Video on Phone Lines: Technology and Applications

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Invited Paper

This paper reviews the telephone loop plant characteristics, current DSL (digital subscriber line) technologies, recent efforts in video coding standards, and the interrelationship between DSL technologies and visual communications over subscriber lines. In overview of the loop plant characteristics we examine its physical makeup and transmission properties, where for the latter we discuss frequency and time responses of wire-pair lines and the impairments of echo, crosstalk, impulse noise, and radio frequency interference. We trace the historical development of various DSL technologies and comment on possible future evolution. Transmission technologies used in the ISDN basic-access DSL, the high bit-rate DSL, and the asymmetric DSL are portrayed. And the issue of spectrum compatibility among different transmission systems is explained. Several important video coding standards are briefly described, including ITU-T's H.261 and ISO's JPEG and MPEG series, which are either completed or emerging. The synergistic relationship between these standards and the DSL technologies is elucidated. As a result, DSL technologies provide the potential of delivering certain broadband services well in advance of direct fiber access for telephone subscribers.

I. Introduction and Historical Review

Just a few years after inventing the telephone, Alexander Graham Bell patented twisted pair wiring in 1881. Since then, the worldwide subscriber loop plant has grown to become one of the world's valuable technological assets. High performance transmission over subscriber loops has historically required considerable engineering in order to accommodate the wide range of loop makeups encountered in the distribution plant. New technologies have increased the potential to exploit the digital information carrying capacity of copper pairs, originally placed to provide narrowband voice service, to the point where certain broadband services can be provided to customers well in advance of direct fiber access. These technologies are embodied in a number of new digital transmission systems for the access network generically referred to as the digital subscriber line (DSL).

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This paper reviews the loop plant characteristics, current digital transmission technologies for subscriber line applications, recent efforts in video coding standards, and the interrelationship between transmission technologies and visual communications over subscriber lines. Our emphasis is on pair-wire lines. Notwithstanding, it is expected that the fiber/coax architecture will play an increasingly significant part in telephone companies' networks. The transmission environment in the coaxial section of a fiber/coax network is very different from that of the pair-wire loop plant. Nevertheless, the fundamental transmission techniques and many of the envisioned video services should remain the same. In this paper, the present section provides a general introduction and historical review of the subject matter. Section II expounds the loop plant characteristics pertaining to digital transmission. Section III describes the various DSL systems, existing and envisioned, as well as the associated issue of spectrum compatibility. Section IV presents an overview of several important visual communications standards. And finally, Section V is the conclusion.

Telephone transmission facilities can be broadly classified into three segments: subscriber lines (or loops), switches, and interoffice lines. Both the subscriber and the interoffice "lines" can be wired or wireless. Digitalization of the transmission facilities began several decades ago, beginning with the longer interoffice connections. As time progressed, shorter links were also economically proven to go digital. Digitalization of the switches followed suit, as did the digital loop carriers which provide pair gain (concentration of telephone traffic on multiple wire pairs onto fewer pairs or even fiber) for the feeder section of the loop plant (section of the plant that connects to the central office). (In referring to digital switching, we mean primarily the switching of digital *signals* rather than digital, stored-program *control* of the switches where the signal paths could be either analog or digital. The latter predates the former. Nevertheless, both appeared after debut of long-haul digital transmission.) Now we have come to a point where massive digitalization of the complete loop segment is being touted.

Separate digitalization of the different transmission segments have generated some interesting issues regarding interworking among different digital and analog facilities, for instance, the need to consider analog links' echo effects in determining the allowable delay in a (digital) speech codec and the need to accommodate both speech and voice-band data in the design of a "speech coding" algorithm for interoffice transmission application [1]–[4]. These concerns will remain for some time. For high-speed subscriber line transmission, it is sometimes the lack of compatible digital switches that one has to grapple with.

Although the copper subscriber lines have been engineered primarily for narrowband voice service, the cables themselves often can support transmission at much wider bandwidths. The limitation resides mainly in the equipment on both sides of a line. For example, the telephone switch on the CO (central office) side assumes that the incoming and outgoing signals are bandlimited to under 4 kHz. In addition, for long loops (typically over 18 kft) a standard practice is to add loading coils (inductors) along the line to equalize the voice-band response of the cable [5]. The use of loading coils significantly increases the cable's attenuation above 4 kHz.

Over the last four decades, there has been continual effort aimed at utilizing this limited bandwidth for data transmission. The result is a flow of voice-band data modems capable of communicating at ever-increasing rates, from low hundreds of bits per second in the early days to well in excess of 20 kb/s today [6]–[10]. The progress has been facilitated by both the advances in transmission technologies employed in modems (e.g., synchronization, equalization, echo cancellation, and coded modulation) and the reduced noise levels in physical transmission facilities. Much of today's voice-band data traffic is generated by fax transmissions.

Successful or not, in the last few years we have also seen several commercial offerings, aimed at the general populace, of picture telephone sets for use over voice-grade links. The picture transmission capability is built on top of the voice-band modem technology. Few of these sets employ sophisticated video coding techniques. Most were meant only for nonreal time transmission of still images. Interest in very low bit-rate video coding is quite strong, however. And both ITU-T (International Telecommunications Union—Telecommunications Sector, formerly CCITT) and ISO's MPEG have recently established standards activities for digital video transmission over voice links.

To exploit a subscriber line's capability beyond 4 kHz, equipment on both ends of the line must be changed. Since loading severely attenuates a line's response above 4 kHz, high-speed digital subscriber line transmission invariably requires the absence of loading coils. Studies on subscriber lines' ultimate (Shannon) capacities under several conditions have been reported by many [11]–[15]. Basically, they are a function of line length and composition (wire gauges, presence and locations of bridged taps, etc.). Derivation of Shannon capacities assumes infinitely complex transceivers which are practically unrealizable. However, practical transceivers can usually realize a performance within several dB of the ultimate limit and hence such results are still useful in guiding system designs.

Since the loop plant is made up of lines of a great variety of lengths and compositions and hence widely differing capacities, a decision must be made as to what transmission rates to engineer for and each for which loop population. Such a decision need be arrived at considering technological, social, and economical factors, and with intuition. Over the years, several transmission rates have been selected for technological study and standards deliberation. Of them the most widely known is perhaps the 144 kb/s (symmetric bidirectional) for ISDN basic-rate access [16]–[18].

In retrospect, the Digital Data System (DDS) can be viewed as a forerunner of today's DSL [19]–[21]. DDS offers data rates at 2.4, 4.8, 9.6, and 56 kb/s over a four-wire loop. One pair is used for each direction of transmission. It was planned not merely as a loop transmission system but also as a national network [20], [22]. Capitalizing on the availability of DDS, several manufacturers offered very low bit rate (56 kb/s) video conferencing equipment. Not all of them fared well, however. But the stage became progressively set for the emergence of ISDN DSL and the present array of international video coding standards.

In the United States, the basic-access DSL standards were set for nearly ubiquitous applicability to nonloaded loops as long as 18 kft [23], [25]. A further 16 kb/s is added to the 144 kb/s payload to serve network administrative functions, making the line rate 160 kb/s. Shorter loops can support higher transmission rates. As a result, the high-bit-rate DSL (HDSL) was proposed [25], [26]. HDSL provides DS1 access using two pairs, each operating at 784 kb/s, over Carrier Serving Area ranges (approximately 9 to 12 kft) [27]. It does not require the costly pair selection, removal of bridged taps, or use of repeaters every 3 to 6 kft as T1 does [25]. And it can be used as a T1 replacement. To date, there have been several manufacturers selling HDSL line cards. These higher rates can support the transmission of higher-quality video for videophone, videoconference, and other applications.

We also note that, since fixed-rate access inevitably underutilizes the loop plant's potential unless we tailor the transmission rate for each individual line, some have explored the idea of adaptive transceivers which can dynamically find the maximum rate a loop will support [28].

Stimulated by many interactive video services which are asymmetric in nature, the asymmetric DSL (ADSL) [29] has been studied for technical feasibility, with an (original) objective of transmitting 1.5 Mb/s in one direction, from the network to the customer, in addition to a low-speed bidirectional control channel and a conventional analog voice channel, all over one nonloaded loop pair. Combined with recent progress in digital video compression (in particular the ISO MPEG1 standard), such data rates should be adequate for several important classes of video services. As such, ADSL has received much attention as a potential low cost access technology for delivering one channel of "VCR quality" compressed digital video to residences [30].

VLSI advances are making DSL's practical at rates well above 1.5 Mb/s. On the other hand, the need for multiple simultaneous channels and for higher quality video in a competitive market is recognized. Therefore, there has been interest in higher rate asymmetric structures for providing higher quality compressed video signals and/or multiple channels to the residence recently. A target was set to reach at least 6 kft at a transmission rate of about 6 Mb/s. In mid-1993 the ANSI T1E1 standards subcommittee selected the discrete multitone (DMT) modulation technique as the standard transmission format for the ADSL technology with bit-rate up to 6 Mb/s. As a whole, ADSL has been viewed as a technology which can fill up the void prior to the arrival of broadband facilities. At least one field trial with standard and nonstandard ADSL units has been conducted. More may be on the way.

Deployment of DS3 private line services is beginning to accelerate as large business customers migrate from multiple DS1 private lines to a single DS3. As demand for DS3 service grows, a market for fractional DS3 is expected to emerge. Telephone companies may therefore be more willing to define a new interface based on very high-speed DSL (VHDSL) technology (>10 Mb/s) in the drop portion of the existing loop plant. This would allow customer wiring to be included as part of the access link, with the VHDSL termination residing in CPE (customer premise equipment). In particular, telephone companies may be motivated to define a new interface for high speed data access. Unlike private lines which typically transport a considerable amount of voice service, the strategy for introducing new high speed data services such as SMDS and ATM broadband access has been to encourage termination of the network access link by CPE in the customer's location. Thus VHDSL could become an efficient means to quickly and cost effectively provision these services at rates between DS1 and OC1 (51 Mb/s).

Scientists have recognized for many years that transmission of high bit rates is possible over short twisted-pair lines [31]. The rapid deployment of unshielded twisted-pair (UTP) LAN technologies operating at 4, 10, and 16 Mb/s using existing customer premises wiring, and the development of copper technologies for CDDI (FDDI over copper) and SONET premises applications at 100 and 155 Mb/s, respectively, demonstrates the trend toward and feasibility of higher rate technologies over short lengths of twisted-pair cable. Coupling the rapid advances in video compression and ATM/SONET standards with the recent development in VHDSL technologies, there has been tremendous interests of delivering ATM to the desktop and to the house at speeds in excess of 10 Mb/s over outside plant drop wiring.

As we move forward into the age of optical fiber, we could view the copper loop plant as an asset to aid evolution, rather than as a liability preventing progress. The embedded copper will continue to make up a significant portion of the local loop plant into the next century, even under the most aggressive fiber deployment scenarios. With advances in DSL, the telephone companies' ownership of the embedded plant can have positive, synergistic consequences relating to the deployment of fiber-in-the-loop (FITL) systems and the provision of visual communication services.

II. Characteristics of the Loop Plant

A. Physical Characteristics

The local exchange plant is a resource which began growing over a century ago and has provided basic telephone service ever since. It now represents one of the major components of capital investment in the public network. It is reported, under the FCC established Uniform System of Accounts, as comprising over 25% of the Regional Bell Companies' assets, or over $50 billion [27].

Initially cables were comprised of individual copper conductors wrapped with paper for insulation. The insulated conductors are twisted together into pairs and pairs are grouped into binder groups of typically 25 to 50 pairs. Binder groups are combined into cables, ranging in sizes from 100 pairs to 2500 pairs. These cables typically emanate from a CO in underground conduits or ducts. As the cables branch out to businesses and residences, they become smaller and aerial suspension becomes more common. Direct burial, without supporting ducts, is also common, particularly in suburban and rural areas.

Paper insulated cables are economic and perform well if kept dry. Avoiding moisture in cables is, however, difficult in underground applications. Loop maintenance centers have long observed high problem rates after periods of heavy rain, as paper insulated cables take on water through inevitable cracks in sheaths and splice cases. In response to maintenance problems due to wet cable, pressurization systems were deployed which constantly force air through the cable, preventing water from entering. In the 1960's, plastic insulation was introduced yielding a polyethylene insulated conductor or PIC cable. Initial hopes were that the use of PIC cable would eliminate water problems, but tiny pinholes in the insulation could set up an electrolysis effect that eroded the copper. Therefore the use of pressurization systems continued. And the use of paper insulation endured in the network long after the introduction of PIC cable. When PIC cable was introduced in the 1960's, there were limitations on the mechanical pulling stresses that could be applied during installation. This limited the pulling of PIC cables in conduits, and paper insulated cables were still commonly used until the early 1980's. To eliminate the need for pressurization systems, filled cables which fill the gaps between pairs with a nonconductive jelly were introduced in the 1970's. Most new cables placed today are filled PIC.

The loop plant widely used today for voice (and data modem) access consists of individual copper wire-pair

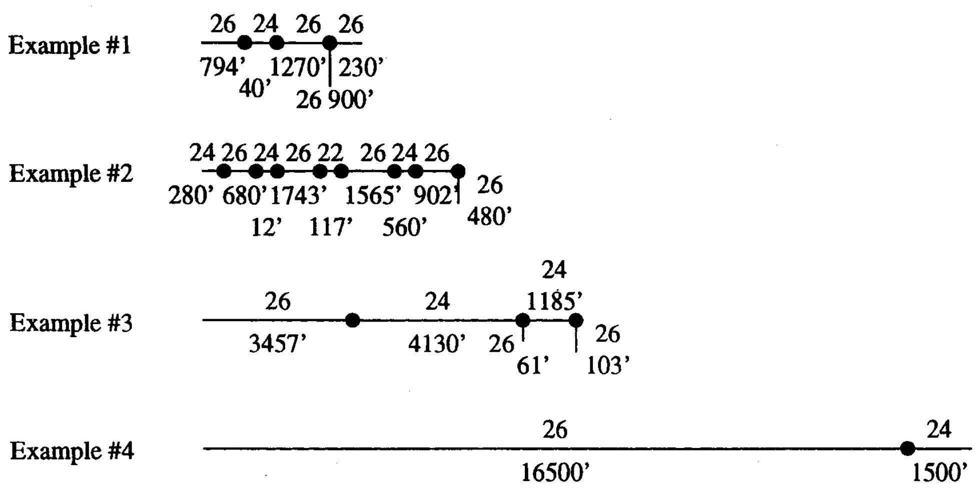

Fig. 1. Typical loop makeups.

connections from a CO (or remote electronics site) to customer locations. Of the over 100 million pairs in service in the Regional Bell Companies today, about 25% are equipped with loading inductors and are not directly usable as new DSL's. The majority of unloaded loop plant is less than 18 kft in length, so this is a good working limit for ubiquitous residential service. Loops which are in Carrier Serving Areas (CSA) are less than roughly 9–12 kft. This is important because performance improves significantly if the application of new DSL technology can be limited to these shorter lengths. For longer lengths and higher data rates, losses of more than 60 dB need to be accommodated making low-noise receivers an important issue.

The path leading to the customer is typically not just a simple pair of wires. A copper, between the CO and the customer, is usually made up of pairs from sections of several cables. Several typical examples are shown in Fig. 1, where the numbers give the wire gauge and length of each section. As shown, a loop can include several gauge changes and bridged taps. Bridged taps are open circuit pairs, either intentionally placed along a main cable route in anticipation of possible service demand at another location, or resulting from past service disconnections and rearrangements. Gauge changes result in impedance discontinuities which tend to distort and reflect high speed digital pulses. Bridged taps also echo signals. The net result is that received signal pulses must not only be amplified but significant reshaping must be done to reverse the damage done by transmission. The exact form of reshaping is unique to every wire pair. Fortunately, we have available to us today special purpose digital signal processors which can not only reshape and separate pulses which have become tangled together, but can automatically adapt in real time to each individual wire pair. It is this new technology that has given wire pairs a new lease on life. Not only can we reverse transmission irregularities which were intolerable for the old T1 systems, but we can do it without any manual craft intervention.

In the 1970s, efforts began to improve bandwidth of the loop plant. The philosophy used was one of upgrading loops whenever maintenance or rehabilitation became necessary, and of placing new facilities in growth areas according to latest administration procedures. An important aspect of this program has been the introduction of the Carrier Serving Area or CSA administration guidelines. These guidelines are aimed at shortening the distribution of loop lengths, eliminating the longer loops and eliminating the use of load coils in order to position the loop plant to provide new digital services at speeds in the region of 64 kb/s [32]. CSA guidelines call for limits of 9 kft of 26 gauge cable and 12 kft of 24 gauge cable. The guidelines also restrict the working length of loops with bridged taps and limit bridged taps and excessive mixing of gauges.

Administration of the loop plant using the CSA guidelines began in the early 1980's, particularly with increasing deployment of digital loop carrier systems or DLC. DLC fueled a trend to place electronics in the distribution network. These electronics are located between the CO and the customer, requiring some type of environmental enclosure such as a hut, a vault or a cabinet. Among the driving forces behind DLC deployment are the desire to use the plant more efficiently and to shorten the lengths of voice frequency copper loops, making the loop plant more amenable to new digital services. In addition, recent DLC systems use fiber in the feeder network, those facilities from the CO to the remote electronics site.

Recent Regional Holding Company survey data show that, despite a decade of strong growth in DLC, penetration on average has yet to exceed 10% of the loops served. The data also show that, although fiber has significantly penetrated the feeder network, penetration all the way to the customer is still relatively low, on the order of a few percent of the total network access lines, with almost all of it concentrated on large business customers. Although the Regional Companies are aggressively moving toward fiber deployment in the distribution network and it is hoped that FITL installation will bring widespread broadband applications to residential customers toward the end of this decade, complete penetration may take several decades, particularly to smaller businesses and residences. Thus there is motivation to use the embedded base of copper facilities effectively during the transition from copper to fiber.

Indeed, according to a recent report, the total investment of copper lines for all Bell Operating Companies spending was 22% (about \$3.15 B) in 1991. Compared with fiber installation, it still was three times more in total spending. Therefore, while an evolution of copper plant to fiber plant seems inevitable, and fiber is the media of the next century, any increasing demand for wider bandwidth services through loop plant will need to be satisfied by increasing the efficiency of the existing ubiquitous copper loop plant. Even as fiber systems are deployed, copper wire will still be used, in many cases, for the last drop to the terminal. There is an urgent need to explore new technologies for immediate applications to meet the requirements of those customers who have wideband service needs until such time as it is replaced by fiber.

B. Transmission Characteristics

The voice-band modems and the ISDN basic-access DSL operate at frequencies where the characteristics of the loops are fairly well known. Models have been developed that accurately predict system performance at their rates. HDSL and ADSL systems operate at frequencies that are considerably greater than those used in the basic-access DSL. The system models that have been developed for HDSL and ADSL have proven to be quite useful. However, further work is required to refine these models, especially in channel characteristics and crosstalk predictions, and impulse noise characterization. As fiber penetrates the feeder network, ADSL at 6 Mb/s over CSA's will be possible, as will the VHDSL. The operating frequencies will be considerably greater than those used in HDSL and DS1-rate ADSL services. Work has just started to develop an understanding of the transmission characteristics and impairments associated with transmission at such high data rates. Below we summarize what are known of the loop transmission characteristics.

Generally speaking, the frequency responses of typical loops drop off rapidly at higher frequencies. For voice service, equalizers can be used to flatten the channel. If we wish to transmit data, we also need to be concerned with the phase response. A copper loop provides a dispersive communications channel; the phase response is not strictly linear with frequency. If a pulse is transmitted down a loop, the nonflat magnitude and nonlinear phase responses will cause the pulse to spread out in time. In a pulse train, the smeared out pulses will overlap with each other, a phenomenon referred to as intersymbol interference or ISI. With bridged taps, the ISI problem can be more severe as the bridged taps may cause strong reflections due to impedance mismatch at where they join the main line and at their open-circuited ends, leading to "ghost pulses" much like the ghosting problem in analog TV reception. The pulses are also considerably attenuated as they travel down the cable. The equalizer and receiver timing recovery circuits are faced with the task of compensating for these degradations.

From transmission line theory, it is well known that a wire-pair's frequency response is given by

$$H(f) = e^{-\gamma(f)l} \tag{1}$$

where l is the length of the line and

$$\gamma(f) = \sqrt{(R + j2\pi f L)(G + j2\pi f C)} \tag{2}$$

with R, L, G, and C being the line's series resistance, series inductance, shunt conductance, and shunt capacitance per unit length. The attenuation therefore increases exponentially with line length. Listing of the so-called primary constants R, L, G, and C for some commonly encountered cables can be found in [23]. A brief discussion of the properties of $H(f)$ can be found in [11]. In summary, at low frequencies (<10 kHz), the primary constants can be considered frequency-independent to yield an approximate magnitude response $|H(f)| = \exp(-k\sqrt{f}l)$ for some k. At high frequencies (>200 kHz), the skin effect [33], [34] becomes prominent to make R proportional to $\sqrt{f}$, again leading to an exponential dependence of $|H(f)|$ on $\sqrt{f}$, albeit with a different constant k. Also, for large frequencies (>200 kHz), the envelope delay (i.e., $d[\arg H(f)]/d\omega$) is almost independent of frequency. But for small frequencies the dependency is strong. Various loop impulse responses examples have been published in many papers. Examples showing that reflections from bridged taps can give rise to strong "ghost pulses" can be found in [35] and [36]. A number of other impulse response examples are given in [37]–[41]. Crude estimate of the range of a DSL transmission scheme can sometimes be obtained by considering the cable attenuation at mid-band of the digital signals and the overall noise power [45, Discussion].

Due to the manufacturing process, individual loop responses may exhibit periodic and random variations about the nominal values at frequencies above approximately 500 kHz [42]–[44]. Effects of such variations on signal transmission are yet to be fully characterized.

In the access network, there is strong economic motivation to minimize the number of pairs needed to provide service. This is why two-wire transmission has been used for voice service, where transmission occurs simultaneously in both directions over the same copper loop. This is accomplished with a hybrid balance network, which separates the two directions of transmission using a four-port balanced bridge. If the impedances are balanced, a two-wire to four-wire conversion is effected. The same principle is used in data transmission, where pulse trains are simultaneously transmitted in both directions over a single loop in a full duplex fashion. Early voice-service hybrids use transform-

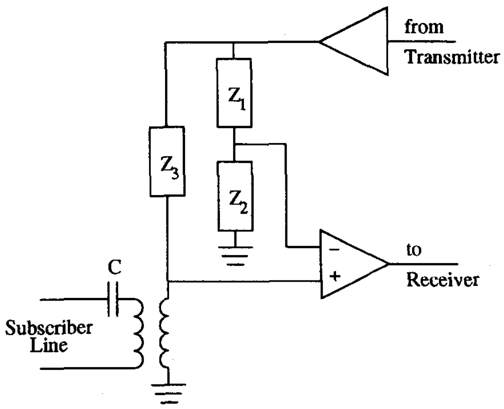

Fig. 2. An example hybrid network.

ers for impedance coupling. Today's electronic hybrids employ active devices and a schematic example is shown in Fig. 2. The transformer facilitates balanced transmission and rejection of common-mode interference. The capacitor on the loop side of the transformer provides a means for network-supplied power: Batteries may be connected to the CO end of the line and leads may be pulled from the two ends of the capacitor to power the electronics. By the superposition principle, the receiving amplifier output contains a sum of the received signal from the line and the leak-through signal from the local transmitter. Let Z_L denote the input impedance of the line as seen through the transformer and the capacitor. Then the leak is zero when $Z_1/Z_2 = Z_3/Z_L$. But such match can never be realized perfectly as the Z_i (i= 1,2,3) are of necessity made of lumped-circuit elements while the loop is a distributed-circuit system. Even if the desired match were possible, there remains the problem of adjusting the Z_i for each individual subscriber line. In summary, some leak of the transmit energy into the receive path is inevitable, resulting in the problem of *echo*. Echo is commonly dealt with using an adaptive echo canceller.

Aside from ISI and echo, there are several noise sources that must be contended with. Within a binder group of a cable there are usually several active pairs. Since these pairs are in close physical proximity over long distances, coupling takes place and the pairs *crosstalk* into each other. There are several types of crosstalk to consider, but for full-duplex transmission a reliable engineering rule is to consider near-end crosstalk or NEXT as one of the major limiting impairments. NEXT comes about when "hot" transmitters, transmitting toward the far end, couple into weak received signals coming from the far end. The situation is illustrated in Fig. 3. When the NEXT is from identical transmission systems as the one being disturbed, it is known as self-NEXT. Also illustrated in Fig. 3 is the far-end crosstalk, or FEXT. Unlike NEXT, FEXT suffers the same line attenuation as the disturbed signal. As a result, self-FEXT is usually a much weaker concern than self-NEXT.

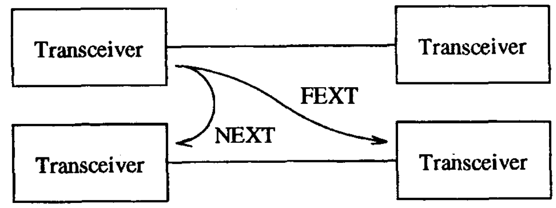

Fig. 3. Two important crosstalk mechanisms.

Due to crosstalk, spectrum compatibility is a major issue in the engineering of a transmission system which will share the same cable with other systems. The strength of crosstalk coupling depends on cable construction. In the US, individually twisted pairs have been used [11]. In some other areas of the world, quad cables are used in which two pairs may be twisted together as a unit. The resulting crosstalk coupling behavior can be significantly different from that of twisted-pair cables [46], [47].

The crosstalk phenomenon has been subject to extensive study. Since some of the earliest work by G. A. Campbell [48], there have been numerous reports on its circuit models, simplified linear system models, probabilistic and statistical characterizations, and engineering implications. An early application concerns open-wire transmission [49]. A sample of more recent work can be found in [50] and the references therein. In circuit-theoretic notion, crosstalk arises from capacitive and inductive unbalance between wire-pairs. Such unbalance has been verified as approximately statistically uncorrelated over the length of the cable [51, Appendix A]. Its values are normally also considered to be independent of frequency. These properties facilitate simple statistical characterizations of crosstalk strength in terms of a few parameters. In addition, the variance of the capacitance unbalance was also found to be related to twist pitches [52, Section V].

The mean power transfer function of NEXT is known to be approximately given by $K_n f^{3/2}$ and that of FEXT by $K_f l f^2$, where f is frequency, l is line length, and K_n and K_f are cable-dependent constants [51], [53], [47]. The NEXT power transfer function does not show an l-dependence as the FEXT because most of the NEXT power comes from crosstalk couplings that occur near the disturbed transceiver. Crosstalk couplings that occur further out suffer substantially more attenuation as the coupled energy travels back to the disturbed transceiver. As a result, the NEXT power transfer function does not depend on l (for long enough lines). Engineering in the crosstalk-dominant noise environment cannot be based on the mean, however, since that would lead to many inoperable systems plagued by excessive crosstalk noise. Rather, it is often targeted at the 1% worst case, meaning that with 99% probability the crosstalk noise power would be under the designed level. The Bellcore 1% worst-case NEXT model, derived via computer simulation, that has been employed in a number of standards studies can be found in [11] and [54]. The model shows some deviation from the $f^{3/2}$ frequency dependence. Some further characterization of the NEXT can be found in [55]. For 1% worst-case FEXT [55, Appendix B] gives $K_f = 8 \times 10^{-20}$ for the case of 49 disturbers and $K_f = 8 \times 10^{-21}$ for the case of 1 disturber (with l in feet and f in Hz).

Crosstalk is treated as stationary in most engineering analyses. However, digital transmission systems can generate cyclostationary crosstalk due to the cyclostationarity in transmitted signals. This can be handled by allowing some margin in system design for its potentially adverse effects. Some have taken a closer look at the implications of such cyclostationarity. We refer to [56]–[59] for further details. Adaptive equalizers can mitigate the effects of crosstalk noise (or for that matter, any noise).

A class of noises referred to as *impulse noise* are created by several mechanisms in which transients are induced in a loop by nearby sources such as dial pulses, electromechanical switches in the office, current surges in heavy equipment such as elevators, and other mechanisms. As the name implies, impulse noise tends to be sporadic with short pulses of energy, on the order of microseconds. If the magnitude of the impulse is large enough, it can cause errors in data transmission. Effects of impulse noise can be mitigated by error-corrective coding. In the case of multicarrier modulation such as DMT, the block processing nature makes it inherently more robust to impulse noise disturbance than single-carrier modulation [60].

In addition to crosstalk and impulse noise, telecommunication equipment, by nature of its application in the telecommunication network, may be exposed to other sources of electromagnetic energy. One possible impairment due to electromagnetic interference effect on many high-bit-rate transmission on copper lines needs to be concerned. Signals from high-bit-rate services on unshielded drop wires and inside wires may fall within the frequency bands assigned to aeronautical safety communication channel, maritime service or radio navigation such as Loran, and commercial AM broadcast. The concern is radio frequency interference [61] to these licensed services due to electromagnetic emissions from any high-bit-rate transmission system. Also this kind of system must have the immunity to conducted radio frequency interference that is present at the drop wire and inside wires.

III. Digital Subscriber Line Transmission Systems

We now briefly describe the transmission schemes, used or envisioned, for basic-access DSL, HDSL, ADSL, VHDSL, and inside-building high-speed digital networks, and touch on the issue of spectrum compatibility.

Historically, high-speed transmission over wire pairs has required considerable special engineering. T1 carrier systems, which have been used in the interoffice plant for many years, require detailed design for proper operation [51]. Separate and carefully separated pairs are used for each direction of transmission. Bridged taps must be disconnected. Repeaters are needed at approximately 6000 ft intervals. All these imply high provisioning and maintenance costs. In addition, provisioning delays on the order of several days to several weeks are typical. Therefore, although such systems are often used to provide DS1 service to business customers, they are really not an appropriate vehicle for broadly deployed residential services. DDS comes closer to being a viable approach in this regard [20], [21]. Though transmitting at much lower rates, it allows bridged taps if not excessive. Still, it requires two wire-pairs for bidirectional transmission. CSDC (circuit-switched digital capability) [62] was later conceived which would use two-wire loops. Nevertheless, as in the case of DDS, the attainable range leaves something to be desired.

Technologically, today's DSL provides a dramatic improvement for provisioning digital data transport in the access network. DSL's use self-adaptive digital filters to automatically and continuously adjust themselves to the echo and ISI characteristics of the copper loop, and so mitigate effects due to impedance mismatch in hybrids, changes in loop lengths and loop makeups, gauge changes, bridged taps, and changing ambient temperatures. Furthermore, these adaptive digital filters also mitigate the effects of some of the other known channel impairments including crosstalk noise from other cables in the same binder group, and impulse noise. This eliminates the need for manual craft adjustments and ensures continued high performance operation. It is not our intention to introduce the operating principles of these adaptive filters in the present review. The interested reader may consult some of the references cited in the following discussion and the vast amount of literature on adaptive filtering, echo cancellation, and equalization.

Due to continuing advances in VLSI, DSL signal processing hardware can now be integrated into a single or few economic devices. These devices are now feasible over a wide range of bit rates, allowing the information carrying capacity of subscriber loops to be more fully exploited.

A. ISDN Basic-Access DSL

The earliest view of ISDN was to extend 64 kb/s channels and common-channel signaling to the user to achieve cost and performance benefits [17]. In the United States at least, the basic-access DSL can be regarded as descended from DDS and the subsequent CSDC. At one time 80 kb/s was considered a potential DSL transmission rate. Later the standard 144 kb/s, composed of two B (bearer) channels at 64 kb/s and one D (delta) channel at 16 kb/s, was adopted. A goal was set to cover 98% of nonloaded loops without the need for bridged-tap removal or pair selection.

Two major techniques that have been considered to achieve the desired full-duplex transmission are time-compression multiplexing (TCM) and echo cancellation (EC). EC attaches an echo canceller across the hybrid to estimate the echo path response and subtract out the echo. TCM makes the transceivers at opposite ends of a line transmit alternately. The echo problem is thus avoided since a transceiver does not transmit and receive at the

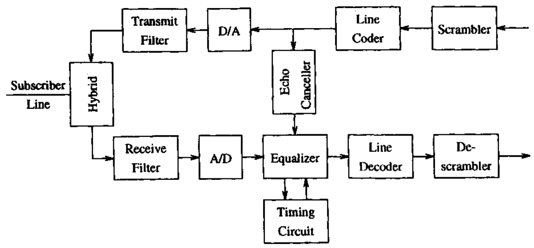

Fig. 4. EC-based DSL transceiver structure.

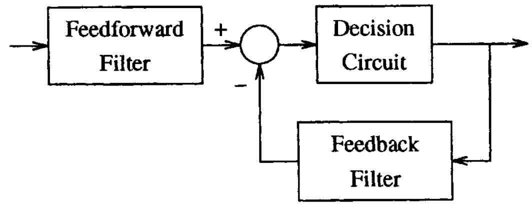

Fig. 5. DFE structure.

same time. If we further time-synchronize the transmission of all the transceivers at each end of a cable, then the problem of NEXT is also eliminated [46]. However, by having to transmit at a higher rate (which is greater than twice the nominal data rate since some guard time need be allowed for transmission delay and the settling down of the channel's response tail), TCM suffers from the greater line attenuation at higher frequencies. Whether one technique is better than the other depends on the loop plant's transmission characteristics. In the US, EC was selected to be the standard after much thought [23]. Elsewhere a variant of EC has also been proposed [63], [64]. In it the transmitted signal is somewhat time-compressed to leave a gap between two successive signal blocks. And the time gaps for two opposite directions of transmission are offset so that there are gaps where only one of the two opposite transceivers is actively transmitting. This transmitting transceiver can then use these time gaps for easier adaptation of its echo canceller. (And the nontransmitting transceiver can also use these gaps for adaptation of its equalizer and timing recovery circuit.)

An EC-based transceiver structure is shown in Fig. 4. References [65]–[67] present some design examples. We briefly describe the transceiver components below. Additional explanation can be found in [68] and the above references. For more detailed specifications in the ANSI standard, see [23] and [54].

The scrambler randomizes the data stream to avoid sending periodic patterns which may lead to strong power spectral lines in the transmitted signal so as to result in severe crosstalk interference. It also causes frequent transitions in data values, aiding timing recovery. The line code chosen for the US standard is simple 2B1Q (2 binary symbols to 1 quaternary symbol) or four-level PAM. A host of other line codes were also considered in the development of the standard, including various binary, ternary, and pseudo-ternary codes. We refer to [45] and [68] for further details. Technically, the 2B1Q code was selected for compression of signal bandwidth for the benefits of less attenuation and less crosstalk.

Depending on line attenuation and echo strength, the echo canceller may need to achieve 50–70 dB of echo reduction. Hence even slight nonlinearity in the echo path may pose a problem for a conventional linear echo canceller. Such nonlinearity may occur in D/A, A/D, and the analog filters. In addition, timing jitter may also raise a concern. Thus nonlinear echo cancellation and jitter-compensation canceller may need to be employed. It is also typical to include a single-pole IIR section in the echo canceller to deal with long tails in echo paths' impulse responses, in addition to a usual FIR section which covers the first few tens symbol periods of such responses.

The equalizer in Fig. 4 is assumed to be digital. In some designs, there may be an analog section for compensation of the $\sqrt{f}$-exponential line attentutation between the receive filter and the A/D. Such compensation, as a part of the overall equalization task, can be accomplished using a digital equalizer. Whether to carry it out in the analog domain or lump it into the function of the digital equalizer is a matter of design choice. A commonly employed form of the digital equalizer is the DFE (decision-feedback equalizer) shown in Fig. 5. The feedforward filter is an adaptive FIR filter. The feedback filter may include a one-pole IIR section, to deal with possible long channel response tails, in addition to an adaptive FIR filter. Length of the FIR feedback filter is normally in the range of a few tens symbol periods while that of the feedforward filter is very short, possibly a single-tap AGC. The echo replica (from the echo canceller) may be subtracted either before the feedforward filter or after it.

The two B channels in basic access can be used separately or together for transmission of video signal with associated audio. Allowing, say, 16 kb/s for audio and some capacity for network-related functions, the available video bandwidth is around 45 and 110 kb/s, respectively. This entails a compression ratio of roughly 270 and 110, respectively, for CIF-format video (defined in next section) at 10 frames/s and a fourth of it for QCIF-format video at 10 frames/s. The resulting quality is usable for videotelephone and some videoconferencing applications.

B. HDSL

As the above ANSI DSL standard was nearing completion, a project for HDSL study was initiated in the T1E1.4 working group [25], [26]. Based on extrapolation of basic-access DSL results, simple back-of-the-envelope kind of calculations indicated that the present technology appeared able to support full-duplex transmission at half the DS1 rate over a single twisted-pair within CSA range [26], [45]. This is rather amazing in view that only a few years earlier the

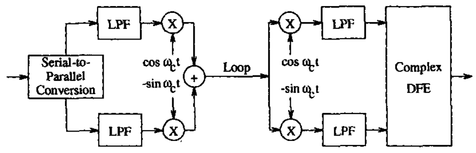

Fig. 6. QAM transmitter-receiver pair.

CSA was considered a plant division for 64-kb/s service. But, of course, transmission capability was not the sole factor leading to the former rate choice. A major reason was the availability of digital switches which could switch 64-kb/s data. For HDSL, the switching issue has also been thought about [69].

The ensuing standards study did not conclude with a standard, however. It was agreed rather that the study served to promote the technology. With HDSL, full-duplex DS1-rate transmission can be provided over two copper pairs (the so-called dual-duplex configuration) within a CSA without the need for repeaters and line conditioning associated with T1 systems. In fact, the primary use of HDSL today is as a cost-effective T1 replacement. Compressed video at DS1 rate or half of it offers adequate quality for nondemanding videoconferencing and tele-instruction applications.

Several modulation techniques that have been proposed for HDSL transmission are baseband 2B1Q (straightforward extension of basic-access DSL transmission method), QAM (quadrature amplitude modulation), and DMT. Interestingly, transmission at two bits per dimension (2 b per baseband symbol or 4 b per QAM symbol) has been found to yield comparable or better results than other considered rates in a number of different HDSL transmission studies [12], [70]–[73]. To date, there are several manufacturers offering HDSL equipment employing different transmission technologies. Bellcore has also issued related tentative generic requirements directed somewhat toward 2B1Q transmission [74].

A schematic diagram of a QAM transmitter-receiver pair is shown in Fig. 6. The transmitter and the receiver are situated at opposite ends of a loop. Omitted from the drawing are the hybrids, for simplicity. The depicted equalization method (baseband complex DFE) is but one of several alternatives. Other possibilities include carrying out the feedforward filtering in the passband [102, Section I.F].

Recently a variant of QAM, known as carrierless AM/PM or CAP, has received some renewed attention. CAP differs from QAM by having a bandpass filter in place of each lowpass-filter-and-modulator pair in the transmitter and in the receiver. An examination of the frequency spectra shows that CAP and QAM are equivalent if the center frequency of the bandpass filters are at ω_c and ω_c is equal to an integer multiple of the symbol rate. A primary reason for favoring CAP over QAM in some implementations is the simplification the former offers [75], [76]. The simplification comes about because, in telephone wire transmission, the center frequency of the bandpass filter is not much different from its bandwidth. This is very different, for example, from typical wireless digital communication systems. In wireless digital transmission, the carrier frequency can be orders of magnitude greater than the signal bandwidth, making the CAP approach disadvantageous by its need to implement a high-Q bandpass filter.

The equalization in both the baseband and the QAM/CAP systems need not be carried out in the receiver but can be performed in the transmitter, the so-called precoding approach, if a feedback channel exists to relay the learned channel characteristics back from the receiver to the transmitter [9], [72], [77]. A qualitative comparison of receiver-side equalization, transmitter-side equalization (both for QAM, CAP, or PAM systems), and DMT will be given after introduction of the DMT technique in the next subsection.

C. ADSL

Prompted by the asymmetric nature of some video services, the ADSL was conceived in 1990. It adopts a data-over-voice approach and employs frequency-division duplex (FDD) for the two directions of data transmission. The highest frequencies are occupied by the high-rate downstream signal. Under it is a low-rate return channel. And both channels are above the voice band. Logically, the high-rate downstream signal actually accommodates two subchannels: the "real" high-speed downstream channel and a low-rate channel which, in association with the low-rate return channel, forms a logical bidirectional data path. By the use of FDD, self-NEXT no longer plagues the digital signals as much as in symmetric transmission. Not even the self-FEXT is a significant transmission issue, but other effects such as circuit noise and spectrum compatibility with other systems. An earliest target, referred to as ADSL1 in some literature, was set to provide DS1-rate one-way downstream transmission and under 16 kb/s of bidirectional data over nonloaded loops.

In time, service and technology considerations led to expansion of the scope of ADSL in terms of its data rates and reach. Two times and four times the one-way downstream rate of ADSL1 and higher low-speed channel rates have been discussed, with possible range limitation to within CSA or even closer for these higher-rate breeds [101]. Strict FDD is no longer upheld and the manufacturer may choose to let the downstream signal overlap the upstream data in spectrum usage. The issues of echo and NEXT therefore resurface, for the low end of the digital signal spectrum.

Fig. 7 shows a network architecture of potential interfaces for the ADSL system. At the subscriber side, the incoming signal will be demultiplexed into the POTS (plain old telephone service), the high-bit-rate signal, and the low-bit-rate control channel. The POTS line is connected to an ordinary telephone. The high-bit-rate signal and the low-bit-rate control channel are terminated in a service module. Depending on the applications, this service module can be an MPEG decoder, a multimedia

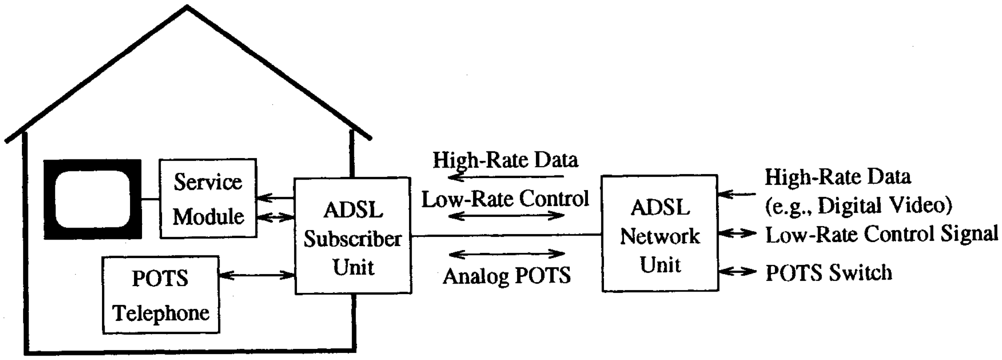

Fig. 7. ADSL concept.

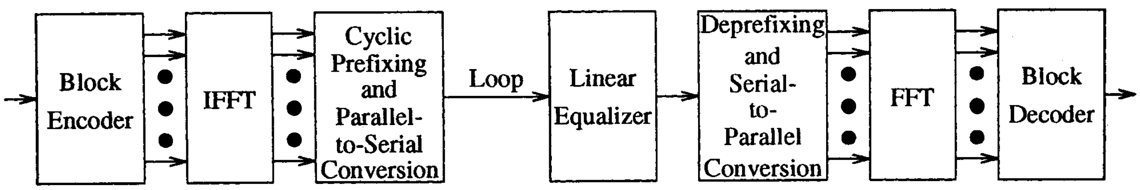

Fig. 8. DMT transmitter-receiver pair.

workstation, or just a personal computer. At the central office, an ADSL system will perform the functions of demultiplexing the upstream POTS signal and control channel, and multiplexing the POTS with high-bit-rate signal in the downstream direction. Using this architecture, many residential applications such as video-on-demand, interactive multimedia communication, distance learning, and other video services can be offered by the local exchange carriers (LEC's) in the near future at an affordable cost. Reference [55] describes additional ADSL system environment and preliminary generic requirements, short of specifying a transmission technique.

As in the case of HDSL, different transmission technologies have been considered for ADSL. Among them the T1E1.4 standards working group picked the DMT in mid-1993 (text of the standard is still being drafted [78]). Nevertheless, nonstandard systems remain in the marketplace. DMT is quite flexible in spectrum usage and rate adaptation. Below we concern ourselves exclusively with the downstream transmission as it is a most critical element in the ADSL system.

Fig. 8 depicts schematically the structure of a DMT transmitter-receiver pair. The basic idea behind this transmission method can be traced back to the 1950's and has emerged in several different forms over the past three decades. References [79], [41], and [80] are good introductions to the DMT technology. The present technology appears to have emerged from the marriage of two paths of development, namely, that of vector coding [81] or structured channel signaling [82], [83] (a method of block orthogonal signaling) and that of traditional multicarrier modulation [84]. Each path has its own theoretical groundings and, especially in the latter case, a history of technological adaptations borne out of practical concerns. In addition, it can be viewed as a variant of FDM (frequency-division multiplexing) which divides the whole channel into a set of parallel subchannels in the frequency domain for transmission.

FDM serves as a good entry point for understanding of DMT, although the latter's theoretical underpinnings are more the orthogonalization of subchannels (in the sense of the subchannel responses being orthogonal and the noise in different subchannels being uncorrelated) and (hopefully) the whitening of noise in each subchannel. At one extreme, we may let the FDM subchannels be completely disjoint in the frequency domain and each be defined by infinitely sharp bandedge cutoffs. If the frequency response of the whole channel is reasonably smooth and the number of subchannels large enough, then each subchannel can be approximated by a constant gain and, therefore, as distortionless. An equalizer is thus not needed. Instead, the receiver provides a simple detector appropriate for distortionless additive-noise channels for each of the subchannels. In the transmitter, different information rates can be assigned to the different subchannels according to their different transmission-gain-to-noise ratios, with the constraint that the aggregate be equal to the desired overall data rate. Under a power constraint on the transmitter output, it can be shown that the lowest total error rate is attained by distributing the overall rate in a way which (approximately) equalizes the error rates of the subchannels.

Note that, inherent in the above discussion is the assumption that the transmitter knows the transmission-gain-to-noise ratio for each subchannel. This typically entails a measurement of the ratio at the receiver and a feedback channel from the receiver to the transmitter for informing the transmitter of this ratio. In an environment where the noise characteristics vary over time, the above measurement and feedback has to be carried out constantly. In terms of practical implementation, the use of parallel independent subchannels, each running at a lower baudrate than the overall rate, eases the speed requirement on the transceiver hardware.

Filters with infinitely sharp bandedge cutoffs are, of course, unrealizable. Practical multitone systems today have been built based on the DFT (discrete Fourier transform) or FFT, its fast version, as the DMT system shown in Fig. 8. At the first sight, one may be puzzled by the use of an inverse FFT at the transmitter. This can be interpreted heuristically by noting that our desire is to send each of the N data streams over a designated frequency subband. Thus each N-vector input to the IFFT unit can be viewed as a frequency spectrum which is defined by the signal and is to be transmitted. We therefore perform an IFFT at the transmitter to convert the spectrum into a time-domain signal for transmission and use an FFT at the receiver to recover the original spectrum. As another remark, we note that each corresponding pair of basis functions in the FFT and the IFFT are in fact each other's matched filter.

In summary, the idea is to send the block-encoder's output vector (call it $\underline{X}$) so that at the block-decoder's input we have a vector whose ith element is equal to H_iX_i (plus noise), where X_i is the ith element in $\underline{X}$ and H_i represents the frequency response of the channel at the ith "carrier frequency." X_i can thus be easily recovered once H_i is estimated.

However, FFT does not yield disjoint subchannels for dispersive media like the subscriber lines. As a result, the subchannels interfere with one another. The transversal equalizer in the receiver and the cyclic prefix in the transmitter are introduced to address this interchannel interference. The function of the transversal equalizer is different from that used in a symbol-by-symbol detection system and hence can be considerably simpler. It is used to compress the channel impulse response so that the equalized channel can be approximated as a short FIR filter. Experience shows that, for transmission up to about 6 Mb/s, it is not difficult to shorten the channel response to about 10–30 symbol periods [41], [79].

Now assume the equalized channel has a length-$(L+1)$ impulse response. The convolution of it with an N-sequence thus yields an $(N+L)$-sequence. To avoid interference among neighboring N-sequences, therefore, we have to space them apart by L dummy symbols. In other words, for each N-sequence of signal symbols we actually transmit an $(N+L)$-sequence in which L symbols are dummies. And the receiver may attempt to recover the original N-sequence based on the received $(N+L)$-sequence. For this, an efficient and noise-resilient method is cyclic prefixing [41], [79]. In cyclic prefixing, the last L symbols in the N-sequence (assuming $L \leq N$) are prepended to the N-sequence to form the $(N+L)$-sequence for transmission. The corresponding output of the equalized channel thus contains the N-point circular convolution between the original N-sequence and the equalized channel response, which can be transformed by an N-point FFT to obtain the desired H_iX_i. The use of cyclic prefix instead of L dummy zeros also increases the SNR.

To avoid over-wasting of channel capacity, we should make $N \gg L$. The delay, however, also grows with N. The DMT ADSL uses N=512 and L=32 [79]. References [85] and [86] discuss echo cancellation methods in DMT ADSL.

A qualitative comparison of the DMT and the QAM/CAP technologies is given in Table 1. For QAM/CAP systems, both receiver-side equalization (DFE) and transmitter-side equalization (Tomlinson precoding) are considered. The DMT properties also apply more generally to other multicarrier modulation techniques. And the QAM/CAP properties also apply to PAM transmission such as 2B1Q. The ideal rates listed in Table 1(a) are normally not attainable. Factors contributing to nonideal performance include the use of short feedforward filters (for QAM/CAP systems) and the presence of cyclic prefixes (for DMT). These will not be treated in detail.

D. VHDSL and Inside-Building Digital Network

A probable scenario for future evolution of DSL transmission is one with copper lines serving increasingly shorter distances between a customer's terminal and a telephone company's equipment (a CO terminal, a remote terminal, an FITL pedestal, etc.). As the length of copper wire-pairs shrinks, transmission loss over such lines is reduced and higher-rate data services can be provided. Associated with the length change in the copper subscriber lines are also statistical changes in the characteristics of the lines. In fact, it has been observed that many of the newly installed drop wires are of better transmission qualities than category-3 inside-building cables. On the other hand, recent years have seen more widespread deployment of twisted-pair LAN's operating at low Mb/s rates and the development of CDDI and SONET premises applications at 100 and 155 Mb/s, respectively, over a stretch of approximately 100 m. Coupled with the recent advances in video compression and ATM standards, it is therefore of interest to deliver ATM to the house and further to the desktop at speeds in excess of 10 Mb/s over outside plant drop wiring.

Many problems are yet to be solved. The slate of transmission technologies may still be the ones previously considered for basic-access DSL, HDSL, and ADSL, or their variants. Indeed, such is what has taken place in the area of high-speed twisted-pair LAN's [76], [88]. For simplicity of equipment interface, the transmission rates

Table 1 Qualitative Comparison of DMT and QAM/CAP Technologies. (a) Comparison in Performance, (b) Comparison in Equalization, and (c) Comparison in Implementation

	QAM/CAP	DMT
Ideal Rate [87]	$\approx \frac{1}{2} \int \log(3 \cdot SNR) df$ in high SNR	$\approx \frac{1}{2} \int \log(3 \cdot SNR+1) df$ when $3 \cdot SNR \geq 1$
Spectrum Usage [87]	Contiguous	Flexible profile
Signaling Delay	Shorter	Longer

(a)

	QAM/CAP with DFE	QAM/CAP with Precoding	DMT
Required Equalizer Length	Long	Long	Short
Problem Error Propagation	Yes	No	No
Combination with Coding	Hard	Easier	Easier

(b)

	QAM/CAP with DFE	QAM/CAP with Precoding	DMT
Feedback Channel	Not required	Required	Required
D/A in Transmitter	Simple	More complex	More complex
Change in Data Rate	May require baudrate change	May require baudrate change	Easier
Required Processing Speed	Higher	Higher	Lower

(c)

may be chosen to be integer fractions of OC1. Opinions of industrial standards organizations, such as the ATM Forum, will play a role in this regard.

E. Spectrum Compatibility

As we have seen, the use of unshielded twisted pair leads to crosstalk between pairs in the same binder group. Impulse noise also couples into and between pairs. When installing a new transmission capability, network engineers must be concerned with the noise environment that the new transmission system will have to withstand, and also the noise levels that the new system will couple into other existing services in the same binder group. To simplify provisioning and engineering, it is highly desirable that all systems in the public network be spectrally compatible, that is they are neither excessively impaired by crosstalk or impulse noise, and they don't excessively crosstalk into other systems.

This has been a prime concern in the development of DSL technology and has led to classification of "protected services," which are all mutually compatible. These services include conventional voice service, digital data services at speeds up to 56 kbps, ISDN basic-access DSL, and DS1 service using T1 carrier. Some analog carrier systems may not operate satisfactorily in the presence of basic-access DSL's.

The introduction of new transmission capabilities must be carefully planned in terms of the transmission spectra and the power levels of the systems. Studies indicate that the basic-access DSL, the (baseband) dual-duplex HDSL, and the ADSL can coexist with each other and with digital data service and T1 carrier. The DSL operates at a line rate of 80 kBd with the transmit spectrum extending up to approximately 50 kHz. The baseband HDSL operates with a transmit spectrum extending up to about 200 kHz. (Passband HDSL's which signal at a rate around 200 kBd should be similar.) In ADSL, depending on the modulation scheme used, the digital section transmits in a band in the neighborhood of 50–500 kHz or up to approximately 1 MHz. This staggering of transmit spectra and the asymmetrical nature of the ADSL will allow these new DSL technologies to be gracefully added to the network.

At the higher rates, spectrum compatibility becomes more difficult because of increased coupling and because of potential radiation to and from systems outside of the cable that they are implemented on. Now in addition to the relatively short range interference among pairs within a binder group, radiation will travel farther and must conform to radiation limitations set by FCC rules. In addition, at rates of 10 Mb/s and above over unshielded pairs, radiation from commercial AM broadcasting, aeronautical transmissions, and maritime transmissions may actually couple into the pairs, creating yet another source of noise. With proper system design, however, these spectrum management issues could be successfully controlled.

IV. Visual Communications Standards

Direct transmission of digital video usually requires a high bandwidth channel. For example, uncompressed digital NTSC television signals require a bit rate of about 100 Mb/s. This far exceeds the current digital transport capability available in the copper loop plant. For effective video communications and services, efficient coding techniques which reduce the transmission rate and the transmission cost for a given picture quality are needed. Remarkable advances in video compression have come about in just the last few years, especially through the international standardization endeavors. Research efforts are directed toward finding efficient video compression and image coding methods and their implementations with inexpensive, power-efficient high-speed VLSI-based systems.

The asymmetrical capability of the ADSL will be ideal for advanced videotext services. With the help of compression techniques, databases and high resolution graphics can be rapidly manipulated by the user. Several standards bodies have been working on algorithms to compress video. Image compression and video coding can be generally divided into two broad categories: intraframe and interframe coding. Intraframe coding is used to remove spatial redundancy in single-frame images. Interframe coding takes advantage of the strong correlation among video frames to reduce the temporal redundancy. Generally speaking, both spatial and temporal redundancy reduction should be considered together in order to achieve the highest possible compression ratio. Below we give a brief overview on the most commonly known visual communication standards. Specifically JPEG, H.261, MPEG1, MPEG2, and the emerging MPEG4 are described concisely. Further technical details of these compression standards can be found in [89]–[97].

A. ISO JPEG Standard

The ISO Joint Photographic Experts Group or JPEG has developed a compression standard for continuous-tone still image applications such as graphic arts, color facsimile, and desktop publishing [89], [90]. The JPEG standard specifies both baseline and extended systems, which are intraframe coding schemes. The baseline specifications are the minimum requirements every JPEG product should follow.

In the JPEG baseline system, the input image is divided into disjoint 8×8 blocks. Two-dimensional discrete cosine transform (DCT) is applied to each block, followed by a quantization to reduce the data dynamic range. The 2-D (8×8) quantized DCT coefficients are then zigzag scanned into a 1-D data sequence where the neighboring contiguous zeros are grouped together into a run length which can be coded more efficiently. The Huffman code, which assigns shorter codewords to symbols of higher probabilities, is used to code the runlengths and nonzero coefficients. In the baseline system, the Huffman code information can be embedded in the bitstream and sent to the decoder for codebook generation. The JPEG standard specifies how the codebook is generated from the encoded information.

In addition to the baseline system, JPEG also specifies an extended system which offers more features (such as arithmetic coding and more precision in the pixel values) and additional coding structures (such as progressive transmission and lossless coding for various applications). The JPEG activity was started around 1987, and has become an international standard in 1991.

Although the JPEG standard was originally targeted for still image applications, it can also be applied to motion video by treating each frame as a still image. Therefore the standard here is regarded as an intraframe scheme. By using the JPEG motion video compression, it is expected that decent NTSC video quality can be achieved with a bandwidth of 8–10 Mb/s.

B. ITU-T H.261 Standard

In order to provide videophone and videoconferencing services over ISDN, ITU-T (formerly CCITT) has recently completed and approved the H.261 standard [91]–[93]. The H.261 standard specifies a real-time encoding-decoding system with a delay less than 150 ms. The algorithm is capable of running over a set of transmission rates of $p \times 64$ kb/s ($p = 1, 2, \ldots, 30$). The format for the input image is based on the Common Intermediate Format (CIF) which is 360 pixels by 288 lines for luminance and 180 pixels by 144 lines for chrominance. The frames are non-interlaced, and the input rate is 29.97 frames/second for NTSC-compatible systems. For videophone applications where low bit rates are required, another format, 1/4 CIF (QCIF) which is 180 pixels by 144 lines for the luminance and 90 pixels by 72 lines for chrominance signals, has also been defined in the standard.

The coding scheme for the H.261 standard can be summarized as follows. Each frame is divided into disjoint macroblocks each of which consists of one 16×16 luminance (Y) block and two 8×8 chrominance (Cr and Cb) blocks. For each luminance block, we find from the previous frame a best-match block. The purpose here is to remove the temporal redundancy. This process is called motion estimation. For a typical video scene, it is highly likely that an object will sustain for some period. Motion estimation searches for a representation of the current macroblock from a previously coded picture. When a suitable representation is found, only information apart from that representation is needed to be coded, and compression is achieved. How to find the best-match block is not specified in the standard. Nevertheless, the most commonly used criterion is the minimization of the displaced block difference (DBD). A best-match motion vector (displacement) is obtained from this process.

The coder then decides whether intra- or predictive (inter-) mode should be used. In the predictive (inter-)

mode, the motion-compensated predictive error, defined as the difference between the current block and the best-match block (the block in the previous frame displaced by the best-match motion vector), is coded. Otherwise, the current block is coded directly. If the best-match block data are not quantitatively close to the current block, the motion-compensated predictive error will be large, and thus coding the current block directly is more advantageous. Note that in predictive mode, the best-match motion vector needs to be sent to the decoder for video reconstruction. In either intra- or predictive mode, a DCT followed by a quantizer is used to code the data. This DCT coding process is almost the same as that given in JPEG's standard, and the details are not repeated here. The quantizer step size is adjusted periodically to govern the resulting bit rate to the desired value. A possible method is to determine the quantizer step size directly from the buffer content.

Although the H.261 standard was designed for two-way videoconferencing, certain applications may emerge in which there is just one direction of video transmission. For example, educational applications are envisioned in which a lecture is encoded in real time and transmitted to remote classrooms and to students at home. At the transmission rate of 1.5 Mb/s, it could provide sufficient quality for coverage of the lecturer and the board or visuals from an overhead projector.

C. ISO MPEG1 Standard

Fueled by the market success of compact disc digital audio, CD-ROM's have recently made significant inroads into the arena of data storage. They are also considered a promising storge medium for multimedia video application. The raw data rate of a compact disc player is around 1.5 Mb/s. Toward this and similar applications, ISO has completed an international standard named MPEG1. The emergence of multimedia video applications is synergistic with the ADSL. With DS1-rate ADSL, users can run applications across the public network with the same level of performance as provided by a local copy.

The ISO activity of the Moving Picture Experts Group (MPEG) was started in 1988 for CD-ROM applications at a bit-rate below 1.5 Mb/s. MPEG1 was originally developed for storage of full-motion video. The recommended input picture size for the MPEG standard is 360×240 pixels for luminance and 180×120 for chrominance, and the frame rate is 29.97 frames per second for NTSC-compatible systems, although these parameters can vary in the standard. The MPEG1 coding scheme is very similar to that of ITU-T H.261. The major difference between the two is that MPEG1 allows bi-directional motion-compensation.

A video sequence is divided into groups of pictures (GOP's). There are three possible types of pictures in a GOP: I-Picture, P-picture, and B-picture. Coding of I- and P-pictures is similar to the scheme used for the JPEG and H.261 standards, i.e., intraframe and motion-compensated interframe techniques with DCT. Coding of B-pictures is slightly different from that of the P-pictures. In P-pictures, motion prediction is obtained from some previous frame only, i.e., forward motion compensation. However, in B-pictures, information from both previous and future frames has to be used for motion prediction, i.e., bi-directional motion compensation. For each macroblock in B-picture we find the best-match blocks from the previous and the next I- or P-pictures. The best-matched block(s) from the previous, from the next I- or P-pictures, or from both can be used as the motion-compensated prediction. The choice is not specified in the standard. As in P-pictures, either the motion-compensated prediction error or the current block is coded with DCT followed by a quantization. Since processing of B-pictures requires the information in the previous and the next I- or P-pictures, the frame processing order cannot follow the natural sequence of frame numbers. Thus, given two B-pictures between two adjacent I- and P-pictures, after processing of Frame 0 (which is intraframe coded) is completed, Frame 3 (which is a P-picture) is processed, followed by Frames 1 and 2 (which are B-pictures), and so on. This process is repeated until the intraframe mode is triggered at the first frame of the next GOP, and another new cycle starts.

The MPEG1 standard only specifies the bit stream syntax and decoding process. At a given bit rate, different standard-conforming encoding schemes are possible. It has been demonstrated that at 1.5 Mb/s, the video quality is comparable to the VCR's. Moreover the quality is expected to keep improving thanks to the encoding flexibility. This poses a great potential for ADSL to provide the video-on-demand service by using the MPEG1 standard where the uplink channel in ADSL can be used for interactive communication between user and information provider. A video-on-demand prototype has been demonstrated by using the ADSL technology and the CD-I providing the MPEG1 compressed bit stream [98].

D. ISO MPEG2 (and ITU-T H.262) Standard

After completing the MPEG1 standard for compressed video targeted at storage applications around 1.5 Mb/s, the MPEG group continued to enact the MPEG2 standard for broader and higher bit-rate applications including broadcasting, consumer electronics, and telecommunications. The MPEG1 standard can provide a broad range of bit-rates due to its parameterization approach, although the standard was originally aimed at below 1.5 Mb/s for CD-ROM applications. However, its quality and flexibility are considered not ideal enough for the above-named applications at higher bit rates. To meet the higher quality and higher bit-rate objective, the MPEG2 project was put together. The technical contents for the MPEG2 video coding standard has been frozen [96]. It is expected that the MPEG2 standard will be ratified in 1994.

Roughly speaking, coding algorithms conforming to the MPEG1 and MPEG2 standards are similar, i.e., motion-compensated interframe coding schemes using DCT and both forward and backward predictions. As in the MPEG1 standard, there are three picture coding types: I-, P-, and B-picture. Processing of these three types of pictures are similar to MPEG1's. Nevertheless, the MPEG2 standard accommodates more features than the MPEG1 standard. Such features include interlaced video manipulation, scalability, compatibility, error resilience, and "hooks and options" for very high resolution video coding. One of the major differences between the coding schemes for the MPEG1 and MPEG2 standards is that the MPEG2 standard is capable of handling interlaced video sequences adaptively with either frame or field modes, while the MPEG1 standard can only deal with one fixed mode. Given an interlaced video, the MPEG1 coder can only treat each field or each frame (by merging the adjacent two fields) as an integral unit, whereas the MPEG2 coder can, in addition, exploit the correlation between the two fields in a frame and select the optimum mode. The MPEG2 standard also provides more options for coding schemes, including choice of zigzag or slanted scanning, linear or nonlinear quantizer table, and the DC resolution for intra-macroblocks. More detailed comparison between the MPEG1 and MPEG2 standards can be found in [99]. Scalability can provide multiple-grade and multiple-bandwidth services. It can also be used for error resilience under error-prone environment. The MPEG2 standard provides five types of scalability tools: data partitioning, SNR scalability, chroma simulcast, spatial scalability, and temporal scalability. The details can be found in [96].

Since MPEG2 is intended to be a generic standard to meet requirements of vastly different types of applications, ISO/MPEG has decided to associate the MPEG2 standard with *Profiles* and *Levels*. A *Profile* uses a subset of the available syntactic elements to support a number of technical features and functionalities required by a cluster of similar applications. The MPEG2 standard has currently defined five Profiles: Simple, Main, SNR, Spatial, and High. Main Profile (MP) aims to support the most common applications that are likely to be introduced first. Simple Profile (SP) allows lower memory and hardware complexity at the expense of fewer technical features and thus lower quality. The last three Profiles provide an environment for scalability and higher chrominance resolutions. *Levels* are defined by the range of parameters such as picture size, frame rate, bit rate, pixel rate, buffer size, etc.

It is expected that a full NTSC video quality can be achieved with a bit rate of 4–6 Mb/s by using the MPEG2 standard. Although the needed bandwidth may eventually be provided by fiber, the current penetration of fiber in the feeder plant (primarily in conjunction with the deployment of digital loop carriers in the residential environment) is still not large enough to immediately provide ubiquitous service. On the other hand, currently there are about 75 million lines in the US nonloaded loop plant which are potential sites for ADSL-based video services. Therefore the copper loop plant may provide a shorter term solution.

Multiple channels in association with the 1.5 Mb/s ADSL technology can provide a way to accommodate such a bandwidth. Alternatively, higher-rate ADSL providing 3–6 Mb/s transport capability over shorter ranges may be employed. Based on an uncoded quadrature amplitude modulation (QAM) passband signaling scheme with perfect timing and carrier recovery, preliminary simulation results [31] indicate that bit rates of 3–4 Mb/s may be possible over CSA's. The DMT signaling scheme is expected to provide comparable performance if not better. A higher-rate ADSL channel allows simultaneous viewing of two different video channels with MPEG1 compression or one channel of NTSC quality video with the forthcoming MPEG2 standard.

Compared to H.261, the resulting quality from MPEG1 and MPEG2 is better due to the use of bidirectional prediction. However, they have a looser requirement on coding delay as is dictated by two-way applications. These standards along with H.261 and the resultant low cost coders and decoders will help stimulate and support sophisticated multimedia and educational applications.

E. MPEG4

ITU-T SG XV formed a group in early 1993 to enact a short-term low bit-rate coding standard aiming at PSTN (public switched telephone network) applications with modem [100]. Although there is no upper limit set for the bandwidth, it is expected that this standard will be used mainly with the V.32 and V.34 modem technology. The video compression technique will again be DCT-based motion-compensated interframe coding. The standards draft will be completed in mid-1994, and will be frozen in early 1995. Since the standard is targeted at visual telephony over POTS, it demands a voice quality at least as good as the voice-only service. By associating the standard with the DSL technologies, the PSTN can provide a greater bandwidth for better video and voice quality. The group is also collaborating with the ISO MPEG4 group for the longer-term MPEG4 standard. The MPEG4 standard is expected to be ratified in 1998.

The MPEG4 has targeted applications which include real time audio-visual communications, multimedia, and remote sensing. It has also identified categories of requirements, including input material, quality, video format, audio format, bit rate, delay, complexity, error resilience, security, interactive operation, network interoperability, annotation of other data, and extensibility. A single compression algorithm may not be able to cover the entire spectrum of the targeted applications and requirements. It is very different from MPEG1/2 in that the coding technology behind it is yet to be selected while for MPEG1 and MPEG2, the technology was clearly converging at the very beginning.

V. Conclusion

As fiber penetrates the feeder network, the remaining embedded copper plant can be used to deliver shorter range, higher bit-rate access capabilities in support of emerging broadband services. ISDN basic-rate access is now emerging to provide ubiquitous 144 kbps digital capability over the nonloaded loop plant. HDSL will provide repeaterless T1 capability within Carrier Serving Areas. It is expected that the ADSL will emerge to deliver one-way 1.544 to 6.312 Mb/s from the network to the residence for compressed video and asymmetric data. As fiber-to-the-curb and other broadband architectures are deployed, copper pairs can provide even higher bit rate access capabilities over shorter distances.

Adaptive DSL transceivers automatically adjust to the wide range of channel characteristics found in telephone loops. Continuous adaptation allows equipment to track changes that will occur in the loop over time, for example due to environmental changes. Adaptive technology not only allows quick, automatic provisioning, but also optimizes transmission on a loop-by-loop basis by adjusting transmission parameters in a highly precise fashion.

The use of unshielded twisted pair for high-rate transmission leads to crosstalk between pairs in the same binder group. To ensure such crosstalk disturbance does not lead to significant operational difficulties, the introduction of new transmission capabilities must be carefully planned in terms of the transmission spectra and the power levels of the systems. As the transmission rates are increased, spectrum compatibility becomes more difficult because of increased coupling and because of potential radiation to and from systems outside of the cable. With proper system design, however, the problems could be successfully controlled.

The copper distribution plant can be used to support the evolution of the public network toward a fiber-based broadband network. The use of DSL technology will facilitate this evolution, allowing copper facilities to be reused at even higher bit rates over shorter distances. Advances in VLSI technology made ubiquitous per-line DSL technology economically feasible in the 1980s. Continuing VLSI advances will allow DSL technology to prove in at higher bit rates. These advances will allow the telephone companies to take advantage of their embedded base of copper facilities to complement deployment of fiber.

Concurrent with the progress in DSL technologies are the advances in digital video compression techniques. It is now feasible to send video of acceptable quality for videotelephone and some videoconferencing applications over basic-access DSL. HDSL can support videoconferencing and tele-instruction functions. Transmitting at the same rate as dual-duplex HDSL, the 1.5-Mb/s ADSL can deliver one channel of VCR-quality video for entertainment and multimedia services on top of POTS. More VCR-quality channels or higher-quality video can be provided through higher-rate ADSL, which also supports higher data speeds in the low-rate channel for increased functionalities. It is expected that the VHDSL and inside-building digital networks built on twisted-pairs will be able to extend and enrich the above video capabilities. At the other end of the spectrum, very low bit-rate video coding techniques are also advancing contemporaneously with continued development in voice-band modem technology. As a result, there could emerge an array of new-generation video communications equipment tuned to different subscriber line transmission mechanisms to provide various visual communications services in the not-to-distant future.

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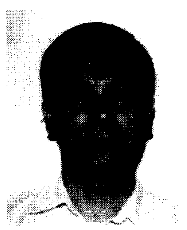

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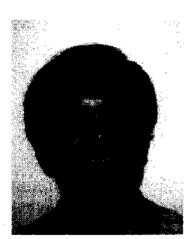

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