frequency of the next hop channel. Since the algorithm keeps track of the number of data bits received, knowledge of the end of the time slots is available to the algorithm. The algorithm maintains synchronisation by switching frequencies according to the hopping pattern at the end of each time slot. The assumptions here are that the synchronisation preamble has not been detected due to a temporary fade or interference on that particular hop channel and that channel conditions will be different for the next hop channel.

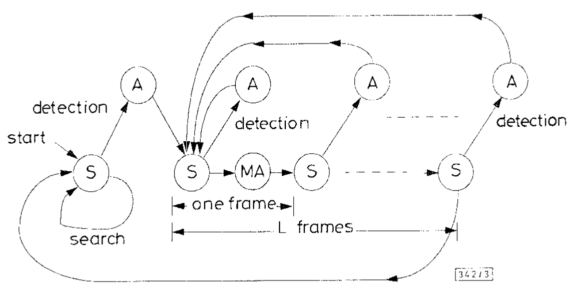

Fig. 3 *State diagram of proposed algorithm*

S: search state
A: acquisition state
MA: maintain acquisition state
L: number of frames not detected before resynchronising afresh

The bit timing required to sample the demodulated bit stream is based on the timing recovered in the last detected frame. This is done because in a faded or interference channel, the bit timing recovered would most probably be incorrect. Timing recovery is resumed once a new synchronisation preamble has been detected. If, after a certain number of hops, the receiver has still not detected the synchronisation sequence, the algorithm decides that synchronisation has been lost completely and re-starts the synchronisation procedure. Fig. 2 shows the flowchart for the proposed algorithm. Fig. 3 shows the simplified state diagram of the proposed technique. As seen from the Figure, the proposed algorithm has an extra state called the maintain acquisition (MA) state where the algorithm tries to maintain synchronisation by extrapolation based on past stored information.

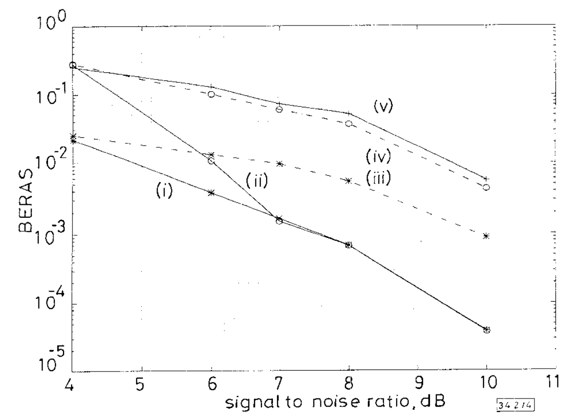

Fig. 4 *BERAS performance*

(i) proposed algorithm in AWGN channel
(ii) CTR in AWGN channel
(iii) proposed algorithm in frequency selective channel
(iv) CTR in frequency selective channel
(v) performance of non-hopped system in frequency selective channel

Simulation results and conclusion: Simulations were carried out to evaluate the performance of the proposed technique. The system simulated was a time division duplexed packet SFH system with a total frame duration of 10ms. GMSK modulation is used with $BT = 0.5$. A limiter discriminator is used to demodulate the frequency hopped GMSK signal. A 31 bit pseudo-noise sequence is used as the synchronisation preamble. The performance of this SFH system with synchronisation implemented by both the conventional and the proposed technique was evaluated for an AWGN and a static frequency selective channel modelled as a two path profile with a power difference of 8dB between the first and the delayed path. The two paths are separated by a delay of one-third the symbol period. The channel bit rate is 156.25kbit/s with the user data rate being 52.083kbit/s. The number of consecutive frames not detected (L) before starting synchronisation afresh is 5. The performance measure used to evaluate system performance is the bit error rate in all slots (BERAS) [4] which links the frame error rate and the bit error rate in good slots [4]. Fig. 4 shows the BERAS performance of the system with the different synchronisation techniques. There is a clear disparity in performance for both the AWGN and the frequency selective channel especially at low signal to noise ratios. Fig. 4(i) shows the performance of a non-hopped system. It is obvious that the potential gains that can be realised by an FH system in a frequency selective channel would be lost if a robust synchronisation technique were not used.

It has been shown in this Letter how the conventional transmitted reference technique can be modified to make it more robust to fading and other channel distortion. Simulation results showing the potential performance gains that can be achieved for a typical application have been presented. This technique can also be easily customised for different FH architectures.

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Robust vector quantisation by transmission energy allocation

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Indexing terms: Vector quantisation, Codes

The authors propose to combat the effects of channel errors on vector-quantised data by optimising the energy allocation for the transmission of codeword symbols. In particular, they demonstrate that the application of transmission energy allocation to the natural binary code obtained from vector quantiser design by the splitting method outperforms established robust techniques such as index assignment, and explicit error control coding of most sensitive bits.

Introduction: Vector quantisation has become a popular and effective technique for signal compression and is now used as the central building block in an increasing number of speech and image coding algorithms [5]. With the advent of wireless communication systems, there has been a growing realisation of the vital need to effectively protect the compressed signal against channel errors. The bandwidth constraints of these systems allow only minimal protection by error correcting codes. Hence, it is necessary to design the vector quantiser (VQ), with significant built-in robustness to channel errors. This problem has received much attention recently [3, 4]. However, most of the reported techniques optimise the VQ while assuming a fixed modulation scheme and thereby limit the achievable gains. In this Letter we show that by designing the VQ appropriately, followed by optimising the transmission energy allocation among the codeword bits, substantial gains in the overall SNR can be obtained. The approach we develop has its roots in the 1958 paper by Bedrosian [1], and the later work of Sundberg (see recent description in [2]), where the transmission energy allocation idea was applied to pulse code modulation. We formulate the problem in a general framework that is applicable to a large class of coding and quantisation methods, and suggest the energy allocation algorithm (EAA) to optimise the modulation scheme. We demonstrate the power of the approach by applying it to a VQ encoder employing BPSK modulation.

Index assignment and natural binary code: Consider a VQ with codebook $C = \{y_0, y_1, ..., y_{2^n-1}\}$, and a source denoted by random vector x. The index I is transmitted over the noisy channel and index J is received. Assuming the squared error distortion measure, and that the codevectors satisfy the centroid rule [6], the overall distortion is given by

$$D = \overbrace{E\|x - y_I\|^2}^{D_q} + \overbrace{E\|y_J - y_I\|^2}^{D_c} \qquad (1)$$

where D_q and D_c denote the distortion due to quantisation and channel errors, respectively. Index assignment [4] aims to minimise D_c by a judicious assignment of binary indices to the codevectors. A simple approach to index assignment is to use the natural binary code obtained from VQ design initialised by the splitting method (see description in [3]). An important observation is that the resulting VQ codeword bits are not equally sensitive to channel errors. To consider the sensitivity of specific bits we first introduce explicit notation for the transmitted index: $I = (i_1, i_2, ..., i_n)$. The sensitivity of the jth bit is defined as the expected amount of distortion caused by a bit error at this location:

$$D_j = E\|y_{i_1,i_2,...,i_j,...,i_n} - y_{i_1,i_2,...,i_j^*,...,i_n}\|^2 \qquad (2)$$

where, $i_j^* = 1-i_j$ is the complement of i_j. Table 1 shows the variation in sensitivity of the index bits of VQs designed for Gauss Markov processes using the splitting initialisation. Note the large variation in the bit sensitivities which suggests that an unequal error protection scheme would be appropriate.

Table 1: Sensitivities of bits, D_j, in natural binary code for two-dimensional VQ of size 256

Bit number j	0	1	2	3	4	5	6	7
ρ = 0	7.29	4.10	2.00	1.78	0.69	0.42	0.19	0.07
ρ = 0.9	7.28	3.77	1.28	0.65	0.31	0.18	0.08	0.03

The VQ was designed for a Gauss-Markov source with correlation coefficient ρ.

Energy allocation algorithm: We assume a transmission scheme where each of the n bits in the codeword is transmitted independently. In other words, any form of coded modulation is carried out by grouping bits of the same sensitivity from consecutive codewords. We are given a quota of transmission energy E_{tot} to be distributed to n bits. Let E_i denote the energy allocated to the ith bit. The probability of channel error for the ith bit can be written as $\varepsilon_i = f(E_i, \sigma_n^2)$, where σ_n^2 is the variance of the Gaussian channel noise, and the function $f(\cdot)$ depends on the type of coded modulation used for transmission. Neglecting the possibility of more than single bit errors, we formulate the distortion due to channel errors as

$$D_c = \sum_{i=1}^{n} D_i \varepsilon_i \qquad (3)$$

We wish to minimise D_c over all choices of $\{E_i\}$, that is by allocating transmission energy to the bits, subject to the total energy constraint

$$\sum_{i=1}^{n} E_i = E_{tot}$$

From the corresponding Lagrangian we obtain the necessary condition for optimality:

$$\frac{\partial D_c}{\partial E_i} = D_i \frac{\partial \varepsilon_i}{\partial E_i} = -\lambda \qquad (4)$$

where λ is the Lagrange multiplier whose value is chosen to satisfy the total energy constraint. Although various descent algorithms can be easily derived, we chose to describe here an algorithm that is analogous to known bit allocation algorithms [5].

Energy allocation algorithm (EAA): Choose a small quantum of energy ΔE, such that $E_{tot} = M\Delta E$, for some integer M. The energy allocated to each bit will be constrained to be an integer multiple of ΔE. We assume that the quantity $[\partial f(E_i,\sigma_n^2)]/\partial E$ can be evaluated or is known for all values of E and σ_n^2. We begin by assigning zero energy to all the n bits. Now we compare the values of $\partial D_c/\partial E_i$ for all i and allocate energy ΔE to the bit with the largest value of $\partial D_c/\partial E_i$. Having allocated the energy we update the value of $\partial\varepsilon_i/\partial E_i$ for that bit. We repeat the procedure of allocating a quantum of energy to the bit with the highest partial derivative and stop when all the available energy has been allocated.

Results for BPSK: In this Section, we apply the proposed EAA to design a robust communication system with a VQ and a simple BPSK modulation scheme. For BPSK modulation, the partial derivative $\partial\varepsilon_i/\partial E_i$ can be evaluated as

$$\frac{\partial \varepsilon_i}{\partial E_i} = -\frac{1}{2\sigma_n\sqrt{2\pi E_i}} e^{-\frac{E_i}{2\sigma_n^2}}$$

Table 2: Overall SNR for two-dimensional vector-quantised Gauss Markov source with correlation coefficient ρ

Channel SNR	EAA		Pseudo-Gray		CC-MSB	
	ρ = 0.9	ρ = 0	ρ = 0.9	ρ = 0	ρ = 0.9	ρ = 0
dB						
5	10.74	7.85	6.06	5.04	10.02	7.71
6	12.92	9.97	8.10	7.06	11.97	9.62
7	15.51	12.43	10.74	9.50	14.31	11.79
8	18.15	15.20	13.78	12.51	16.90	14.07
9	21.16	18.00	17.35	15.77	19.30	16.23
10	23.47	20.21	21.16	18.96	21.10	17.71

The energy allocation algorithm FAA is compared with two standard alternatives: Pseudo-Gray for index assignment, and convolutional code protection of the MSB (CC-MSB). The codebook size is 256 for EAA and the pseudo-Gray techniques. CC-MSB uses a codebook size of 128 and employs a rate 1/2 convolutional code to protect the MSB.

The results for Gauss Markov sources are given in Table 2. We compare the performance of the proposed EAA with that of two popular schemes which allocate equal energy to all the bits. The first is the pseudo-Gray coding method for index assignment [4]. The second, CC-MSB, uses a VQ designed by the splitting initialisation, followed by a rate 1/2 convolutional code to protect the MSB, while the remaining bits are unprotected. Here, index assignment is performed on the unprotected bits to reduce the effects of channel errors [3]. We observe that EAA significantly outperforms the other two schemes. At high levels of channel noise, CC-MSB comes close to the performance of EAA but its relative performance degrades significantly (by more than 2dB) for less noisy channels. Compared to pseudo-Gray coding, EAA achieves ~3–4dB gains in overall SNR for moderately noisy channels. Thus the proposed EAA method shows much promise as a method for robust transmission of VQ-compressed signals over noisy channels.

Extensions and future work: It should be noted that although we have presented the energy allocation algorithm for the case of full search VQ, it is also applicable to a large class of structured VQs such as multistage VQ and tree-structured VQ. Extensions of the method, and results on its application to structured VQ schemes combined with more powerful coded modulation techniques, are under preparation for future publication.

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Viterbi algorithm with data-aided phase reference estimation for convolutionally encoded PSK

Li Bin and P. Ho

Indexing terms: Phase shift keying, Viterbi decoding, Convolutional codes

A Viterbi algorithm with data-aided phase reference estimation for convolutionally encoded PSK is presented. The simulated error performance results are given. Very attractive results are obtained, which show that the new Viterbi algorithm with data-aided phase reference estimation can provide almost the same performance as that of the Viterbi algorithm with perfect phase reference.

Introduction: PSK modulation will be of use in future mobile communications applications because of its bandwidth efficiency. Convolutional encoding of BPSK (or QPSK) and Viterbi decoding are used to reduce the required transmission power and/or to improve the communication performance over noisy links, e.g. in satellite communication systems where the transmitter power is limited. In many applications where carrier phase and/or frequency are likely to be uncertain, e.g. a multipath fading environment, differential detection is often more power efficient and robust than coherent detection. The prime reason for this is that it is very difficult for us to acquire and track a coherent demodulation reference signal (i.e. coherent carrier-phase reference signal) by a conventional phase-locked loop. However, the Viterbi algorithm with differential detection suffers from a performance penalty (additional required signal-to-noise ratio at a given bit error rate) when compared to the Viterbi algorithm with ideal (perfect carrier phase reference) coherent detection. To solve this problem, we have developed a new Viterbi algorithm with so called data-aided phase reference estimation. It is observed from the simulation results that the error performance of the new Viterbi algorithm is almost the same as that of the Viterbi algorithm with ideal coherent detection.

New Viterbi algorithm: For an additive white Gaussian noise channel, MPSK in the interval $kT \leq t \leq (k+1)T$ has the complex form

$$r_k = \sqrt{\frac{2E_s}{T}} e^{j\phi_k + j\theta_k} + n_k \qquad (1)$$

where E_s denotes the constant signal energy, T denotes the MPSK symbol interval, and ϕ_k denotes the transmitted phase which takes on one of M uniformly distributed values $\beta_m = 2\pi m/M$; $m = 0, 1, ..., M-1$ around the unit circle, and θ_k is an unknown arbitrary phase introduced by the channel which is assumed to be constant or be slowly-varying relatively to the data rate. n_k is a zero-mean complex Gaussian noise sample with variance $\sigma_n^2 = (2N_0/T)$ (N_0 is the one-sided power spectrum density).

In a differential encoded/detected PSK system, we are essentially using the received sample r_{k-1} as an estimate of $\exp(j\theta_k)$. This simple technique works quite well. A more sophisticated estimator/detector is the so called data-aided detector structure considered in [1]. In particular, the estimate of $\exp(j\theta_k)$ is given by

$$\sum_{i=1}^{\infty} \alpha^{i-1} r_{k-i} e^{-j\hat{\phi}_{k-i}}$$

where $\hat{\phi}_{k-i}$ is the phase of a previously detected channel symbol and α ($0 \leq a \leq 1$) is the rate of decay of a digital filter with an exponential impulse response.

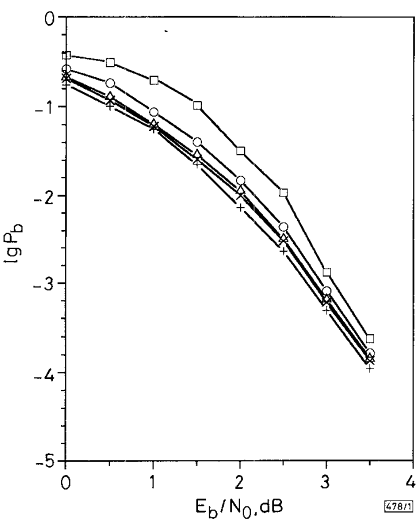

Fig. 1 *Simulated bit error probability of Viterbi decoding with phase reference estimation*

□ $k_\alpha = 10$, ○ $k_\alpha = 20$, △ $k_\alpha = 40$, × $k_\alpha = 60$, + perfect phase reference

Based on the above data-aided phase detection technique, a new Viterbi algorithm is developed for convolutionally coded PSK signals as follows. The so called path phases, like the path metrics, are used at every state. Each trellis state has its own phase reference, and the phase reference at each trellis state is obtained recursively from the phase reference at the previous trellis state in the corresponding survivor path. When a survivor path (e.g. from state S_{n-1} to state S_n) is selected, the phase reference $\exp(j\theta_n)$ at state S_n is estimated by $\alpha\exp(j\theta_{n-1}) + r_n \times \exp(-j\phi_n)$, where $\exp(j\theta_{n-1})$ is the phase reference at state S_{n-1}, r_n is a row vector of the received signals in the time interval from state S_{n-1} to S_n and $\exp(-j\phi_n)$ is a column vector of the exponential phases corresponding to the transmitted information in the branch from state S_{n-1} to S_n. After Viterbi detection, a differential decoder is used to overcome the phase ambiguities. Although the proposed method of joint phase estimation and data determination increases the complexity of the receiver, it provides good phase tracking ability when the phase θ_k varies slowly. Fig. 1 shows the simulated error performance of convolutionally encoded BPSK (with the optimal rate 1/2, constraint length 7 code) and the new Viterbi algorithm, where $k_\alpha = (1+\alpha)/(1-\alpha)$. It is observed that the error performance of the proposed Viterbi algorithm with phase estimation is very close to that of a perfect phase reference.

Simplified phase reference estimation: The proposed Viterbi detector for joint data detection and phase estimation can be simplified by using only one phase reference obtained from the detected symbols at the output of the Viterbi decoder, instead of using a separate phase estimate at every state. Fig. 2 shows the structure of this simplified phase estimator. Although simpler to implement, the detector/estimator in Fig. 2 may provide unsatisfactory performance if the channel phase has a period of coherence that is significantly shorter than the detecting window width of the Viterbi receiver. However for a pure additive white Gaussian noise channel, the simplified detector/estimator will provide good performance with modest implementation complexity. Fig. 3 shows