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Edited by
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# Book of Abstracts
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# on
# Information and Communication Theory

**Eindhoven, The Netherlands**

**September 3–5, 1989**

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# Preface

As a result of the contacts between scientists in the Benelux and Japàn the first joint workshop on Information and Communication theory has been organized.
Although Japan and the Benelux are half a world away from each other, this workshop shows that scientists do not suffer really from this distance.
I hope that the workshop contributes to a more intensive cooperation and that discussions between industrial researchers and scientists are very valuable.
The workshop is supported by the IEEE Benelux section, FUJI Photo Film Tilburg and the Benelux werkgemeenschap voor Informatie– en Communicatietheorie.
Paul de Bot assisted in the organization of the workshop.
At last, but not at least, I want to thank Kees Schouhamer–Immink for his interesting and stimulating conversations concerning the organization of the workshop.

Han Vinck, August 1989.

# Partial–Response Decoding of Rate 1/2 Modulation Codes for Digital Storage

Jan Bergmans[1], S.Mita[2] and M. Izumita[2]

In digital storage systems, receivers for rate 1/2 modulation codes are usually oversampled by a factor of 2 with respect to the data stream that they attempt to reconstruct. In this lecture we show that oversampling may be avoided by using partial–response techniques to detect instead of the encoded binary signal a decimated ternary one, from which the original data can be recovered by means of a simple decoder. A method is described to find all such decoders for a given rate 1/2 code. Examples treated are FM, MFM, Miller–Squared, (2,7), 3PM and Quad–Phase. The mean–square performances of the new reception schemes are analyzed and compared with their predecessors for the classical Lorantzian channel model for magnetic storage. The relative merits are found to depend quite heavily on the information density, favouring some of the new schemes at high densities.

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# Erasure assignment for Maximum Distance Separable Codes

Paul G.M. de Bot[1]

Decoding algorithms of most Maximum Distance Separable (MDS) codes are able to handle erasures in an efficient way. Using soft–decision information, one can assign erasure flags to unreliable symbols. It is possible to derive optimal criteria for evaluating the reliability of symbols. However, the decision whether a symbol has to be erased or not does not only depend on the reliability of the symbol itself, but also on the reliability of the other symbols of its codeword. Several strategies can be distinguished with one or more decoder passes and with typical ways for each pass to determine the number of erasures in a codeword as a function of the reliability of the symbols. Using product codes, one needs combinations of strategies to perform error & erasure decoding on both columns and rows. Unless some strategies can be proven to be optimal in the asymptotic case ($E_b/N_0 \to \infty$), in practice some other strategies have better performances. Selection of a strategy depends further on decoding speed and complexity constraints.

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[1] G. D'Aria, "Reed–Solomon Error and Erasure Correction for PAM/FM Mobile Radio Systems," Proc. 38th IEEE Veh. Techn. Conf., Philadelphia, pp. 485–488, June 15–17, 1988.

[2] P.G.M. de Bot, "An Error–Correcting Coding Scheme for Digital Audio Broadcasting," graduate report, Eindhoven University of Technology, Department of Electrical Engineering, February 1989.

[1] Eindhoven University of Technology, POB 513, 5600 MB Eindhoven, The Netherlands

# Error Correction on digital video recording

Ad Denissen[1]

Digital recording of video on magnetic tape with a helical scan recorder needs for two reasons strong error correction. First the playback signal is corrupted with single errors (caused by noise) and burst of errors (caused by imperfections and scratches on tape). Secondly the digital information is compressed to store at least 2 hours high–quality video on one cassette. And due to this compression the digital data is sensitive to errors and hard to conceal.

The aim of an error correction system is enlarging the mean time between two uncorrectable frames to about 1 hour, so that the playback video signal hardly contains any annoying errors. Reed–Solomon product codes can correct single and burst errors with an acceptable amount of overhead and hardware complexity. And worst case performance estimations leads to the conclusion, that it is possible to protect recorded digital video with Reed–Solomon product codes.

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# The performance of asynchronous frequency–hopped multilevel frequency–shift keying spread–spectrum multiple–access systems

Uwe Fiebig[1]

Time–frequency (TF) coded or frequency–hopped multilevel frequency shift keying (FH/MFSK) systems for use in asynchronous spread–spectrum multiple–access (SSMA) systems are analyzed. These systems provide quite high system spectral efficiency in comparison with common SSMA techniques. In contrast to previous investigations where synchronous multiple access (MA) is considered, no coordination in timing among the simultaneous users is assumed.

The results of the MA system capabilities, bit–error rates and system spectral efficiency are presented and compared with the synchronous systems.

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# Universal coding for data compression with side information

Te Sun Han[1]

The problem of universal data compressions with specified rates is of great significance from the engineering point of view, because it is concerned with the stability or reliability of the operating systems.

Such an problem has first been investigated for the Slepian–Wolf system [1], by Csiszar and Korner [2] along with the specification of an achievable error exponent at rates in the SW–region, and recently also by Han and Ohama [3] who have determined the optimal exponent at rates outside the SW–region. However, the problem for other kinds of data compression systems seem to be more involved, because it can not help the intervention of several auxiliary random variables in the specification of the achievable rates.

We consider here, as an illustrative case of those systems with auxiliary random variables, the universal coding for the Wyner–Korner–Ahlswede system [4], [5], and shows a universal encoding/decoding scheme, thereby establishing a universally attainable error exponent at rates in the WAK–region. The technical point is to consider joint types of not only source sequences but also auxiliary sequences. Our approach can immediately be applicable to a general class of data compression systems including all the previously studied systems with or without distortion criterion (of Csiszar and Korner [6], and Han and Kobayashi[7]).

It should be noted that this kind of universal codings plays an fundamental role in investing the effective parameter estimation (and also composite hypothesis testing) in the multiterminal framework (of Zhang and Berger[8], Han and Kobayashi[9], and Han and Amari[10].

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[10] T.S. Han and S. Amari, "Multiterminal estimation theory," presented at IEEE/CAM Workshop on Information Theory, Cornell Univ., Ithaca, Jun. 25–29, 1989.

[1] Dept. of Inf. Systems, Senshu Univ., 2–1–1 Higashimita, Tama–Ku, Kawasaki 214, Japan.

# Some aspects of a jointly Markov process when a marginal process is Markovian

Kingo Kobayashi[1]

A marginal process $\{X_i\}$ of a jointly Markov process $\{X_iY_i\}$ is not necessarily Markovian. A non–trivial sufficient condition, which will be necessary for most cases, is

$$M_{ab}M_{bc}\mathbf{1} = M_{ab}\mathbf{1}\cdot\frac{\mathbf{u}}{|\mathbf{u}|}\cdot M_{bc}\mathbf{1}$$

for any a,b $\in \mathscr{X}$ (the alphabet of process $\{X_i\}$), where $M_{ab}$'s are submatrices of the joint Markov matrix $M = (M_{ab})$ corresponding to the partition induced by the alphabet $\mathscr{X}$, and $\mathbf{u} = (u_a)$ is the stationary distribution of the joint Markov process.

We deduce several properties from this equation and discuss the geometrical interpretation of these properties.

[1] Osaka University, Japan

# On Variable Length quantization Codes

Hiroyoshi Morita[1] and Te Sun Han[2]

We consider the quantization problem of a random real number X. The real line R is assumed to be partitioned into N intervals $I_i$, $i = 1,2,\ldots,N$. Let denote a representative point of the interval $I_i$ as $q_i$. A quantizer consists of these N intervals and their representative points. The distorsion of the quantizer is measured as

$$D = \sum \int_{x \in I_i} (x - q_i)2p(x)dx,$$

where p(x) is the probability density on R. When the encoder (or quantizer) receives a real number X, it finds an interval which includes X and outputs its index. Obtaining this index, the decoder outputs the representative point of the interval as a reproduction of X.

Let us consider the following two types of quantization:

1. To minimize D over all quantizers with N fixed,
2. To minimize D subject to a constraint on the quantizer output entropy $H_Q = -\Sigma\, p_i \log(p_i)$, where $p_i = \mathrm{Prob}\{X \in I_i\}$.

For an asymptotic case such that N is very large and overload distortion is negligible, the optimal bounds have been already obtained by Gish & Pierce, Zador and Gersho. However, it is difficult to obtain optimal bounds for a moderate distortion.

To obtain an efficient coding scheme even for a moderate distortion, we propose a variable quantization scheme with arithmetic codes. The main idea of our approach is to represent each $q_i$ with certain finite digits on a given numerical notation, say a binary notation and to index each interval $I_i$ by $q_i$. The length of $q_i$ is determined as to be approximately proportional to $-\log(p_i)$ where $p_i = \mathrm{Prob}\{X \in I_i\}$.

The $q_i$ can be encoded by arithmetic codes as follows: Since the encoder knows the probability density, the probability $p(q_i)$ can be calculated. Now we represent $p(q_i)$ as $p(q_{i,1})\Pi_{j>1} p(q_{i,j} | q_{i,1},\ldots,q_{i,j-1})$, where $q_{i,j}$ is the value of the j–th digit of $q_i$. That is, a


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random variable X is considered as a random process of digits of a given random deal. The process is a nonstationary Markov source with a finite alphabet. We adopt an arithmetic coding method to encode this process so that the obtained codeword of $q_i$ is nearly equal to $-\log(q_i)$. As a result, each interval will be encoded with variable length.

The advantage of the strategy of using arithmetic codes is to directly obtain the representative points as an output of the arithmetic codes. It is not necessary to prepare the huge lookup table which gives a corresponding representative point for each index. The proposed method provides a practical implementation of quantizers with entropy coding.

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# Four cases for write unidirectional memory codes over arbitrary alphabets

W.M.C.J. van Overveld[1]

Consider a write–unidirectional memory (WUM) with N cells, over alphabet $\{0,1,..,q-1\}$. We store information using a WUM code with parameters $(N,M_0,M_1,..,M_{q-1})$. Let $k = c \bmod q$ $(0 \leq c \leq q-1)$ and consider the $k^{th}$ time of writing. Let $\underline{s}_k$ be the state of the WUM before writing, $w_k \in \{0,1,..M_c-1\}$ the message and $\underline{x}_k \in \{c,\square\}^N$ the codeword. The new state $\underline{s}_{k+1}$ satisfies

$\underline{s}_{k+1}(n) = c$ if $\underline{x}_k(n) = c$, and

$\underline{s}_{k+1}(n) = \underline{s}_k(n)$ if $\underline{x}_k(n) = \square$ $(n = 1,2,..,N)$.

Depending on whether encoder and/or decoder know the state $(\underline{s}_k)$ of the disk before writing, we have four different types of WUM codes. Therefore we distinguish:

1: both users know $\underline{s}_k$

2: only the encoder knows $\underline{s}_k$

3: only the decoder knows $\underline{s}_k$

4: neither user knows $\underline{s}_k$

For each case we define the rate tuple of the WUM code as

$$R = (R_0,R_1,..,R_{q-1}),\ R_c := \frac{\log(M_c)}{N} \text{ for all } c,$$

and, as usual, we define the capacity region as the set of achievable rate tuples. Hence we have four capacity regions: $C_1$, $C_2$, $C_3$ and $C_4$. In this paper we derive exact expressions for $C_1$, $C_2$ and $C_3$; furthermore we have a partial result for $C_4$.

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# The foundation of speech coding systems with linear predictors

J. Rooyackers[1]

A signal model is introduced, that forms the background of all signal and speech processing systems using linear predictors. The optimal predictor is treated and a fast algorithm is described.

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# Notes on Weight Distribution of R = k / n convolutional codes

Hiroshi Sasano[1] and Masao Kasahara[2]

Nowadays convolutional codes are extensively used on the various communication channels. Weight distribution of codewords of a convolutional code is an important parameter of the code. Unfortunately, for $R = k / n$ convolutional codes, it is not easy to obtain the weight distribution using two types of method. The first method, which is presented for calculating the weight distribution of $R = k / n$ convolutional codes, regarding it as a parallel sequence. We then investigate the features of these two methods.

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[2] Kyoto Institute of Technology, Matsugasaki, Kyoto 606, Japan

# Variable–length source coding and error probability

Hajime Sato[1] and Kingo Kobayashi[2]

We will discuss about an advantage of variable–length coding for any i.i.d. source over fixed–length coding. The error probability of a code (f,g) at distortion level $\Delta$ is defined as $\Pr\{d(x,g(f(\mathbf{x}))) > \Delta\}$ for a distortion measure $d$. It can be proved that the infimum of achievable rates $R^V(\Delta,\epsilon)$ with error probability under $\epsilon$ $(0 \le \epsilon \le 1)$, is proportional to $1-\epsilon$, that is,

$$R^V(\Delta,\epsilon) = (1-\epsilon)R(\Delta),$$

where $R(\Delta)$ is what is called a rate–distortion function. This result shows a remarkable difference from fixed–length coding and the well–known rate–distortion theorem that the infimum of achievable rates $R^F(\Delta,\epsilon)$ is equal to $R(\Delta)$ and does not depend on error probability $\epsilon$ $(0 < \epsilon < 1)$.

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[2] Osaka University, Osaka, Japan

# Very High Speed Soft–Decoding for Convolutional Codes.

Robert Schweikert[1] and Han Vinck[2]

We consider a concatenated **coding** scheme that consists of:

- n parallel convolutional codes, with free distance $d_{free}$, as inner code;
- a single parity check (SPC) outer code.

The **decoder** consists of:

- n parallel operating Viterbi Decoders(VD) followed by;
- soft–decision SPC decoding in case an error event is detected.

The structure of the soft–decision SPC decoding is such that we achieve the following characteristics:

- decoding speed is n–times the individual VD speed;
- the decoding error rate approaches the ML performance for a convolutional code with free distance $2d_{free}$ of the original inner code;
- decoding burst error statistics according to the statistics of the inner code.

If for example, we use a K=7, $d_{free}$=10 convolutional code as inner code, our decoder approaches the performance of an ML decoder for a $d_{free}$=20 convolutional code at the speed of n–times the VD speed for the K=7 code. The decoding complexity is determined mainly by the n constraint length K=7 Viterbi Decoders. The convolutional code that has a $d_{free}$=20 has constraint length K=16, and is thus impossible to implement.

[1] DLR, German Space Agency, Oberpfaffenhofen, West Germany.

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# Elimination of co–channel interferences in an asynchronous spread–spectrum multiple–access systems using real–valued modulatable orthogonal sequences

Naoki Suehiro[1]

This paper proposes a method for elimination of co–channel interferences in an asynchronous Spread–Spectrum Multiple–Access (SSMA) system using the real–valued modulatable orthogonal sequences. This method eliminates co–channel interferences by using a nice property of the sequences. When the number of the co–channel interferences is small, the system eliminates them completely, even if the levels of them are high, by losing small amount of the signal energy. The elimination process is used before the synchronization of the system. So, the method is a good solution for the near–far problem. The proposed method is similar to a method of the same purpose in the asynchronous SSMA system using polyphase modulatable orthogonal sequences. Moreover, the system can estimate which transmitters the co–channel interferences come from. The method will be a good solution for the near–far problem.

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# Modulation for optical recording channels

Cees M.J. van Uijen[1]

The density performance of a large number of channel codes for optical storage is evaluated using a recorder that is interfaced to a software encoder and decoder. The experiments are performed both on mastered discs, and on write–once and erasable media. The study includes a new class of codes devised for an asymmetrical recording channel.

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# Coding for binary multi–user channels

Peter Vanroose[1]

All possible binary deterministic multi–user channels with two information sources are investigated. For several of those channels, coding techniques will be presented and reviewed. These include, more particularly, a class of codes reaching the boundary of the zero–error capacity region of the binary switching multiple–access channel and a class of codes reaching the boundary of the zero–error capacity region of the Blackwell broadcast channel.

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# Channel capacity evaluation of multiple–access channel with binary output.

Yoichiro Watanabe[1]

A channel capacity evaluation is studied for a (discrete memoryless) multiple–access channel (MAC). This includes a sufficient condition of determining the channel capacity and an iteration procedure of calculating the channel capacity. The channel capacity is indispensable for calculation of the capacity region [1] of an MAC. The Kuhn–Tucker conditions concerning an MAC are necessary conditions for the channel capacity, i.e. give a saddle point of the mutual information, since the mutual information takes the value on the non–convex domain. Fortunately, it can be proved that for an MAC with binary output, a local maximum point of the mutual information takes the channel capacity.

An iteration procedure for calculating a saddle point of the mutual information is proposed for a general MAC, which is totally different from the Arimoto–Blahut algorithm [2]/[3] for the ordinary discrete memoryless channel. When this is combined with the sufficient condition, the channel capacity of the MAC with binary output can be obtained with a desired accuracy.

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[3] R.E. Blahut, "Computation of Channel Capacity and Rate–distorsion Functions," IEEE Trans. Inform. Theory, vol. IT–18, pp. 460–473, july 1972.

[1] Stanford University, on leave of the Department of Electronics, Faculty of Engineering, Doshisha University, Kyoto, 602, Japan.

# An adaptive channel error protection scheme for subband encoded images

P.H. Westerink, J.H. Weber, J.W. Limpers and D.E. Boekee[1]

This paper deals with the channel error protection of images that are encoded using subband coding. In this scheme the low–pass subband is encoded using DPCM and the other subbands are encoded using a scalar quantizer. The quantizers are all Lloyd–Max quantizers, where the representation levels have fixed length codewords. First, by considering only single errors in each codeword, a channel error distortion measure is derived for each quantizer, that is, for each subband. Codewords will be assigned to the quantizer representation levels, yielding a low value of the distortion measure. Next, sets $S_{ij}$ will be formed consisting of the 1–th bit from each codeword of subband j. Each set $S_{ij}$ will be assigned a particular BCH code $C_{ij}$ to each set $S_{ij}$, based on a channel error distortion measure for the entire image. The protection scheme can be seen to be adaptive, because each set of bits within each subband can be assigned a different error protection code. Examples will show the favourable approach over, for instance, assigning equal error protection codes to each set of bits. It is shown that in the case of a channel error probability of $10^{-3}$ in this manner only a 5 to 10% extra number of bits are needed for an adequate channel error protection of an image.

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# Video source coding for digital video recording

Peter de With[1]

Digital video recording is expected to become particularly important in those applications where multiple copying is often required, such as electronic imaging. Moreover, the growing interest in HDTV signals might considerably influence a possible application of digital recording for home use.

For successful use of digital video recording in these applications, it is of paramount importance that the bit rate to be recorded is as low as possible, i.e. only a fraction of the say 200 Mbit/s (TV) which would be required for straight PCM recording. It is the only way to obtain sufficient playing time with a small cassette.

VLSI technology offers the opportunity for cost–effective implementation of advanced bit–rate reduction schemes with high reduction factors. Among such schemes intraframe DCT threshold coding has proven to be an effective algorithm for video data compression. This type of coding has been modified to meet the constraints for a recording system. Fast DCT algorithms and bit–assignment techniques will also be discussed.

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