to 100 ATM cells and the MCR for each source was set to 30% of its mean bit rate. The simulations were carried out separately for the case of 'ERICA only', 'ERICA+ only' and 'ERICA with buffer and queue control algorithm'. Their relative performance was compared in terms of: (i) the end-to-end cell queueing delay, which is measured from the input of the transcoder to the multiplexing switch output; and (ii) the picture quality of the received ABR video. Table 1 shows the results of the end-to-end cell delay for one of the sources. All other sources had a similar behaviour.

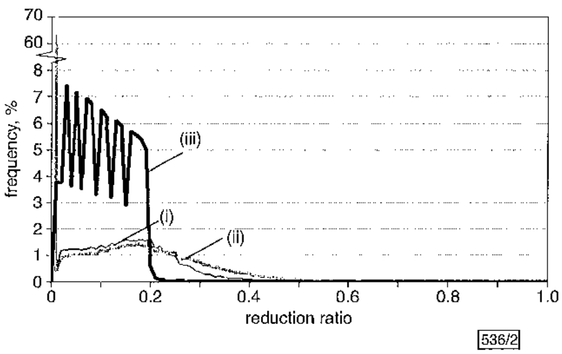

Fig. 2 *Transcoder reduction ratio of ABR video*

(i) ERICA
(ii) ERICA+
(iii) ERICA with smoothing buffer and queue control algorithm

Table 1: End-to-end cell queueing delay simulation results

	ERICA only	ERICA+ only	ERICA with buffer and queue control algorithm
Mean end-to-end cell delay [ms]	0.98	0.47	0.42
Standard deviation of the end-to-end cell delay [ms]	1.14	0.38	0.47

As demonstrated in Table 1, 'ERICA with buffer and queue control algorithm' has a similar end-to-end cell delay to 'ERICA+ only' which is lower than that of 'ERICA only'. The required smoothing buffer size to prevent cell loss is very small, 14 ATM cells (less than 1Kbyte) for this experiment. Fig. 2 illustrates the bit rate reduction ratio distribution (defined as $1 - R_{new}$) for one of the sources, which has a direct impact on the picture quality. The higher the reduction ratio, the lower is the output bit rate of the transcoded video and hence the more severe is the picture quality degradation. As the results show, adding a small smoothing buffer to ERICA with a simple control, the frequency of high reduction ratio (reduction ratio > 0.2) compared to the other two cases has been drastically reduced. This implies that the video bit stream was compressed within a narrower range of 0 and 0.2 throughout the whole sequence, which leads to less noticeable picture distortion. The ERICA+ scheme has the highest frequency of high reduction ratio and therefore results in the poorest quality.

Conclusion: With the ERICA congestion control algorithm, we have demonstrated that, by adding a small smoothing buffer and a linear queue control algorithm at the video transcoder output, the performance of ABR video services can be improved without causing much queueing delay. Compared to the use of the ERICA and ERICA+ algoithms only, the picture quality is much better while the end-to-end cell queueing delay is comparable to that of the ERICA+ scheme.

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Reducing the peak-to-average power ratio of orthogonal frequency division multiplexing signal through bit or symbol interleaving

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An interleaver-based technique for improving the peak-to-average power ratio (PAP) of an orthogonal frequency division multiplexing signal is presented. For this technique, $K - 1$ random interleavers are used to produce $(K - 1)$ permuted sequences from the same information sequence. The PAPs of the permuted sequences and the original information sequence are then computed using K oversampled fast Fourier transforms (OFFTs). The sequence with the lowest PAP is chosen for transmission. Results show that for 256 subcarriers and quadrature phase shift keying data symbols, even with $K = 2$, the 0.1% PAP is reduced by 1.3dB and with $K = 4$, it is reduced by 2dB. The 0.1% PAP can be reduced by 3dB and the 0.01% PAP by 4dB at a cost of 16 OFFTs and a data rate loss of < 0.8% with $K = 16$.

Introduction: Orthogonal frequency division multiplexing (OFDM) is one of the most promising solutions for obtaining high data rate transmission in frequency selective fading radio channels [1]. Advances in VLSI technologies have led to new interest in this technique for transmitting information through overlapping carriers. Current OFDM applications include digital audio broadcasting, outside broadcast links, asynchronous digital subscriber lines, digital video broadcasting and wireless local area networks (LANs). OFDM is commonly implemented using discrete Fourier transform (DFT) techniques. A main disadvantage of OFDM is potentially high peak-to-average power ratio (PAP) values. In practice, peak signal levels are constrained by design factors such as battery power (on portable equipment) or regulatory limits that prevent adjacent channel interference. The use of nonlinear amplifiers and digital hard limiting causes inefficiency, interference and performance degradation.

These drawbacks have prompted searches for PAP reduction techniques and many solutions have been proposed. Both the partial transmit sequence (PTS) and the selective mapping (SLM) approaches proposed by Muller and Huber [2] are based on the phase shifting of sub-blocks of data symbols and the multiplication of the data frame by random vectors. In the PTS scheme, the input data frame of N symbols is partitioned into M sub-blocks. The N/M subcarriers in each sub-block (except the first sub-block) are phase shifted by a constant factor; $M - 1$ phase factors are then selected so that the resultant PAP is minimised. Unfortunately, finding the best phase factors is a highly complex and nonlinear optimisation problem. Therefore attempts have been made to reduce the complexity of the optimisation: a suboptimal iterative flipping algorithm was presented in [3], while in [4] an alternative optimisation criterion was presented. Yet even with full optimisation, the maximum 0.1% PAP reduction is ~4.2dB for a 256 QPSK-carrier system. The complexity of the PTS scheme is demonstrated by the fact that, if the $M - 1$ phase factors are limited to 0 and π, 2^{M-1} PAP computations are still required in order to find the optimum phase factors. In fact, to achieve the 4.2dB reduction mentioned above, 32768 linear combinations of the sub-block IDFTs need to be tested. This is assuming an exhaustive search; a partial search could be performed (e.g. in [3] the full search requires 2000 steps).

Using the SLM approach, the input data frame is multiplied by M random sequences and the resultant sequence with the lowest PAP is then selected for transmission. To enable the receiver to recover the data, a pointer to the multiplying sequence can be transmitted as side information. As a result, the computational cost includes both MN multiplications and M OFFTs.

In this Letter we propose a data randomisation technique for improving the PAP of an OFDM system. This technique is based on the heuristic notion that highly correlated data frames have large PAPs, which could thus be reduced, if the long correlation patterns were broken down. We therefore employ $K-1$ interleavers, each of which produces a re-ordering or permutation of the N symbols of the input data frame. The frame with the lowest PAP of all the K frames (the original plus $K-1$ permutations) is selected for transmission. To recover the data, the receiver need only know which interleaved sequence has been transmitted; a pointer (integer between 0 and $K-1$) to the selected interleaver is a minimal additional overhead (i.e. data rate loss) during transmission. For an N subcarrier QPSK system, the overhead created by additional information is $\log_2(K)/2N$, which is negligible for large N. Note that either symbol interleaving or bit interleaving can be used and the latter offers more gain.

It is noteworthy that the PAP of the continuous-time OFDM signal cannot be computed precisely by the use of the Nyquist sampling rate, which amounts to N samples per symbol. In this case, signal peaks are missed and PAP reduction estimates are unduly optimistic. Oversampling by a factor of 4 is sufficiently accurate and is achieved by simply computing a $4N$-point zero-padded IDFT of the data frame. Therefore, oversampling by a factor of 4 is assumed throughout this Letter.

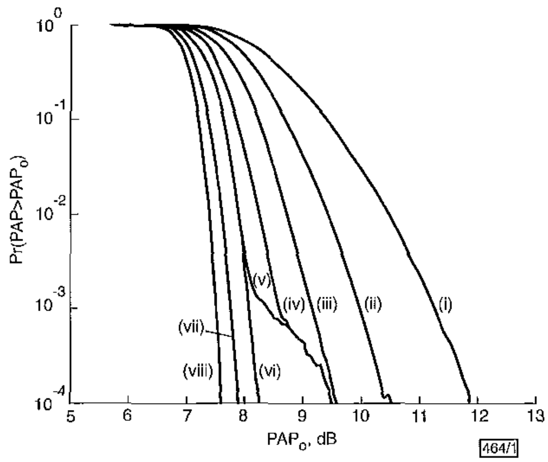

Fig. 1 *CCDF of OFDM signal with random K – 1 interleavers*

(i) $K = 1$ (uncoded)
(ii) $K = 2$ (symbol interleaving)
(iii) $K = 4$ (symbol interleaving)
(iv) $K = 8$ (symbol interleaving)
(v) $K = 16$ (symbol interleaving)
(vi) $K = 16$ (bit interleaving)
(vii) $K = 32$ (bit interleaving)
(viii) $K = 64$ (bit interleaving)

PAP reduction:

(i) *Peak-to-average power ratio*: In OFDM, a block of N symbols $\{X_n, n = 0, 1, ..., N-1\}$ is formed with each symbol modulating one of a set of N subcarriers, $\{f_n, n = 0, 1, ..., N-1\}$. The N subcarriers are chosen to be orthogonal, i.e. $f_n = n\Delta f$, where $\Delta f = 1/NT$ and T is the original symbol period. The resulting signal can be expressed as

$$x(t) = \sum_{n=0}^{N-1} X_n e^{j2\pi f_n t} \qquad 0 \le t \le NT \qquad (1)$$

The PAP of the transmitted signal in eqn. 1 can be defined as

$$PAP = \frac{\max |x(t)|^2}{E[|x(t)|^2]} \qquad (2)$$

(ii) *Partial transmit sequences*: For comparative purposes, a brief description of the PTS method is provided here. We define the data frame as a vector $\mathbf{X} = [X_0, X_1, ..., X_{N-1}]^T$ and partition it into M disjoint sub-blocks, represented by the vectors $\{\mathbf{X}_m, m = 1, 2, ..., M\}$, such that $\mathbf{X} = [\mathbf{X}_1, \mathbf{X}_2, ..., \mathbf{X}_M]^T$. It is assumed that the sub-blocks consist of a contiguous set of subcarriers and are of equal size. The IDFT of $\mathbf{X}_m$s are called partial transmit sequences. Let $y_{n,m}$ for $n = 0, 1, 2, ..., 4N-1$, $m = 1, 2, ..., M$, be the $4N$ point IDFTs of $\mathbf{X}_m$, appropriately zero-padded. Using the linearity property of IDFTs, the time domain samples can be represented as

$$S_n = \sum_{m=1}^{M} b_m y_{n,m} \qquad n = 0, 1, ..., 4N \qquad (3)$$

where $\{b_m, m = 1, 2, ..., M\}$ are weighting factors. They are further assumed to be pure rotations (i.e. $b_m = e^{j\phi_m}$). The weighting factors are chosen to minimise the peak $|S_n|$. To reduce the complexity of this minimisation, we use a suboptimal choice by limiting all b_m to +1 or –1. Without any loss of performance we can set $b_1 = 1$ and observe that there are $(M-1)$ binary variables to be optimised. Finally the optimal PAP can be found as

$$PAP_{optimal} = \min_{b_1,...,b_M} \left(\max_{0 \le n \le 4N} \left| \sum_{m=1}^{M} b_m y_{n,m} \right|^2 \right) \qquad (4)$$

In the PTS scheme, the PAP in eqn. 4 is computed for all the binary combinations and hence requires 2^{M-1} iterations. Note that the above development is similar to that in [2] except for the use of oversampling.

Random interleaver approach:

(i) *Random interleaver realisation*: A random interleaver is a block interleaver that operates on a block of (N) symbols and re-orders or permutes them in a pseudorandom order; i.e. symbol sequence $(X_0, X_1, ..., X_n, ..., X_{N-1})$ becomes $(X_{\pi(0)}, X_{\pi(1)}, ..., X_{\pi(n)}, ..., X_{\pi(N-1)})$ where $\{n\} \leftrightarrow \{\pi(n)\}$ is a one to one mapping and $\pi(n) \in \{0, 1, ..., N-1\}$ for all n. Both the transmitter and the receiver store the permutation indices $\{\pi(n)\}$ in memory. Thus interleaving and de-interleaving are very simple exercises.

Pseudorandom permutations can be obtained from linear congruent sequences. However, we use a specific algorithm to generate each permutation (interleaver). At the nth step, we generate a large random number R repeatedly (if necessary), until R mod N is not equal to any one of $\pi(0), \pi(1), ..., \pi(n)$, and $\pi(n)$ is set equal to R mod N. This step is repeated for $n = 0, 1, ..., N-1$. The whole procedure is repeated $K-1$ times to generate the required interleavers.

(ii) *RI system model*: N data symbols constitute a frame and $K-1$ symbol interleavers produce $K-1$ permuted frames. The $4 \times$ OFFT of each frame is used to compute its PAP. The minimum PAP frame of the K frames is selected for transmission. The identity of the corresponding interleaver is also sent to the receiver as side information.

Results: A 256 carrier OFDM system with QPSK modulation was simulated and 10^5 OFDM symbols were generated. For the use of symbol interleaving, the 256 QPSK symbols of a frame were permuted by $K-1$ RIs. For the use of bit interleaving, the 512 bits of a frame were permuted by $K-1$ RIs. Fig. 1 shows the complementary cumulative distribution function (CCDF), $\Pr(PAP > PAP_o)$. The $K = 1$ curve represents the original OFDM signal. The PAP statistics improve with increasing K. For $K = 4$, the 0.1% PAP is reduced by ~2.2dB. For $K = 16$, the reduction is about 3 dB. This appears to be the maximum PAP reduction achievable with symbol interleaving. For a given K, both bit interleaving and symbol interleaving offer the same PAP reduction in the region $CCDF > 10^{-3}$. It should also be noted that the Walsh sequence method [3] reduces 0.1% PAP by 2.8dB at a cost of 16 PAP computations. The RI method also achieves a very similar result at a cost of eight OFFTs (see curve (iv)).

Conclusion: In this Letter we have presented a data randomisation technique for PAP reduction. For this RI approach, K permutations are generated from the input data frame and the permuted frame with the lowest PAP is selected for transmission. The method is of similar complexity to the PTS method, and achieves comparable results. Notably, the RI approach can also be used in conjunction with other methods.

While it is well known that interleaving can be used to combat the effects of noise bursts and fading in error correction systems, the use of interleaving as in this Letter serves a different purpose, one which has not been recognised before. By interleaving a data frame, the peaks in the associated OFDM signal can be compressed. This transformation is most effective for data frames with moderate PAP values (very high PAP values which are nearly N cannot be reduced by this method). The maximum PAP reduction of this method occurs at ~0.1% PAP level. This is most useful because the PAP of practical systems is usually < 0.1% PAP. Conversely, the worst case PAP value of N for OFDM signals is extremely unlikely for random data. Since the RI approach does not attempt to reduce the worst case PAP, the data rate loss of this method is negligible. The RI approach lends itself to the use of error correction codes and this topic is undergoing further investigation.

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Computational improvement for G.729 standard

Shaw-Hwa Hwang

G.729 is a CS-ACELP-based [1] standard for 8kbit/s speech compression and is defined by the ITU. In the G.729 encoding phase, the open-loop pitch T_{op} is estimated for each frame to reduce the computation required to search for the best adaptive codebook delay. However, > 25% CPU time is required for the open-loop pitch analysis. The weighted delta-LSP(WD-LSP) function is employed here to reduce the computation required for open-loop pitch analysis. An 81% saving is made in the computation of T_{op}. A total computational saving of 21% for the encoding phase is achieved and the speech quality still sounds good (MOS = 3.5).

Introduction: Speech compression technology is widely used in digital communication systems, such as mobile phone, VoIP, video conference technology, etc. Speech compression reduces the redundant data and provides bandwidth savings for the communication channel. The code-excited linear predictive (CELP) algorithm has been successfully used for speech compression [1]. The quality of speech obtained and the complexity of the CELP vocoder are superior to that of other vocoders [2 – 4]. Thus, many CELP-based speech compression standards such as FS1016 [5], G.723.1 [6] and G.729 [7] have been proposed. For practical applications, low cost and high efficiency are very important. There is a particular demand for real-time speech compression in PC or standalone systems. Thus, improving the computation involved in speech compression is very important.

Recently, with the development of internet applications, VoIP (voice over internet protocol) technology has been widely used to meet the requirements of personal communication systems. Thus, the G.729 standard, which is a CS-ACELP-based speech compression standard has been widely used as a VoIP product. However, the computational requirement of the G.729 standard is still extremely large. In the past, a great deal of effort has been invested in the search for a fixed codebook [8]. In Table 1, a total of 21% of the computation time is devoted to the search for the fixed codebook. However, more than 25.84 and 29.32% computation are spent on the open-loop pitch search and adaptive codebook search. Thus, in this Letter, a WD-LSP-based algorithm is employed to reduce the computation required for the open-loop pitch analysis. A computational saving of more than 81% for the open-loop pitch analysis can be achieved. A total computational saving of 21% for the G.729 encoding phase is achieved and the speech quality is still good.

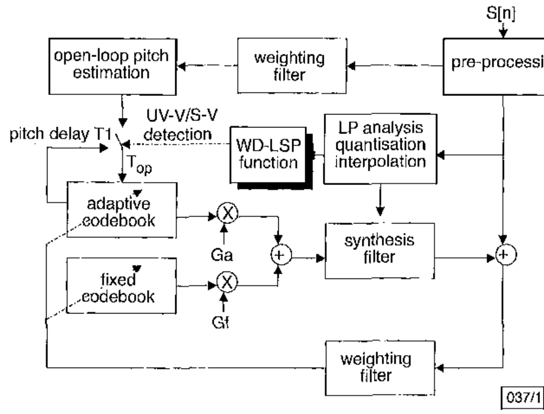

Fig. 1 *Encoding principle of modified G.729*

System description: The G.729 standard is for speech compression at 8 kbit/s. In this standard, the conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP) algorithm is employed for speech coding. Fig. 1 shows the encoding principle of the modified G.729 standard. First, each 10 ms frame data is used to estimate the tenth-order LSP coefficients. The fixed and adaptive codewords are then multiplied with gains, summed, and filtered using a short-term filter to generate synthesis speech for each 5ms subframe. Finally, the weighted error signal and close-loop analysis method are used to estimate the best codeword indices and gains for each subframe. Table 1 lists the encoding step for each frame. The CPU time requirement for each step is also included.

Table 1: Encoding step and CPU time requirement

Step	CPU time requirement %	Function description
1	23.14	Pre-processing, estimate, quantised and interpolated LSP
2	25.84	Open-loop pitch analysis
3	14.16	Adaptive-codeword 1 index and gain estimate
4	10.50	Fixed-codeword 1 index and gain estimate
6	15.16	Adaptive-codeword 2 index and gain estimate
7	10.50	Fixed-codeword 2 index and gain estimate
8	0.7	Bitstream packing and memory update

In Fig. 1, the open-loop pitch analysis is employed to estimate the pitch delay T_{op}. The search range of the first adaptive codeword T_1 is then limited into $[T_{op}-3, T_{op}+3]$ and the second subframe is limited into $[T_{1-5}, T_1+5]$. Thus, the computation required to search for the adaptive codebook is dramatically reduced. However, estimating the open-loop pitch requires more than 25.84% CPU time. The open-loop pitch analysis uses the following correlation function $R(k)$ to detect three maxima in different ranges.

$$R(k) = \sum_{n=0}^{79} sw(n)sw(n-k)$$

The window speech $sw(n)$ is obtained by using the Hamming-Cosine window function and the three maxima $R(t_1)$, $R(t_2)$, $R(t_3)$