passes the information field to a higher layer.

QoS at the ATM layer is defined by network parameters such as cell transfer delay (CTD), cell delay variation (CDV), cell loss ratio (CLR), and cell error ratio (CER). CTD is the sum of the total cell transmission delay and node processing delay between two measurement points. CDV is a component of CTD, induced by buffering and cell scheduling. CLR is a negotiated QoS parameter and acceptable values are network specific. CER is the ratio of errored cells in a transmission in relation to the total cells sent in a transmission. At the time of connection establishment, the user requests a specific QoS from the QoS classes which the network provides. The traffic contract agreed to at that time is the network's commitment to furnish the agreed-upon QoS for as long as the user complies with the terms of the contract. Usage parameter control/network parameter control (UPC/NPC)——a set of actions taken by user equipment and/or the ADSL network at the user/network access point to monitor and control traffic——assures conformance to the contracted parameters. Because of the impact certain ADSL network features have on the ability of the supporting ADSL network to fulfill ATM QoS guarantees, selected network features must receive special consideration. Specific features include: (1) channel processing capabilities, (2) the adequacy of forward error correction capabilities, and (3) the ability of the ADSL system to perform rate adaption. Each of these features will be examined in more detail below.

3.1 CHANNEL PROCESSING CAPABILITIES

The ADSL Forum specifies that: "For the transport of ATM on modems compliant with ANSI T1.413 standard, channels shall be independently set to any bit rate that is an integer multiple of 32 Kbps, up to a maximum aggregate capacity determined by the start-up process. In addition, for each channel the bit rates for the upstream and downstream directions may be set independently from each other."[2] While the ADSL data multiplexing format is flexible enough to allow other transport data rates, such as channelization based on 1.544 Mbps, the rates for all bearer channels are based on multiples of 32 Kbps. Part of the ADSL system overhead is shared among the bearer channels for synchronization. The remainder of each channel's data rate that exceeds a multiple of 32 Kbps must be transported in the shared overhead. For ATM systems, the ADSL physical layer is viewed as simply a point-to-point bit stream provider. The channelization of different ATM payloads is embedded within the ATM data stream. Transport over ADSL is accomplished by mapping the ATM data stream into the ADSL bearer channels.

According to the draft ANSI standard [3], the processing of ATM data for transport over ADSL can roughly be divided into three logical functions performed by the ADSL modem: (1) a channel processor function, (2) a TC sublayer function, and (3) a PMD function. The channel processor function multiplexes the incoming bearer channels, and generates two-byte fast and interleaved data streams. The TC sublayer function accepts the two-byte data streams from the channel processor and performs synchronization, framing, scrambling, FEC, and interleaving. The PMD function accepts two-byte data streams from the TC sublayer function and performs the DMT modulation required to produce analog signals and place them on line for transmission. The functions performed by this layer include bit alignment, serial to parallel bit conversion, line coding, tone ordering, and analog to digital conversion. Figure 3-1, ADSL Functional Modules, is a notional depiction of the relationship between the three functional modules described above.

In examining the three logical functions described above, there appears to be some duplication in the functions performed by the ADSL system modem and two of the three layers of the B-ISDN/ATM reference model. In the B-ISDN/ATM reference model, synchronization is a physical layer function——more specifically, the TC sublayer of the physical layer. It is the TC sublayer that performs all functions required to transform a flow of cells into a data stream, or flow of data units, which can be transmitted and received over a physical medium. One of the primary functions of the TC sublayer is the generation and recovery of transmission frames. Another function is transmission frame adaption——structuring the flow of cells according to the payload structure of the transmission frame in the transmit direction, and extracting the cell flow out of the transmission frame in the receive direction. The transmission frame may be a cell equivalent where no external envelope is added to the cell flow, an SDH/SONET envelope, an E1/T1 envelope, or in this case, an ADSL transmission frame. On transmit, the TC sublayer maps the cells to the frame format, generates the HEC, and sends idle cells when the ATM layer has none to send in order to adapt the rate of valid cells to the payload capacity of the ADSL transmission channel. On reception, the TC sublayer delineates individual cells in the received bit stream, and uses the HEC to detect and correct received errors.

The ATM layer performs the multiplexing/demultiplexing function and cell routing to the

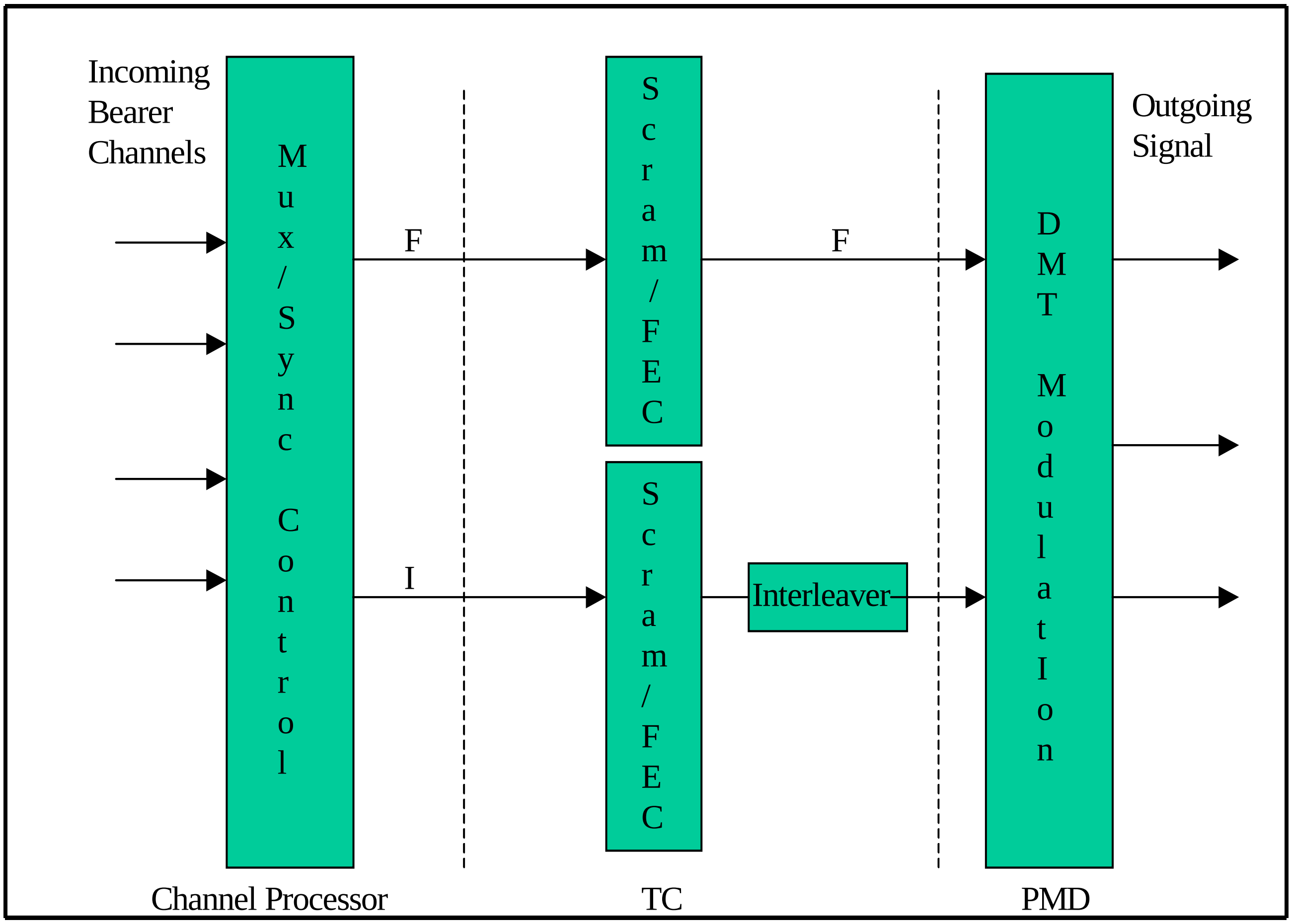


Figure 3-1. ADSL Functional Modules

appropriate ADSL modem physical port based on the virtual channel identifiers (VCIs) and virtual path identifiers (VPIs) contained in the ATM cell header. A VC is a communications channel that provides for the sequential unidirectional transport of ATM cells——it must be established prior to the transmission of user data. A VP is a unidirectional logical association or bundle of VCs. Since the above functions are already performed in respective layers of the B-ISDN/ATM reference model, it appears that including synchronization as an ADSL modem TC sublayer function and multiplexing as an ADSL modem channel processing function for ATM transport could result in unnecessary duplication. Consequently, it appears that this duplication could result in reduced overall throughput due to an unnecessary increase in overhead and implementation complexity. Under certain scenarios, it appears that the potential reduction in transmission capacity available for bearer payloads could have significant impact on real-time, high data rate NS/EP applications such as real-time video——especially during periods of heavy traffic or degraded communications conditions.

Another consideration in optimizing the channel processing capabilities of ADSL systems is the selection of the technique for dividing the available bandwidth. Of the two multiple channel techniques discussed earlier—FDM and echo cancellation—for systems with simultaneous symmetric data rates in both directions over a twisted wire pair, the latter appears to be the preferred way to provide the required separation. However, frequency or time separation involves a bandwidth penalty. When the ratio of upstream to downstream bandwidth is low——such as with the use of ADSL for Internet access——it appears questionable whether the relatively small benefits in bandwidth utilization offered by the latter is worth the high cost and complexity of its implementation.

Currently, echo cancellation is not used in single-channel ADSL designs. Instead, sharp active analog filters are employed to separate the relatively narrow low-frequency upstream band of frequencies from the wider downstream band of higher frequencies. However, in multitone systems echo cancellation is generally required. In a typical DMT system, all processing is confined to a single block. In order to avoid interblock interference, a guard interval of time is inserted at the transmitter between transmitted blocks. In order to avoid having to perform time domain equalization functions at the receiver prior to DMT processing, the duration of the guard interval should be less than the dispersion of the channel. If a multitone system depended on frequency separation, time dispersion introduced by the phase distortion of the separation filters and the elimination of sub-channel sidelobes could be greater than the guard interval. The resulting requirement for time domain equalization prior to the receiver's DMT processing could obviate one of the prime advantages of multitone——the need to perform equalization only in the frequency domain. The outcome of this dependency on frequency separation may be increased cost, complexity, and latency with potential impact on delay/error sensitive NS/EP traffic.

3.2 FORWARD ERROR CORRECTION

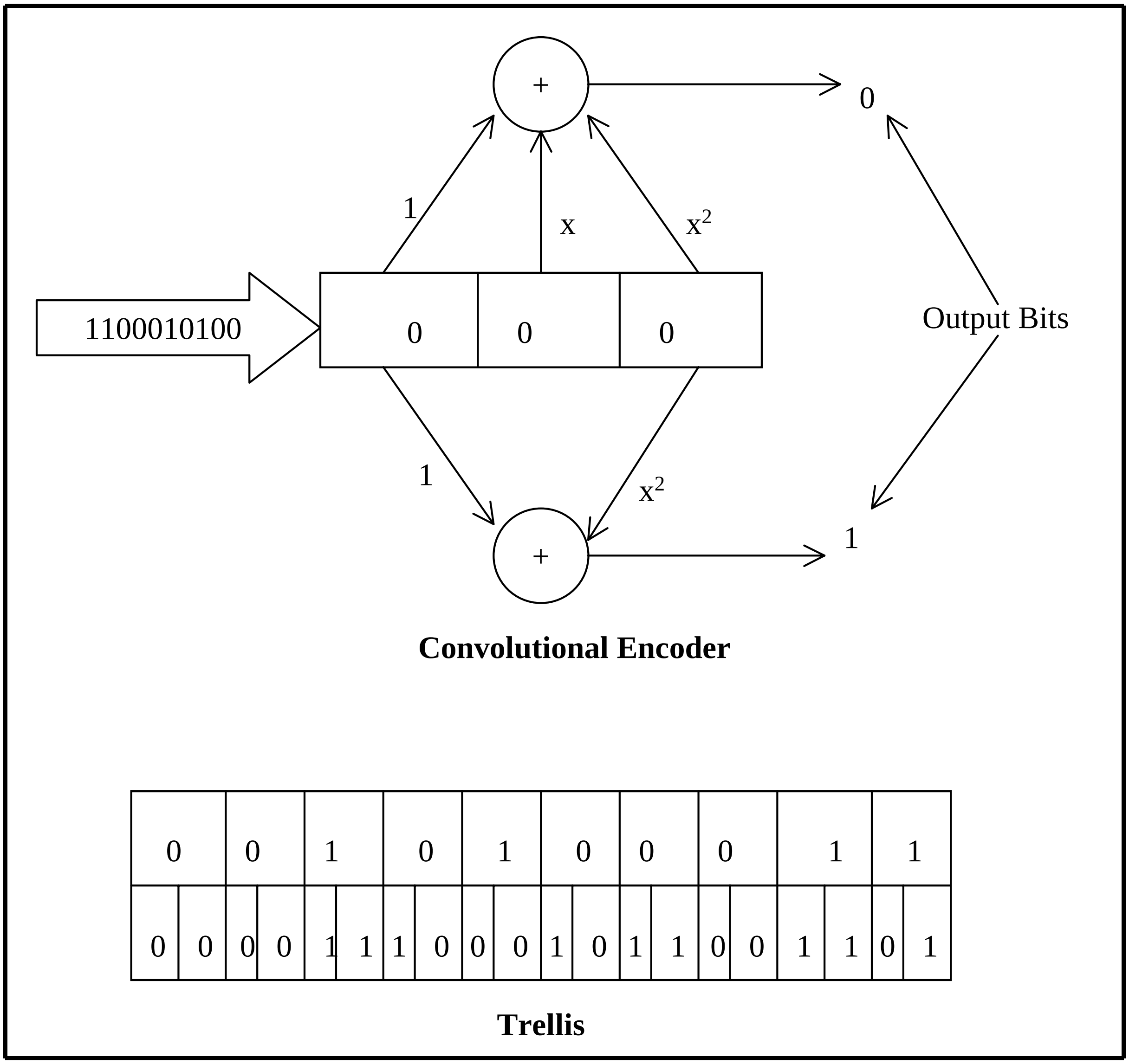
Impulse noise——noise consisting of random occurrences of energy spikes, having random amplitude and bandwidth——can be a significant source of errors in data channels. FEC codes are used to detect and correct randomly occurring bytes in long data streams. FEC is a technique for controlling errors in data transmissions by using self-correcting binary codes to detect and correct at the receiver, any character or code block which contains a predetermined number of bits in error. FEC is accomplished by adding bits to each transmitted character or code block using a predetermined algorithm. It does not require a feedback channel, and therefore may be used with a one-way transmission system. While a FEC block adds redundancy to the data to be transmitted, this redundancy is usually only a small portion of the actual transmitted payload. The additional redundancy is often more than compensated for by the ability of a well designed FEC block to correct bits that the demodulator decodes incorrectly.

The two types of FEC commonly used in DSL systems are a cyclic block code commonly called *Reed-Solomon coding* and a convolution code known as *trellis-coded modulation (TCM)*. Because of their simplicity, CRC codes are used in most layer 2 protocols including high-level data link control (HDLC)-based schemes and Ethernet. Normally, CRC codes operate on bits and are used for error detection but not error correction. With Reed-Solomon codes, each symbol is made up of eight bits——one byte. Each symbol can take on one of 256 values——zero through 255. The algorithm rules define the results when two symbols are added, subtracted, multiplied, or divided. Error correction is accomplished at the receiver.

With TCM, the convolution coding is tightly tied to the constellation encoding used in both CAP/QAM and DMT. Encoding is relatively simple. However, decoding is considerably more complex. Convolution coding differs from block coding in that the former does not have codewords made up of distinct data sections and block sections. Instead, redundancy is distributed throughout the coded data. With TCM, the Convolutional encoder can be viewed as a finite state machine which changes state based on an input condition and produces an output when it changes states. Figure 3-2, Convolutional Coding: Input/Output, shows a simple example of using an encoder to encode a set of bits. Note that the initial condition of the encoder is assumed to be all zeros and that the top output is taken before the bottom output. Each data bit input to the encoder is shifted to produce a predetermined number of data output bits. The outputs are defined by generator polynomials.

Each node in the trellis represents a state. Each line represents a pair of received bits. The decoding algorithm allows for multiple paths through the trellis to be kept active. At each node, a running sum, or metric, is computed that counts the mismatches between the received bits and the path bits. By tallying the number of mismatches or errors that have occurred along each path, the decoding algorithm terminates those with a high error count. As the decoding process continues, the path with the lowest error count or metric will become obvious, and all other paths will be terminated. It is the low metric path that will be used to produce the output at the decoder. If no error occurs, the bits received while in a received state correspond to one of the paths leaving the state. When errors occur on the line, the received bits may correspond to neither of the lines leaving the state, and knowing which line to follow is unclear. However, by using the decoding algorithm, bad paths are eventually terminated and only one main path survives. The output generally produces the correct sequence.

For ATM transmissions, the ATM cell is divided into two parts——a five octet header portion and a 48 octet payload portion. Error correction occurs only in the 8-bit HEC field of the cell header. The code used is capable of either single-bit error correction or multiple-bit error detection, but only if they occur in the cell header. There is no capability within the ATM cell to correct errors in the payload. The transmitting side computes the HEC field value. Each cell header is examined as it is received using CRC. The receiver is capable of operating in two modes. The correction mode is the default mode. If an error is detected, one of two actions occurs. With the receiver in the correction mode, a single-bit error is corrected and the receiver switches to the detection mode. In the detection mode, all cells with detected header errors are simply discarded. When a header is examined and found not to be in error, the receiver returns to the default mode[13].



Fi

gure 3-2. Convolutional Coding: Input/Output

However, even with the combined FEC cell header and transmission oriented capabilities inherent in ATM over ADSL, FEC may not be sufficiently effective in certain environments because of the length of the continuous blocks of ATM data transmitted over ADSL. FEC codes can typically correct only eight bytes in a transmitted data block. Data blocks containing more than eight bytes in error are generally discarded. To reduce the possibility of cell discard, ADSL modems adopt an interleaving process to re-order the byte stream so that bursts of errors are distributed evenly among FEC code words. Interleaving spreads the effect of impulse noise interference over many widely separated bits with the aim of reducing the number of byte errors in a data block to a number correctable by the FEC code. For example, an interleaver with a depth or memory capacity of 64 code words, could distribute up to 8 x 64=512 byte errors over 64 code words. The greater the depth of the interleaver, the longer the impulse that can be corrected. However, interleaving adds delay to the end-to-end transport of data and also requires memory buffers at both the transmitter and receiver. Consequently, re-ordering the byte stream implies buffering many codewords, and the introduction of delay proportional to the depth of the interleaver.

For high-speed ATM cell transport, the process of interleaving may result in intermittent congestion at the ADSL buffer. In such instances, ATM cells may be discarded due to overloading of the buffer, or late arrival of the cells at the buffer because of excessive CDV. As a consequence of congestion, interleaving itself could initiate buffer overflows resulting in the loss, discard, or delay of cells. The number of consecutive cells dropped for a given VCI/VPI will depend on the service bit rate, multiplexing queue, and the capacity of the buffer. A standard implementation delay might be on the order of a few milliseconds. However, it is possible that the time scale of noise events could be significantly greater with a corresponding increase in interleaving delay. Even with induced errors corrected, such delays might not be acceptable for certain NS/EP applications——in particular those applications that are more concerned with delay than error rate.

3.3 RATE ADAPTION

The wide variation in characteristics of existing copper wirelines has the potential to allow non-linearities in copper-based ADSL networks. Non-linearities frequently result in impairments introduced by equipment or line conditions which may vary with time and/or weather conditions. Some examples are: (1) the lack of constancy of SNR across the frequency band, (2) the increase in signal loss and noise at the higher frequencies, (3) clipping caused by transmit amplifier saturation and digital-analog (D/A) and analog-digital (A/D) range limits, (4) the generation of spurious signals at various sum and difference frequencies of combinations of signal components, (5) frequency offset and phase noise, (6) timing jitter, (7) impulse noise, and (8) narrowband interference. To compensate for non-linearities in the local loop, the bit rate, particularly in the downstream direction, should be automatically adjustable in order to provide acceptable performance over a variety of subscriber lines. The capability to determine and provide the maximum bit rate that a particular line can support should be inherent in the DMT bit allocation algorithm. In the current version of ADSL, each ADSL modem receiver must synchronize to the other modem's transmit signal during the initialization sequence. This is accomplished by sending a known signal down the wireline that a receiver can use to compare the received signal against the original non-distorted known signal. The receiver then calculates the distortion effects of the channel and compensates accordingly. Based on the calculations, a determination is then made of the loop's ability to support transmission of the requested configuration's aggregate data rate across the given loop.

Rate adaption refers to the ADSL modem's capability to continually monitor the condition of the line and change its data rate in response to changing line conditions. To achieve the criterion of approximate equal error probability among the sub-channels, an iterative measurement of performance of each sub-channel is performed at the receiver, and instructions are sent back to the transmitter to change the constellation size. This is accomplished by continually evaluating the SNR on the line. The higher the SNR, the higher the data rate that can be accommodated with a given QoS. Successful transmission of ATM over ADSL requires a reliable mechanism for adapting the data flow rate to the transmission medium at start-up, upon initialization following line re-connection, and dynamically with significant changes in line conditions. Modem rate adaption can be static or dynamic. Typically, with static rate adaption SNR is measured at startup and a predetermined margin is subtracted to allow for possible additional noise or signal attenuation. Based on the adjusted SNR——with the margin subtracted——a maximum reliable data rate is then selected. The higher the margin, the lower the data rate, but the more adaptable the system is to changes in line conditions. An often used example of static rate adaptation is the static adjustment of the data rate of a voiceband modem attempting to connect at 28.8 Kbps down to a lower data rate such as 19.2 Kbps or even 14.4 Kbps if the line is too noisy. Static rate adaptation is possible with the T1.413 standard using ATM over ADSL, but it is not explicitly determined/defined.

Dynamic rate adaptation refers to a change in data rate during a steady state while user data is being transmitted. With dynamic rate adaption, the modem continually monitors the SNR. If the margin falls, the data rate is dynamically reduced until the margin returns to its original value. Correspondingly, if the margin increases, the data rate could likewise be increased to bring the margin back to the original level. Dynamic rate adaptation includes not only rate modification, but also rate reapportioning between the fast and interleaved paths of the ADSL modem as well as the corruption of several symbols, and purging the interleaver for the interleaved paths. This purging will result in a loss of bits equal to the interleaver depth, and therefore an effective interruption in the PMD service for a duration equal to the ratio of the interleaved depth to the bit rate. The maximum duration of this interruption is yet to be determined by the ANSI T1E1.4 Committee and the ATM Forum, but it is estimated to be 100 or 125 ms. In the fast path some of the symbols will also be corrupted. The exact handling of this corrupted data during these periods prior to swap has not been specified by the ADSL Forum. However, the rate adaptation process at the ADSL modem may impact the QoS of ATM users, especially for service contracts that are not only time sensitive but also require low error rate. One approach being considered is to fill the interleaver and the deinterleaver with idle cells and transmitting idle cells in ADSL frames which may be corrupted.

Dynamic rate adaptation seems to be the rate adaptation mode of most interest to NS/EP users of critical applications requiring constant bit rate service——e.g., video or data applications with rigorous timing control and performance parameters. Dynamic rate adaption is not currently supported by the T1.413 standard, but it is being discussed for future revisions. In the interim, to ensure uninterrupted service for high priority traffic during emergencies, NS/EP users of ATM over ADSL might consider negotiating a significantly higher than normal margin value even with a corresponding reduction in data rate thus precluding the need for a rate change.

LIST OF REFERENCES

1. Rauschmayer, D., 1998, *ADSL/VDSL: A Practical and Precise Study of Asymmetric Digital Subscriber Lines and Very High-speed Digital Subscriber Lines*, Macmillian Technical Publishing, Indianapolis, IN.

2. ADSL Forum, 1997, *ADSL Forum TR-001: ADSL System Reference Model*, The ADSL Forum, Fremont, CA.

3. ADSL Forum, 1997, *ADSL Forum TR-002: ATM over ADSL Recommendations*, The ADSL Forum, Fremont, CA.

4. ANSI Technical Subcommittee T1E1, 1997, *Draft American National Standard for Telecommunications: Interface Between Networks and Customer Installation - Asymmetric Digital Subscriber Line (ADSL) Metallic Interface*, T1E1.4/97-007, American National Standards Institute, Inc., New York, NY.

5. ANSI Technical Subcommittee T1E1, May 1998, *Standards Project for Interfacing Carrier to Customer Connection of Asymmetrical Digital Subscriber Line (ADSL) Equipment: T1.413 Issue 2*, T1E1.4/98-007R1, American National Standards Institute, Inc., New York, NY.

6. The ATM Forum Technical Committee, April 1998, *Future Work for the RBB Group*, ATM Forum/98-0304, The ATM Forum, Berlin, Germany.

7. Warrier, P., June 1998, *DSP Solutions for Voiceband and ADSL Modems*, Application Report SPAA005, Texas Instruments, Dallas, TX.

8. The ATM Forum Technical Committee, June 1995, *ADSL Access Network Reference Model*, ATM Forum/95-0610, The ATM Forum, Mountain View, CA.

9. The ATM Forum Technical Committee, December 1996, *Report of the December, 1996 RBB/ADSL Forum ATM WG Meeting*, ATM Forum/96-1789, The ATM Forum, Mountain View, CA.

10. The ATM Forum Technical Committee, April 1996, *Traffic Management Specification v4.0*, The ATM Forum, Mountain View, CA.

11. The ATM Forum Technical Committee, July 1997, *Frame Based User-To-Network Interface (FUNI) Specification - Version 2.0*, The ATM Forum, Mountain View, CA.

12. ANSI Technical Subcommittee T1E1, September 1998, *The Essential Merit of Bit Swapping*, T1E1.4/98-318, American National Standards Institute, Inc., New York, NY.

13. The ATM Forum Technical Committee, April 1996, *Traffic Management Specification - Version 4.0*, The ATM Forum, Mountain View, CA.

14. ANSI Technical Subcommittee T1E1, October 1998, *Response to Liaison from SG 15 on Evaluation of ADSL Interference into POTS*, T1E1.4/98-099, American National Standards Institute, Inc., New York, NY.

15. ANSI Technical Subcommittee T1E1, May 1998, *Suggested Management Model for DSL Systems*, T1E1.4/98-138, American National Standards Institute, Inc., Dallas, TX.

16. Network Equipment Technologies, 1998, *Advanced Traffic Management for Multiservice ATM Networks: An NET White Paper*, http://internet.net.com/repository/white_papers/adtmatm_wp/.

17. ANSI Technical Subcommittee T1E1, May 1998, *ADSL OAM: Line and Payload Definitions and Parameters*, T1E1.4/98-141, American National Standards Institute, Inc., Dallas, TX.

18. The ATM Forum Technical Committee, July 1996, *ATM User-Network Interface (UNI) Signalling Specification - Version 4.0*, The ATM Forum, Mountain View, CA.

19. Freeman, R., 1996, *Telecommunications System Engineering - Third Edition*, John Wiley & Sons, Inc., New York, NY.

20. Black, U., 1997, *Emerging Communications Technologies - Second Edition*, Prentice Hall PTR, Upper Saddle River, NJ.

21. Marven, C. and Ewers, G., 1996, *A Simple Approach to Digital Signal Processing*, John Wiley & Sons, Inc., New York, NY.

22. Freeman, R., 1995, *Practical Data Communications*, John Wiley & Sons, Inc., New York, NY.

23. Ziemann, P., undated, *Application Note 53: Testing ADSL Modems - Verification of Analog Interfaces as per ANSI T1.413*, Wandel & Goltermann Communications, Eningen, Germany.

24. Kellogg, September 1998, *Asymmetric Digital Subscriber Line (ADSL): Specific Capabilities of ADSL*, http://www.kellogg.nwu.edu/student/4q/TELCOM/GT_ADSL.htm.

25. Shepard, S., March 1998, "The Promise of ADSL," *Telephony Magazine*, Intertec Publishing

Corp., Overland Park, KS.
26. ATM Forum Technical Committee, January 1994, *About the Need for Cell Loss Correction for Audiovisual Services*, ATM Forum/94-0076, The ATM Forum, Mountain View, CA.

27. ATM Forum Technical Committee, May 1998, *The Home ATM Network*, ATM Forum/94-0076, The ATM Forum, Chicago, IL.

28. ATM Forum Technical Committee, September 1997, *Proposed Text for the ADSL and VDSL Sections of the RBB Baseline Text*, ATM Forum/97-0785, The ATM Forum, Paris, France.

29. The ATM Forum Technical Committee, December 1996, *Report of the December, 1996 RBB/ADSL Forum ATM WG Meeting*, ATM Forum/96-1789, The ATM Forum, Mountain View, CA.

30. The ATM Forum Technical Committee, December 1996, *Comments on ADSL Forum ATM Mode* Document, ATM Forum/96-1634, The ATM Forum, Mountain View, CA.

31. The ATM Forum Technical Committee, June 1996, *ADSL ATM TC Activities Update*, ATM Forum/96-0925, The ATM Forum, Orlando, FL.

32. ATM Forum Technical Committee, February 1997, *ATM over ADSL and VDSL: PMD Specific Attributes that Affect the Network*, ATM Forum/97-0156, The ATM Forum, Mountain View, CA.

33. Saltzberg, B., et. al., November 1998, "Comparison of Single-Carrier and Multitone Digital Modulation for ADSL Applications," *IEEE Communications Magazine*, New York, NY.

ACRONYMS

A/D	Analog-to-Digital
ADSL	Asymmetric Digital Subscriber Line
ANSI	American National Standards Institute
AOC	ADSL Overhead Channel
ATM	Asynchronous Transfer Mode
ATU	ADSL Transceiver Unit
ATU-C	ATU-Central
ATU-R	ATU-Remote
B-ISDN	Broadband-ISDN
BRI	Basic Rate Interface
CAP	Carrier Amplitude/Phase
CDV	Cell Delay Variation
CER	Cell Error Ratio
CLR	Cell Loss Ratio
CRC	Cyclic Redundancy Check
CTD	Cell Transfer Delay
D/A	Digital-to-Analog
DMT	Discrete Multi-Tone
DSL	Digital Subscriber Line
DSP	Digital Signal Processing
FDM	Frequency Division Multiplexing
FEC	Forward Error Correction
HDSL	High data rate DSL
HDTV	High Definition Television
HDLC	High-Level Data Link Control
HEC	Header Error Check
Hz	Hertz
ISDN	Integrated Services Digital Network
Kbps	Kilobits Per Second
kHz	Kilohertz
LAN	Local Area Network

Mbps	Megabits Per Second
Mhz	Megahertz
ms	Millisecond
N6	Technology and Standards Division
NCS	National Communications System
NPC	Network Parameter Control
NS/EP	National Security and Emergency Preparedness
OAM	Operation, Administration, and Maintenance
OMNCS	Office of the Manager, National Communications System
PC	Personal Computer
PMD	Physical Medium Dependent
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quaternary Phase Shift Keying
SDH	Synchronous Digital Hierarchy
SNR	Signal-to-Noise Ratio
SONET	Synchronous Optical Network
STM	Synchronous Transfer Mode
TASI	Time Assigned Speech Interpolation
TC	Transmission Convergence
TC-F	TC-Fast
TC-I	TC-Interleaved
TDM	Time Division Multiplexing
TCM	Trellis-Coded Modulation
UPC	Usage Parameter Control
VC	Virtual Channel
VCI	Virtual Channel Indicator
VDSL	Very high data rate DSL
VP	Virtual Path
VPI	Virtual Path Indicator
WAN	Wide Area Network
WWW	World Wide Web