Adaptive Modulation and Channel Coding Using Reliability Information

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Abstract— **A classical strategy for optimizing digital communication systems is the *minimization* of the error rate. In this paper we propose a new paradigm: Our main goal is the *estimation* of the error rate. This alternative strategy offers us (i) to use iterative demodulation and decoding techniques (e.g. turbo codes), (ii) to optimize adaptive modulation and adaptive source and channel coding techniques (e.g. adaptive multi-rate speech codecs) with respect to the average data throughput given a target error rate, and (iii) motivates a "soft-decision" Monte Carlo simulation technique. These three principles are applied here to obtain an adaptive mobile radio system operating near Shannon's capacity bound. Furthermore, the reliability information (i) may be used for blind detection of the modulation and coding scheme, (ii) may be used as a stop criterion for iterative processing, and (iii) makes error detection coding unnecessary.**

Keywords— **Adaptive modulation, mobile radio, APP decoding, reliability information, Monte Carlo simulation, soft-decision simulation.**

I. Introduction

As opposed to wireline modems, where adaptive modulation and channel coding is state-of-the-art, first and second generation mobile radio systems are non-adaptive. In a few classical papers, see e.g. [1]-[3], as well as numerous recent publications, see e.g. [4]-[10], it was demonstrated that adaptive modulation and channel coding systems offer a significant throughput and/or power gain compared to non-adaptive transmission, particularly when the channel conditions are slowly time varying and a reliable feedback channel is available. As a result of these efforts, wireless radio systems employing adaptive modulation and channel coding are soon to be expected [11].

To our best knowledge, however, the level switching criteria that have been proposed are either ad-hoc or depend on a particular channel model[1], and quality control information was generally supposed, which increases the overhead.

In this tutorial, we propose and investigate a rather natural level switching criterion based on reliability information. Intentionally, we skip all math. We propose a pragmatic approach using single or multi-carrier modulation (e.g. OFDM) and channel coding (e.g. turbo codes), optionally without sending additional quality control information. Given a TDD system with low-delay feedback link, our adaptive transmission scheme (i) promises a performance reasonably close to Shannon's capacity bound and (ii) guarantees a target bit error rate profile. The scheme may be well combined with an adaptive multi-rate speech codec, for example.

The main idea is to use reliability information at the output of an a posteriori probability (APP) demodulator, APP equalizer, or APP channel decoder in order to *estimate* the (bit, symbol, or block) error rate [12]. This estimate, typically computed once per data block, is then compared with a pre-selected target (bit, symbol, or block) error rate. If the actual error rate is less than the target error rate, the size of the signal alphabet, M, is increased in the next block, otherwise it is decreased. (Of course, thresholds may be defined to prevent from level flipping and an error rate predictor may be applied to improve outdated information.) As a side effect, the reliability information may be used as a stop criterion in iterative demodulation or channel decoding, and the reliability information may be used to detect the actual size of the signal alphabet, i.e. a blind mode (in the sense that no quality control information is necessary) is possible. The proposed scheme is suitable for frequency-flat channels as well as frequency-selective channels. Another side effect of the availability of reliability information is that error detection coding (as opposed to error correction coding) is not necessary.

As a design study, we have modified a GSM burst.

[1]The presumably most prominent level switching criterion, the SNR criterion, fails for example in Rician channels with a varying or unknown Rice factor, or in frequency-selective channels with time-varying statistics.

The basic signal structure (tail symbols, midamble, guard time, number of data symbols, symbol duration) is chosen to be exactly the same as in GSM. The binary data symbols used in GSM, however, are substituted by M-ary QAM symbols, where $M = 2, 4, 8, \ldots$. Therefore, the average data throughput can be significantly improved given the same bandwidth.

II. FUNDAMENTALS

In this section we review some recipes required for our system concept.

A. Pragmatic Trellis-Coded Modulation

Consider the known power/bandwidth diagram in Fig. 1, where we have plotted the bandwidth efficiency versus the required signal-to-noise ratio per information bit, E_b/N_0, in order to achieve an average target bit error rate of 10^{-5} on the AWGN channel given various linear modulation schemes with Sinc pulse shaping. It is interesting to note that the loss of M-QAM with respect to the capacity bound is about 7.5 dB for all M. This gap may be closed with trellis-coded modulation (TCM) [13]. A simple form of TCM is the use of a rate R convolutional code followed by a short bit-by-bit interleaver and a linear M-ary modulator, see Fig. 2. R and M are chosen so that the desired effective number of bits/symbol, $R \cdot \mathrm{ld}\, M$, is obtained. This so called pragmatic trellis-coded modulation scheme [14] may be refined by using a standard rate R turbo code substituting the convolutional code [15]. Pragmatic turbo TCM offers a performance reasonably close to the capacity bound, typically within a few dB, although the encoder and the classical iterative decoder are suboptimal.

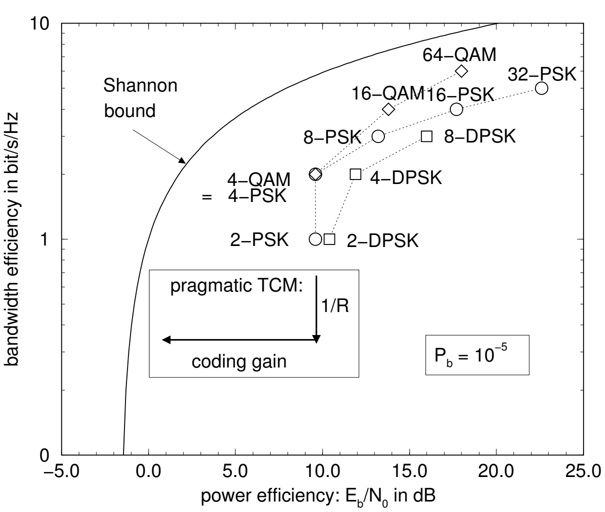

Fig. 1. Power/bandwidth diagram for linear modulation schemes (AWGN channel, Sinc pulse).

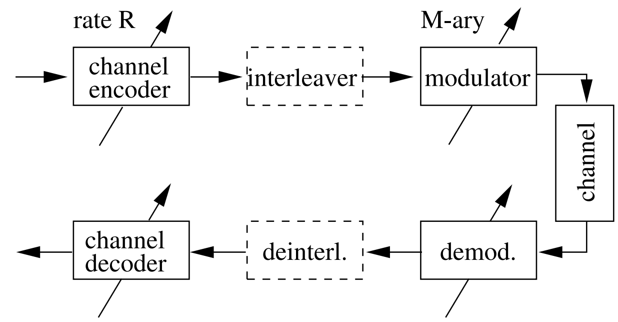

Fig. 2. Pragmatic TCM.

The main difference between pragmatic TCM and coset codes, used e.g. in [8], is the absence of uncoded bits in case of pragmatic TCM.

B. APP Demodulation, APP Equalization, and APP Decoding

The APP demodulator, APP equalizer [16], or APP decoder [17] is a demodulator, an equalizer, or a channel decoder, respectively, which computes the a posteriori probabilities for each individual information bit (or symbol) or channel bit (or symbol), respectively, given the entire input sequence and statistical side information. This implies knowledge of the structure and the statistics of the equivalent channel model. Instead of passing on probabilities, one may also output likelihood values or log-likelihood values. In the context of fading, it is advantageous to perform all operations in the log-domain because of numerical stability [18].

APP algorithms may be interpreted as optimal bit (or symbol) *error rate estimators*. So far, error rate estimators are applied e.g. in the context of iterative (turbo) decoding in order to pass reliability values between constituent decoders. In our context of adaptive schemes, we use an APP algorithm (or approximations thereof) to estimate the actual error probability (on a burst-by-burst basis) in order to determine an optimized number of info bits per symbol. This affects the transmitter side as well.

C. Soft-Decision Monte Carlo Simulation

In a classical Monte Carlo (MC) simulation, the transmission errors are counted in order to obtain an unbiased estimate of the error rate. This implies a reliable reference channel and a sufficient number of independent errors.

If we are able to build an APP algorithm, we can obtain an unbiased estimate of the error rate by averaging the error probabilities computed by the APP algorithm. This improved simulation technique we call "soft-decision" MC simulation [12]. Since the errors are not counted as integers any more, the variance of the estimation error is less for "soft-decision" MC simulation, see Fig. 3 (from [12]). Furthermore, no reference channel is needed.

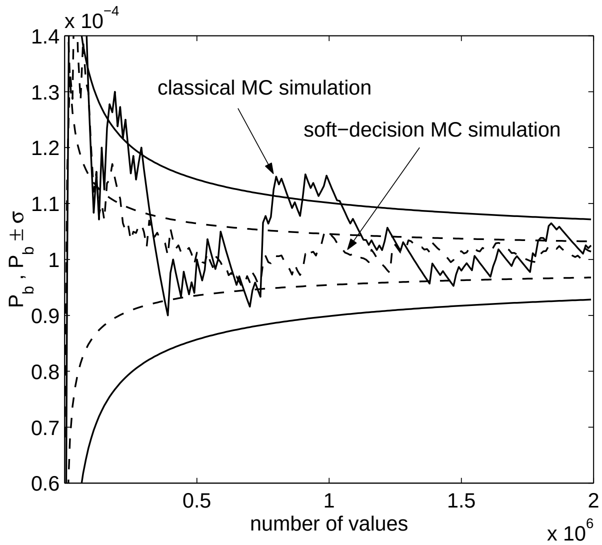

Fig. 3. Classical MC simulation versus soft-decision MC simulation. The target bit error rate (BER) is 10^{-4}. The smooth curves feature the 1σ confidence interval, the other two curves show sample examples.

So far, we have argued that soft-decision MC simulation is particularly useful for the simulation of rare error events. Now, we additionally argue that soft-decision MC simulation is useful as an error indicator, for example in adaptive schemes.

III. System Concept

Our goal is to maximize the average data throughput given a target error rate. The system model to approach this goal is shown in Fig. 4.

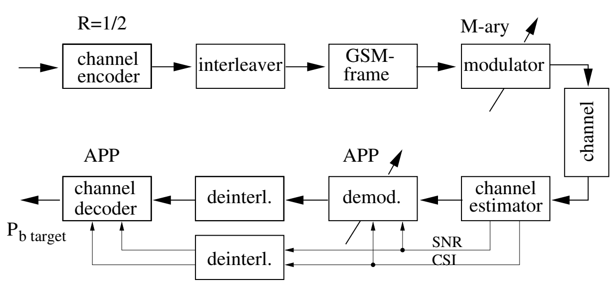

Fig. 4. Block diagram of the proposed transmission scheme. An APP equalizer is necessary in case of frequency-selective fading or partial-response signals, which is not shown here.

As a design study, we have modified a GSM burst as shown in Fig. 5. The basic signal structure (tail symbols, midamble, guard time, number of data symbols, symbol duration) is chosen to be exactly the same as in GSM. Hence, in conjunction with linearized Gaussian minimum shift keying (GMSK) pulse shaping, the Tx power density spectrum is very similar to the power density spectrum of a GSM transmitter. The binary data symbols used in GSM, however, are substituted by M-ary QAM symbols, where $M = 2, 4, 8, \ldots$, in order improve the average data throughput given the same bandwidth. (The EDGE concept [11] is very similar in this respect.) The prize to pay with M-QAM is a non-constant envelope, and frequency hopping is prohibited.

Concerning channel coding, we used a rate 1/2 convolutional code, which may be substituted by a turbo code. Low rate codes ($R < 1/2$) were obtained by repetition. In conjunction with an interleaver (in order to spread burst errors at the decoder output), the modulation/coding scheme is a pragmatic TCM scheme. A lower limit on R are synchronization constraints, whereas hardware constraints (such as linearity, quantization, etc.) provide an upper limit on M.

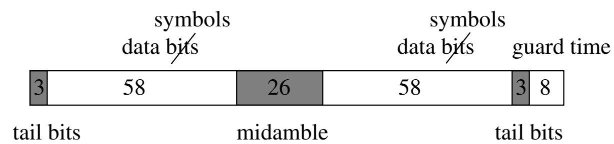

Fig. 5. Modified GSM burst.

The signal flow diagram illustrating the adaptation strategy is shown in Fig. 6. The presence of a perfect feedback channel is assumed. Soft-decision bit error rate estimation is performed e.g. once per GSM burst by means of an APP decoder. The threshold ϵ is subject for optimization. An extension to unequal error protection is straightforward, but any reduction of the number of samples results in a higher variance of the BER estimates, see Fig. 3.

The design study is based on a single carrier system. However, the principle idea is also well suited for multi-carrier systems such as orthogonal frequency-division multiplexing (OFDM). In ODFM, each subcarrier may be individually modulated according to the "water-filling" principle. Some subcarriers may be intentionally even left unmodulated, depending on the fading or noise spectrum or on the user requirements.

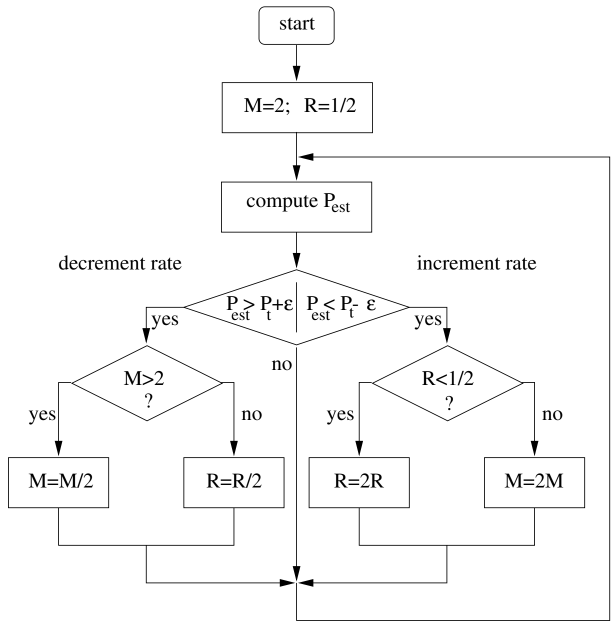

Fig. 6. Signal flow diagram.

IV. System Evaluation

Compared to the SNR level switching criterion, APP algorithms are much more robust against SNR estimation errors, and are capable of indicating other sources of impairments, such as sync errors, etc.

The system functionality will be/was presented at the workshop in terms of a Ptolemy[2] demonstration. A detailed system evaluation is subject for future research.

V. Conclusions

As a design criterion for digital transmission systems the paradigm of estimating (instead of minimizing) the error rate is proposed. This criterion rather naturally motivates adaptive modulation schemes and adaptive source and channel coding. The optimal bit (or symbol) error rate estimator is an APP algorithm. Possible application scenarios include slowly time-varying channels and channels with different "states" (such as good or bad, in the simplest case). Despite capacity arguments [7], adaptive transmission schemes are likely to benefit from different user requirements.

Open questions include a detailed system analysis including a non-perfect feedback channel with errors and delays, blind adaptation, and implementation aspects. One of the most demanding practical challenges appears to be the computation of reliable error estimates, and the handling of variable data rates.

[2] http://ptolemy.eecs.berkeley.edu

References

[1] J. Hayes, "Adaptive feedback communications," *IEEE Trans. Commun. Techn.*, vol. COM-16, pp. 29-34, Feb. 1968.

[2] J. Cavers, "Variable-rate transmission for Rayleigh fading channels," *IEEE Trans. Commun. Techn.*, vol. COM-20, pp. 15-22, Feb. 1972.

[3] V. Hentinen, "Error performance for adaptive transmission on fading channels," *IEEE Trans. Commun. Techn.*, vol. COM-22, pp. 1331-1337, Sept. 1974.

[4] S. Sampei, S. Komaki, and N. Morinaga, "Adaptive modulation/TDMA scheme for a large capacity personal multimedia communication system," *IEICE Trans. Commun.*, vol. E77-B, pp. 1096-1103, Sept. 1994.

[5] W.T. Webb and R. Steele, "Variable rate QAM for mobile radio," *IEEE Trans. Commun.*, vol. 43, pp. 2223-2230, July 1995.

[6] J.M. Torrance and L. Hanzo, "On the upper bound performance of adaptive QAM in a slow Rayleigh fading," *IEE Elec. Letters*, pp. 169-171, Apr. 1996.

[7] A.J. Goldsmith and S.G. Chua, "Variable rate variable power MQAM for fading channels," *IEEE Trans. Commun.*, vol. 45, pp. 1218-1230, Oct. 1997.

[8] A.J. Goldsmith and S.G. Chua, "Adaptive coded modulation for fading channels," *IEEE Trans. Commun.*, vol. 46, pp. 595-602, May 1998.

[9] D.L. Goeckel, "Adaptive coding for time-varying channels using outdated fading estimates," *IEEE Trans. Commun.*, vol. 47, pp. 844-855, June 1999.

[10] S. Hu, A. Duel-Hallen, and H. Hallen, "Adaptive modulation using long range prediction for flat Rayleigh fading channels," in *Proc.* 2000 *IEEE Int. Symp. Inform. Theory*, Sorrento, Italy, p. 159, June 2000.

[11] A. Furuskär, S. Mazur, F. Müller, and H. Olofsson, "EDGE: Enhanced data rates for evolution of GSM and TDMA/136," *IEEE Personal Commun. Mag.*, vol. 6, no. 3, pp. 56-66, June 1999.

[12] P. Hoeher, U. Sorger, and I. Land, "Log-likelihood values and Monte Carlo simulation," in Proc. *Int. Symp. on Turbo Codes & Related Topics*, Brest, France, Sept 2000.

[13] G. Ungerboeck, "Channel coding with multilevel/phase signals," *IEEE Trans. Inform. Theory*, vol. 28, no. 1, pp. 55-67, Jan. 1982.

[14] A.J. Viterbi, J.K. Wolf, E. Zehavi, and R. Padovani, "A pragmatic approach to trellis-coded modulation," *IEEE Commun. Magazine*, pp. 11-19, July 1989.

[15] S. Le Goff, A. Glavieux, and C. Berrou, "Turbo-codes and high spectral efficiency modulation," in *Proc. IEEE ICC* '94, New Orleans, Louisiana, pp. 645-649, May 1994.

[16] K. Abend and B.D. Fritchman, "Statistical detection for communication channels with intersymbol interference," *Proc. of the IEEE*, vol. 58, no. 5, pp. 779-785, May 1970.

[17] L.R. Bahl, J. Cocke, F. Jelinek, and J. Raviv, "Optimal decoding of linear codes for minimizing symbol error rate," *IEEE Trans. Inform. Theory*, vol. IT-20, pp. 284-287, Mar. 1974.

[18] P. Robertson, P. Hoeher, and E. Villebrun, "Optimal and suboptimal maximum a posteriori algorithms suitable for turbo-decoding," *European Transactions on Telecommun. (ETT)*, vol. 8, no. 2, pp. 119-125, Mar./Apr. 1997.