Application of Capacity-Approaching Coding Techniques to Digital Subscriber Lines

Evangelos Eleftheriou and Sedat Ölçer, IBM Zurich Research Laboratory
Hamid Sadjadpour, University of California at Santa Cruz

ABSTRACT

The use of coding for error control is an integral part of the design of modern communication systems. Capacity-approaching codes such as turbo and LDPC codes, discovered or rediscovered in the past decade, offer near-Shannon-limit performance on the AWGN channel with rather low implementation complexity and are therefore increasingly being applied for error control in various fields of data communications. This article describes a generic multilevel modulation and coding scheme based on the use of turbo, turbo-like, or LDPC codes for DSL systems. It is shown that such codes provide significant gains in performance and allow an increase in data rate and/or loop reach that can be instrumental to the widespread deployment of future DSL services. Such techniques are also suitable for general multilevel modulation systems in other application areas.

INTRODUCTION

Fueled by the need for high-speed Internet access, the deployment of digital subscriber line (DSL) services is nowadays proceeding at a rapid pace. The success of DSL services builds, to a large extent, on the underlying transmission technology, which has shown continual progress since the publication of the first asymmetric DSL (ADSL) specification T1.413 by the American National Standards Institute (ANSI) in 1995. In fact, not only did successive DSL specifications introduce novel features and steady enhancements at the physical layer or in terms of network management, testing, diagnostics, and so on, but manufacturers also significantly improved their chipsets and system solutions over several generations. At the same time, local exchange carriers have acquired valuable field experience, which has helped guide the standards processes and influenced equipment designs.

It appears that DSL systems will still be evolving over many years under the auspices of several standards organizations. A common requirement for all future DSLs will be the attainment of higher transmission performance. For example, the long-reach DSL project developed at the International Telecommunication Union (ITU) aims to extend the loop reach of current systems. Range-extended ADSL has already been adopted as a first step of this project. The solutions needed to meet the challenges of future systems will no doubt draw on the combined use of a number of advanced transmission techniques. Range-extended ADSL, for example, starts to introduce the use of selectable masks for the spectra of the transmitted signals and thereby achieves loop-reach extension of a few kilofeet under severe crosstalk conditions. Other solutions envisioned today include dynamic spectrum management, currently being defined in Committee T1, and multi-input multi-output transmission techniques akin to those studied for wireless systems. Yet another avenue is offered by advanced error correction coding (ECC) techniques, which is the topic addressed in this article.

The T1.413 specification by ANSI was the first standard to incorporate ECC for DSL systems. This document specifies the use of Reed-Solomon (RS) coding as a forward error correction technique for ADSL systems that employ discrete multitone (DMT) modulation [1]. In DMT modulation, the signal frequency band is subdivided into a set of contiguous bands over which transmission is performed in parallel. DMT is a powerful modulation scheme and has certain advantages over conventional single-carrier schemes for wire transmission. The ANSI document also includes the optional use of a trellis coded modulation (TCM) scheme as an inner coding mechanism to improve communication reliability further [2]. TCM schemes employ redundant multilevel signaling in conjunction with an encoder that is used to select the sequence of multilevel symbols to be transmitted. TCM is employed in a variety of wireline and wireless communication systems. Subsequent ADSL specifications have retained the above coding structure, with some variations on the set of allowed parameter values. The current very-high-speed DSL (VDSL) standards (Standards Committee T1: T1.424, European Telecommunications Standards Institute [ETSI] TS 101 270) have only included outer RS coding, whereas the single-

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pair high-speed DSL (SHDSL) specification (ITU — Telecommunication Standardization Sector [ITU-T] G.991.2) has only included inner trellis coding without an outer error correction code. The latest ADSL2 Recommendation by ITU-T makes the use of inner TCM mandatory for upstream as well as downstream transmission.

This tutorial article considers the incorporation of more powerful near-Shannon-limit coding techniques to DSL systems. These techniques were developed within the coding community over the past 10 years, following the invention of turbo codes in 1993. They are now being adopted in many communication standards, particularly for wireless systems. The article demonstrates that the application of turbo coding [3] and low-density parity check (LDPC) coding [4] allows significant performance improvements in DSL without incurring a substantial increase in transceiver complexity. It is therefore argued that the inclusion of such coding techniques in DSL systems will facilitate the path toward allowing higher service penetration, higher data rates, and more robust system operation. Both downstream and upstream transmissions shall benefit from capacity-approaching coding.

The coding structure described in this article is generic and can be used in conjunction with other capacity-approaching coding techniques (e.g., block product codes [5]) that are also decodable in an iterative fashion. It is suitable for general multilevel modulation in other application areas as well. The important question of whether these coding schemes should be part of a concatenated structure with outer RS coding is a topic of intensive current research.

This article assumes that the reader is familiar with basic turbo coding and LDPC coding terminology. The recent Feature Topic in *IEEE Communications Magazine* [6] on capacity-approaching coding techniques provides an excellent introduction to this field.

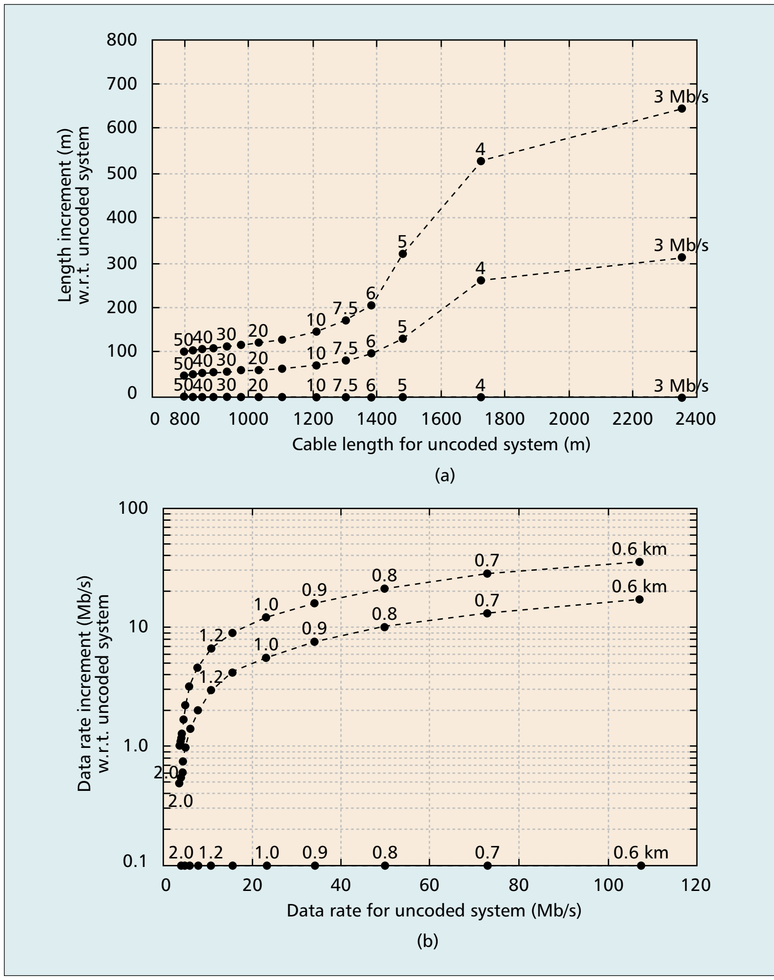


Figure 1. *Examples of performance improvements due to coding for DSL transmission: a) cable length increase achieved by coding (the numbers above the curves indicate achievable data rates in megabits per second); b) data rate increase achieved by coding (the numbers above the curves indicate cable length in kilometers; the data rate increment for the reference uncoded system is artificially displayed at the ordinate value of 0.1 instead of zero because of the logarithmic scale). The specific assumptions used to derive the results were signal attenuation of 75 dB at 5 MHz for a cable length of 1 km; AWGN with a power spectral density of –140 dBm/Hz; NEXT due to 49 ADSL (ITU-T G.992.1) downstream transmitters that use all the subchannels; transmit power of 15 dBm; SNR margin of 6 dB; gap to capacity for uncoded transmission of 9.95 dB, which implies operation at a symbol error rate of 10^{-7}.*

Benefits of Coding for Digital Subscriber Lines

Before examining coding issues in more detail, it is appropriate to understand the benefits of ECC in DSL systems by means of a simple example. To this end, we consider data transmission over an ordinary unshielded twisted pair and assume that the received signal is disturbed by near-end crosstalk (NEXT) as well as additive white Gaussian noise (AWGN). Furthermore, we have assumed that NEXT disturbance is due to 49 ADSL downstream transmitters that share the same cable binder as the data transmission system under consideration. The following questions are of interest: assuming operation at a fixed data rate, by how much can the cable length be increased for this system through ECC? Alternatively, by how much can the data rate be increased through ECC for a fixed cable length? It is possible to answer these questions by first computing the capacity of this communication channel for different data rates and cable lengths. The results are summarized in Fig. 1.

For example, Fig. 1a shows that for operation at 3 Mb/s, the loop reach of this system is limited to 2355 m. If coding is employed with a net coding gain of 3 dB (6 dB), the loop reach can be extended by 315 m (645 m). As another example, Fig. 1b shows that for a loop reach of 2.0 km, an uncoded system can achieve a data rate of 3.5 Mb/s, and a coding gain of 3 dB (6 dB) allows this data rate to be increased by 480 kb/s (1 Mb/s). This shows that coding can play an important role in increasing the performance of DSL systems in terms of data rate and loop reach. Clearly, similar results can be derived for a variety of channel and noise models.

Simulation studies indicate that coding gains of 6 dB and more can practically be achieved through the use of capacity-approaching coding techniques in DSLs. Anticipating the results to be presented later in the article, some examples are given in Fig. 2. Here spectral efficiency vs. signal-to-noise ratio (SNR) representations, often used in digital communications to compare different modulation schemes, are shown in con-

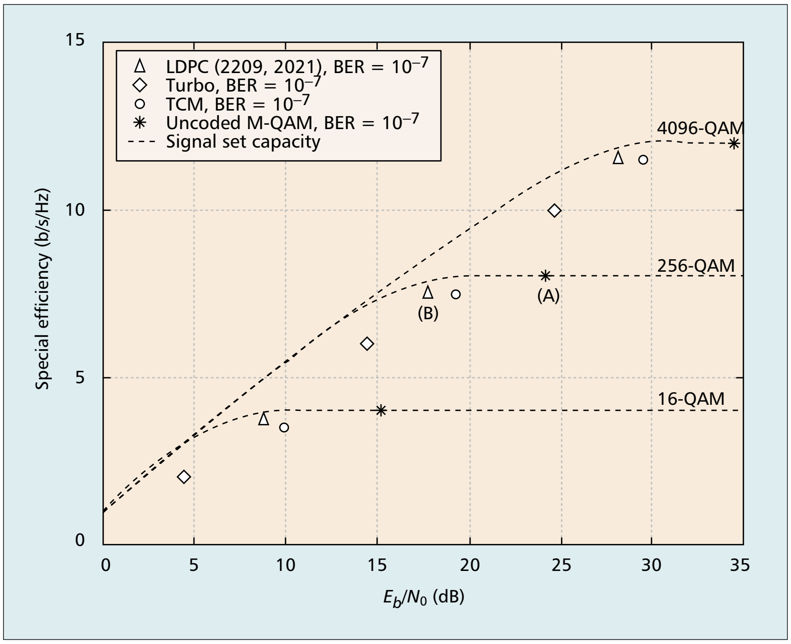


Figure 2. *A comparison of several coded modulation schemes at 10^{-7} BER.*

nection with 16, 256, and 4096 quadrature amplitude modulation (QAM).

Consider, for example, 256-QAM. The "star" (A) indicates that uncoded transmission at a spectral efficiency of 8 b/s/Hz requires an SNR of E_b/N_0 = 24.4 dB, where E_b denotes the energy per bit and N_0 is the AWGN power spectral density. This value corresponds to a bit error rate (BER) of 10^{-7}. If an LDPC code mapping 2021 information bits to 2209 encoded bits is employed with the same 256-QAM constellation, the SNR value can be lowered to 17.8 dB while still ensuring operation at a BER of 10^{-7} (triangle B). In this case, transmission occurs at a spectral efficiency of 7.49 b/s/Hz because of the redundancy needed for coding. In other words, this LDPC coded system achieves a coding gain of ~ 6.6 dB at a BER of 10^{-7} with 256-QAM. Similar results can be read from this figure for the other constellations sizes, as well as for turbo coding (diamonds) and TCM (circles). The dashed lines show the capacity of the band-limited AWGN channel with "discrete" QAM symbol constellations as a function of the SNR.

Having illustrated the advantages of coding in DSL systems, it is now meaningful to ask what are the important aspects that must be considered in designing coding schemes for these systems? First, *high code rates* are desirable to achieve high spectral efficiencies for bandwidth-constrained DSL transmission. Second, it should be possible to *adapt the code parameters* to given transmission channel characteristics and application-specific constraints to achieve the best performance. A simple solution would consist of allowing the receiver to select the most appropriate code from a small set of predefined codes after the channel measurement phase during transceiver initialization. An approach that appears to be even more attractive, because of its greater flexibility, is one that would allow code construction on the fly, provided that the processing effort needed to compute interleaver patterns for turbo codes, or parity check matrices for LDPC codes, is small. A third aspect is *linear time encodability*, meaning that encoding for a code of length N requires $\mathcal{O}(N)$ operations. This very important property, which is natural in turbo coding, should also be achieved by LDPC codes.

In DSL transmission, overall delay, or *latency*, is also a critical issue. Voice applications are known to demand rather low latency, whereas other applications, such as video streaming, tolerate larger delays but need stronger error correction capability. Thus, in studying new coding techniques for DSLs, trade-offs between coding gain and latency have to be clearly established.

Finally, another important issue is the *implementation complexity* associated with coding. Complexity is a critical parameter, especially at central office access multiplexers or remote terminals, because it directly affects equipment cost and power consumption.

Transmitter and Receiver Functions

DMT-based DSL systems [1] employ a flexible multicarrier modulation method, whereby each subcarrier can be modulated by symbols taken from constellations of different sizes, such as binary phase shift keying (BPSK), quaternary PSK (QPSK), or 16, 32, 64-QAM, ultimately up to a 2^{15}-symbol constellation. The mapping of bit sequences to modulation symbols and the corresponding inverse mapping at the receiver thus represent an important functionality. An attractive generic realization of the modulator and demodulator in a DMT system that uses advanced coding schemes and iterative decoding is shown in the block diagram of Fig. 3. The structure depicted is generic in that it can be employed to implement turbo-coded, LDPC-coded, or other capacity-approaching coded modulation schemes.

Information bits representing data or control messages are encoded into a binary codeword of length N. Here both the turbo and LDPC coding schemes are regarded as rate-K/N binary block codes. For turbo coding, the number of information bits K corresponds to the size of the interleaver. The symbol mapper collects groups of coded bits, possibly along with uncoded information bits as shown in the figure, and builds QAM symbols for frequency domain modulation by an inverse discrete Fourier transform. In Fig. 3 uncoded bits are mapped to the more significant bits of a QAM symbol, which are less prone to detection errors at the receiver and thus do not need the same level of protection as the less significant bits do. As usual, the size of the QAM constellation used on each subchannel is determined through a bit loading algorithm [1].

The number of uncoded and coded bits per transmit symbol is a design parameter that involves a trade-off in terms of performance and decoding complexity. For full flexibility, this parameter can be specified by the receiver for each symbol constellation during initialization. Assuming, for example, that 1024-QAM is employed on each subchannel, with a total number of 200 subchannels, and that 3 coded bits are carried per dimension, a codeword of length N = (3 + 3) bits × 200 = 1200 bits along with 4

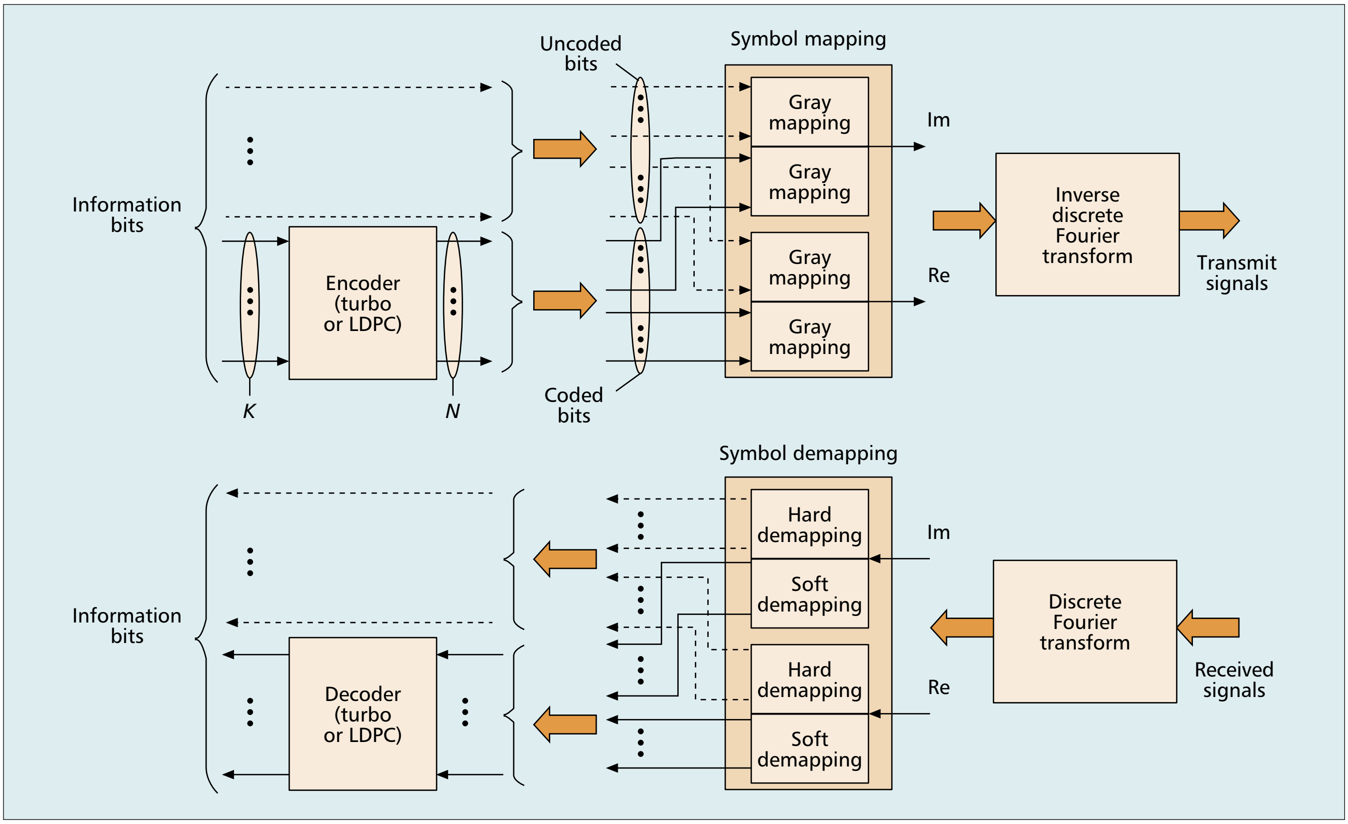


Figure 3. *Encoding/symbol mapping and symbol demapping/decoding functions for DMT-based DSL transmission. Mapping and demapping functions are shown for the real (Re) and imaginary (Im) parts of a complex QAM symbol.*

bits × 200 = 800 uncoded bits can be mapped into a DMT frame. In this case, with a DMT frame rate of 4000 Hz, the line data rate is 10 bits × 200 × 4000 Hz = 8 Mb/s.

In the scheme of Fig. 3, symbol mapping relies on *double Gray-code labeling*, in which the less significant and more significant uncoded bits are Gray-coded separately. The reason for this mapping technique is that a good trade-off is obtained in terms of achievable performance and implementation complexity, as shown in [7]. Alternative approaches of capacity-approaching bandwidth-efficient coded modulation schemes based on LDPC codes were proposed in [8].

At the receiver, the complex noisy symbols obtained at the output of the discrete Fourier transform are processed by a demapper, whose function is to generate soft reliability information on the individual code bits and hard decisions on the uncoded information bits. Because the less significant coded bits are Gray-coded, near-optimum extraction of soft information is possible. Likewise, because the uncoded more significant bits can be recovered via simple threshold detection, the separate Gray coding used for them permits the BER to be lowered. Note that reliability information can take the form of a posteriori probabilities (APPs) or (log) likelihood ratios. It is used for soft iterative decoding using the BCJR algorithm [3] in the turbo case or the sum-product algorithm [4, 9] in the LDPC case, or some reduced-complexity decoding scheme. The information bits recovered are finally output from the receiver.

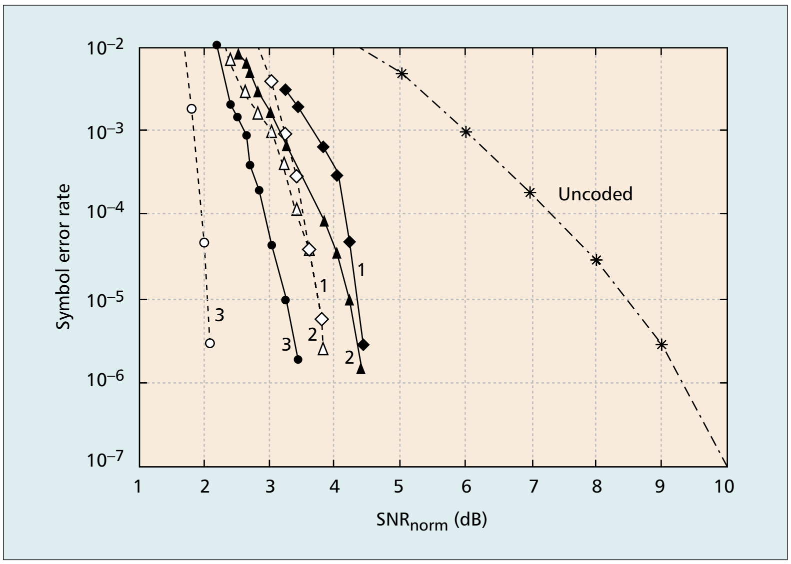
The above description suggests the possibility of specifying a generic coding scheme for DSL systems such that either turbo-coded or LDPC-coded transmission would be possible. In this approach, all transmitters would be capable of turbo and LDPC encoding of data. A receiver would only implement either turbo or LDPC decoding (not both) and would indicate during startup which of the two techniques should be used. The clear advantage would be increased freedom for receiver design.

Performance

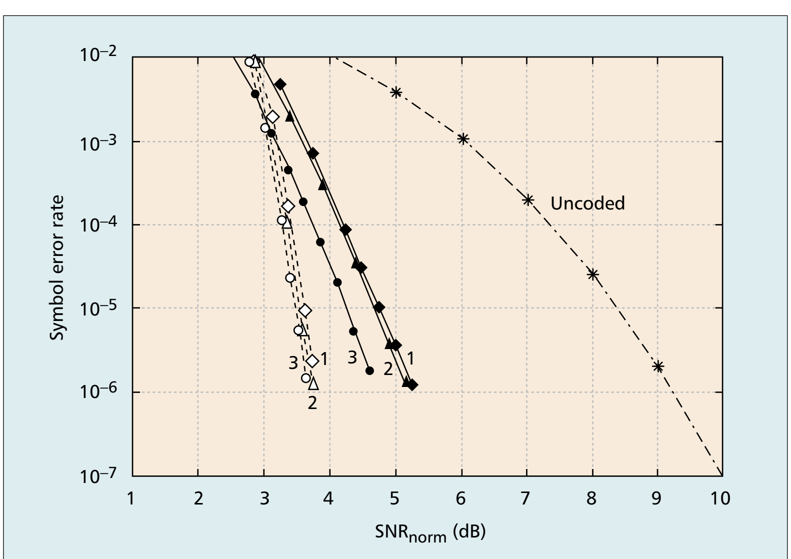
In this section the typical performance that can be achieved by the turbo and LDPC-coding schemes described above is illustrated by means of simulations. The telephone-twisted-pair channel introduces frequency-dependent signal distortion as well as several other forms of disturbances, of which crosstalk is the most important. In the following, disturbance by AWGN only will be assumed. The reason for this is that if each DMT subchannel has a sufficiently narrow bandwidth, each independently approximates an AWGN channel with a particular SNR value. Clearly, impulse noise and narrowband interference of various origins (e.g., AM radio signals) also affect the reliability of communications in DSLs. Performance should ultimately be assessed using actual test-loop conditions. Note, finally, that many ADSL systems today use far more than 6 dB margin in practice because of the possibility of unforeseen crosstalk and other types of noise that arise during operation. The potential gain that might result in this respect from the use of capacity-approaching codes would need to be investigated.

To evaluate performance, both uncoded and coded systems are represented in terms of symbol error rate (SER) vs. the normalized SNR, SNR_{norm}, which for a modulation and coding scheme operating at a given rate of η (in bits per two-dimensional symbol) is defined as [10]

$$SNR_{norm} = \frac{SNR}{2^{\eta}-1} = \frac{\eta}{2^{\eta}-1}\frac{E_b}{N_0}.$$



■ **Figure 4.** *Performance of two turbo codes for transmission over the AWGN channel. Solid (dashed) lines denote the case with an interleaver length* K *of 462 (4224) bits. Curves 1: 16-QAM; curves 2: 256-QAM, and curves 3: 4096-QAM. Each component code is generated by an 8-state recursive systematic convolutional encoder with feedback and feedforward polynomials equal to $(15)_{oct}$ and $(17)_{oct}$, respectively. Semi-random interleaving is used. For decoding, a total of 20 iterations of the log-MAP algorithm are performed.*



■ **Figure 5.** *Performance of two LDPC codes for transmission over the AWGN channel. Solid (dashed) lines denote the case with a block length* N *of 529 (4489) bits. Curves 1: 16-QAM; curves 2: 256-QAM, and curves 3: 4096-QAM. For decoding, the sum-product algorithm is employed, with the number of iterations limited to 20.*

Note that in the case of uncoded M-QAM transmission $\eta = \log_2 M$. Also, for a capacity-achieving scheme that transmits C b/symbol, $\eta = C$ and SNR $= 2^C - 1$, which implies that $SNR_{norm} = 1$ (0 dB).

Turbo Coding

Various coding and decoding approaches according to the turbo coding principle introduced in [3] have been proposed in the literature. In particular, parallel or serially concatenated convolutional codes, repeat accumulate codes, and various combinations thereof have been shown to approach the capacity of the AWGN channel. The common features of all these approaches is the interleaving and deinterleaving functions as well as the soft-input soft-output APP decoder usually implemented by the BCJR algorithm. For an overview, see [6, references therein].

In this article a turbo coding scheme with two component codes is used, similar to the architecture introduced in [3]. Figure 4 shows its SER performance with two different interleaver lengths and 16, 256, and 4096-QAM. For QAM symbol mapping, one systematic bit and one parity bit are used in each dimension, and the remaining bits are uncoded. The unused parity bits are punctured. The resulting code rates for 16, 256, and 4096-QAM are 0.5, 0.75, and 0.833, respectively. These differences in code rates lead to different SER performance of the turbo codes depending on the constellation size. Note that it is also possible to encode all the information bits (i.e., no uncoded bits are used for symbol mapping). The advantage of this technique [11] is its immunity to impulse noise, but it is computationally more complex.

LDPC Coding

Binary LDPC codes have been known since the early 1960s but their capacity-approaching performance has been discovered only in the past decade. There is currently intensive research activity in deterministic constructions of such codes as well as in understanding their theoretical limits [6]. It appears that high-rate LDPC codes with medium block length, whose parity check matrices are constructed similarly to those of array codes [12], exhibit as good performance as random LDPC codes. Array-code-based LDPC coding was shown in [7] to offer a number of advantages for DSL transmission. The LPDC parity-check matrix is specified, in that case, by a small set of parameters and constructed deterministically without requiring preprocessing operations. Furthermore, array LDPC codes are amenable to linear time encoding.

Figure 5 shows the SER performance of two array-code-based LDPC codes of block lengths $N = 529$ and 4489 bits, and code rates 0.870 and 0.940, respectively. For 16-QAM symbol mapping, all the bits are coded. For 256 and 4096-QAM, there are three coded bits along each dimension, and the remaining bits are uncoded.

Latency

As mentioned earlier, latency is an important issue in DSL systems. Generally speaking, if higher latencies can be tolerated, longer and hence more powerful codes can be employed. Conversely, lower coding gains are imposed by small latencies.

Recall that DMT frames in ADSL systems are generated at the rate of 4000 Hz. Therefore, if one information block is encoded into one DMT frame, the encoding and decoding functions introduce a latency of 250 μs each, resulting in a total latency of 0.5 ms. When a codeword spans more than a single frame, latency increases accordingly.

Figure 6 shows the net coding gain as a function of latency for some turbo and LDPC coding schemes. To obtain the results, a simplified DMT system was assumed in which each DMT frame is transmitted via 100 or 200 subchannels. Furthermore, either 16- or 4096-QAM is used over all the subchannels. Clearly, the number of subchannels, constellation size, number of uncoded bits, and latency determine the codeword length. For example, 200 subchannels together with 16-QAM implies that 200×4 bits = 800 bits are carried per DMT frame. If all the bits correspond to code bits, a codeword of length $N = 800$ is carried with minimum latency (0.5 ms). If the latency is doubled, a codeword can be carried by two consecutive DMT frames, and the code length can be increased to twice that value. In our study, the latency increases linearly with respect to interleaver size or LDPC codeword size.

It is seen from Fig. 6 that good coding gains are achieved even for very tight latency constraints. For a latency of 0.5 ms, the simulated turbo coding scheme provides a coding gain of up to 5.7 and 5.1 dB for 16- and 4096-QAM, respectively. Once the latency restriction is relaxed, higher coding gains can be achieved. The gain for turbo coding can be further improved by increasing the interleaver length or the number of states for each component code (from the eight states presented in this article to 16 or 32 states). Likewise, for a latency of 0.5 ms, LDPC coding provides a coding gain of up to 5.0 dB and 4.8 dB for 16- and 4096-QAM, respectively. By increasing the codeword length to encompass more than one DMT frame, additional coding gains are realized.

Note that the TCM scheme defined in the ADSL standards achieves a net coding gain of 4.4 and 4.2 dB for 16- and 4096-QAM, respectively, at an SER of 10^{-7}. The average rate is equal to 3.5 b/2-D symbol for 16-QAM and 11.5 b/2-D symbol for 4096-QAM. The encoding and decoding operations extend over one frame period each, so latency amounts to 0.5 ms. Hence, turbo and LDPC codes generally exhibit their full advantage when the code length, and hence latency, is increased.

A word of caution is needed at this point. The objective of Fig. 6 (or Figs 4 and 5) is not to compare turbo and LDPC codes because they mostly operate at different spectral efficiencies, a fact that is apparent from Fig. 2. A one-to-one performance comparison would require several parameters, such as code rate, latency, and implementation complexity, to be kept equal. It is only noted here that code rate and coding gain can often be traded off for DSLs through the bit loading process. That is, for a specified minimum operating margin, a particular data rate that can be achieved with a specific code-rate/coding-gain combination can also be achieved, for example, by increasing the code rate and decreasing the coding gain, or vice versa.

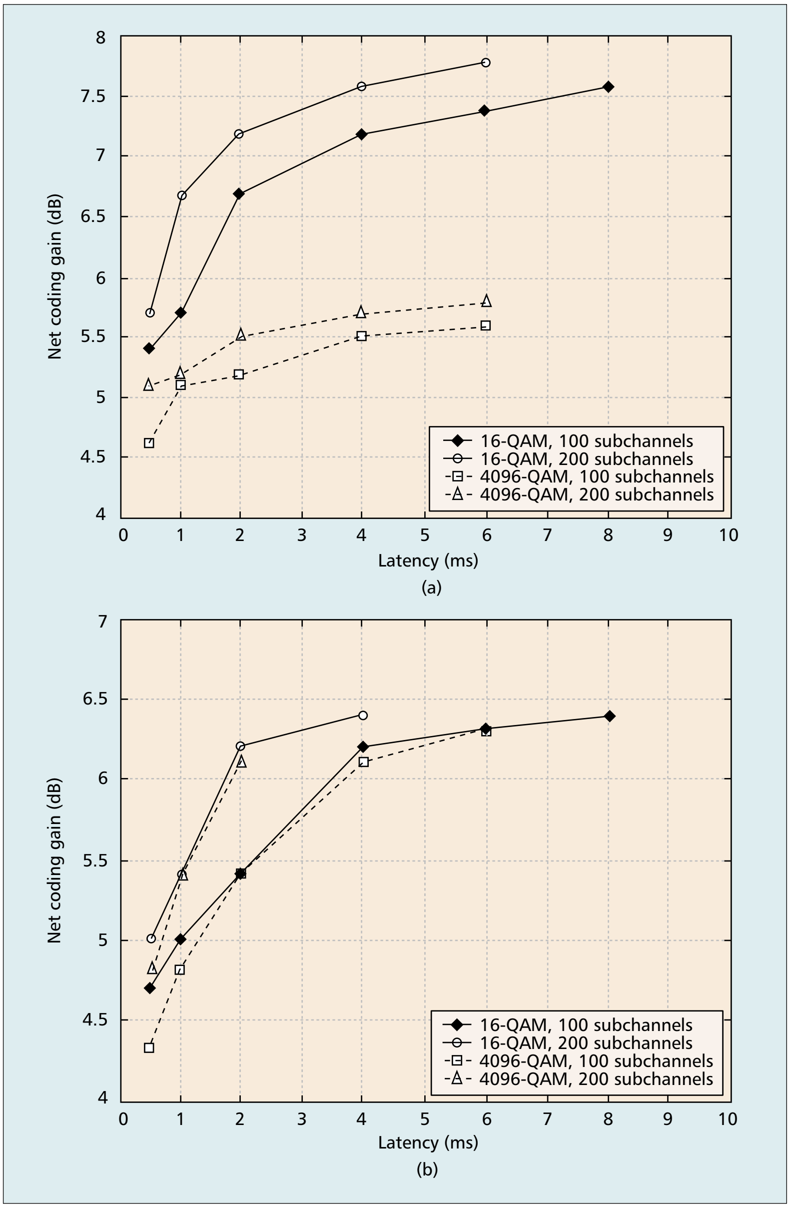


Figure 6. *Net coding gains as a function of latency achieved at a SER of 10^{-7} by a) turbo coding; b) LDPC coding for 16 and 4096-QAM and a total number of 100 or 200 subchannels. No outer RS coding is included. Some of the points in the plots are omitted because computation times tend to become prohibitive for these cases. For turbo coding, the same code is used for the different latency cases; only the interleaver size is changed. The maximum interleaver size is 4800 bits, and the rates are 2 and 10 b/symbol with 16- and 4096-QAM, respectively. For LDPC coding, codes with different lengths and rates are used for the different cases. The code rates are chosen in the range of 0.82–0.95, so the rates vary in the range of 3.31–11.65 b/symbol. The maximum code length is 7200 bits.*

Complexity

For encoding, it can be assumed that the computational complexity is essentially identical for turbo coding and TCM. For LDPC coding, if the code word is obtained by multiplying the information block with the generator matrix of the code, encoding requires $\mathcal{O}(N^2)$ operations, where N is the length of the code. However, the family of LDPC codes proposed in [7] and used in the preceding section enjoys the desirable property of linear time encodability according to which

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encoding requires $\mathcal{O}(N)$ operations. Indeed, it was shown for this case that the complexity of LDPC encoding typically amounts to three times the encoding complexity of TCM.

As mentioned, there are two basic algorithms for decoding turbo and LDPC codes: the BCJR algorithm and the sum-product or belief propagation algorithm, respectively. To minimize the number of multiplications in a practical implementation, it is advantageous for both algorithms to compute and propagate messages that represent log-likelihood ratios. In both cases, simplified algorithms exist aimed at lowering the implementation complexity at the cost of some loss in performance. A comparison of the complexity of the various decoding algorithms would, however, exceed the scope of this article. Nevertheless, a generally accepted fact is that LDPC decoding by the sum-product algorithm is computationally less complex than turbo decoding by the BCJR algorithm.

Finally, it should be mentioned that turbo coding and LDPC coding techniques may have stringent memory requirements, especially for long codes. This is an important aspect in the design of DSL transceivers.

CONCLUSIONS

Capacity-approaching coding techniques can provide additional coding gains to the coding schemes used in current DSL standards. It was shown that this coding gain is a valuable resource for increasing the data rate and/or loop reach, which can be instrumental in optimum usage of the local loop and widespread deployment of DSL services. Two practical approaches based on turbo and LDPC coding were presented. The possibility of incorporating both turbo and LDPC coding capability into ADSL systems, a potentially interesting approach, was also discussed. We have not attempted to provide a one-to-one comparison of these two approaches because in general code parameters, encoding and decoding complexity, as well as other factors are different. The main conclusion is that both techniques appear to be practical for implementation with reasonable increase in transmitter/receiver complexity. It is expected that capacity-approaching coding techniques, such as those described in this article, will soon find their way into future generations of DSL modems and cable transmission systems in general.

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BIOGRAPHIES

EVANGELOS ELEFTHERIOU [F'02] received a Diploma in electrical engineering from the University of Patras, Greece, in 1979, and M.Eng. and Ph.D. degrees in electrical engineering from Carleton University, Ottawa, Canada, in 1981 and 1985, respectively. He joined the IBM Zurich Research Laboratory, Rüschlikon, Switzerland, in 1986, where he worked on various projects related to wired and wireless communications as well as magnetic recording. He currently manages the storage technologies group there. His primary research interests lie in the areas of communications and information theory, particularly signal processing and coding for transmission and recording systems. Recently his research activities have expanded into the area of nanotechnology, in particular probe-storage techniques. He holds over 30 patents (granted and pending applications). He was Editor of *IEEE Transactions on Communications* from 1994 to 1999 in the area of Equalization and Coding, and Guest Editor of the IEEE Journal on Selected Areas in Communications special issue The Turbo Principle: From Theory to Practice.

SEDAT ÖLÇER [SM] received a Diploma in electrical engineering and a Ph.D. degree from the Swiss Federal Institute of Technology, Lausanne (EPFL), in 1978 and 1982, respectively. From 1982 to 1984 he was a research associate at the Information Systems Laboratory of Stanford University, California, and Yale University, New Haven, Connecticut. In 1984 he joined the IBM Zurich Research Laboratory, Rüschlikon, Switzerland. He participated in and contributed to the work of several standards bodies including ITU-T, IEEE802, ATM Forum, and ANSI T1E1. His research interests are in digital communications, signal processing, and coding, with applications to broadband network access and storage systems. He was Guest Editor for a special issue of *Computer Networks* on Broadband Access Networks in 1994, and for the June 2002 issue of *IEEE Journal on Selected Areas in Communications* entitled "Twisted Pair Transmission — Ever Increasing Performances on Ancient Telephone Wires." He also served as Guest Editor for *IEEE Communications Magazine* Feature Topics on Very High-Speed Digital Subscriber Lines (2000) and Capacity Approaching Codes, Iterative Decoding Algorithms, and Their Applications (2003).

HAMID R. SADJADPOUR received his Ph.D. degree from the University of Southern of California in 1996. After graduation, he joined AT&T Laboratory as a member of technical staff, senior technical staff member, and finally principal member of technical staff at AT&T Research Labs, Florham Park, New Jersey until 2001. In fall 2001 he joined the University of California, Santa Cruz (UCSC) as an assistant professor. He has served on the technical program committees of ICC 2003 and RAWCON 2004, and as guest editor of a special issue, Multicarrier Communications and Signal Processing, for *EURASIP* in 2003. His research interests include communication theory and signal processing, coding theory for wireless, wired, and fiber optic applications, space-time processing for wireless systems, turbo codes, and signal processing for mobile ad hoc networks.