References

1 MURTHY, I.S.N., and DUGA PRASAD, G.S.S.: 'Analysis of ECG from pole-zero models', *IEEE Trans. Biomedical Eng.*, 1992, **39**, pp. 741–751

2 PAUL, J.S., KUMAR, V.J., and REDDY, M.R.S.: 'Model order determination of ECG signals using rational function approximations'. Proc. 3rd Int. workshop in Signal/Image Processing, IWSIP '96, Manchester, UK, November 1996, pp. 469–474

Support of low delay data services in GSM

J. Dunlop, J.A. Martinez and J. Gozalvez

The authors examine a potential method for supporting UMTS low delay data (LDD) services [1] in a GSM system, using a combination of slot interleaving and ARQ. The parameters of the UMTS LDD service to be accommodated are: delay < 30ms and BER < 10^{-6}.

Introduction: The third generation UMTS will provide several bearer services with specified parameters, the most demanding of which is the low delay data service. As third generation systems will coexist with second generation systems, such as GSM, it is of interest to examine the possibility of supporting third generation bearers on a GSM infrastructure. In standard GSM, the data services operate either in transparent mode or non-transparent mode using a frame interleaving depth of 22. Ignoring the effect of errors in transmission, interleaving over 22 frames incurs a minimum delay of $22 \times 60/13$ms = 101.5ms which clearly exceeds the 30ms delay constraint. If the system employs a radio link protocol to accommodate uncorrected channel errors, this delay can substantially exceed 200ms.

The interleaving delay can be reduced considerably if interleaving takes place on a slot, rather than on a frame basis. The increase in BER that such a strategy produces may be compensated for by additional forward error correction supplemented with an ARQ mechanism. This Letter examines combinations of these techniques which will allow both the delay constraints and the residual bit error rate of the LDD service to be accommodated by GSM.

Channel coding: The results presented in this Letter are based on the use of the (240, 216) cyclic error detection code used as the inner code in the GSM non-transparent data service. For this code, which has a frame check sequence of 24 bits, if the offset between the first and last error is < 25, the error detection probability is 100% [2], the non-detection probability for an offset of 25 is 1.19×10^{-7}, which is sufficient to fulfil the requirement of BER < 10^{-6} for the LDD service. The outer FEC code is the standard half rate punctured convolutional code, also specified for the GSM non-transparent mode data service [3].

Slot interleaving strategy: The effect of implementing the slot interleaving strategy is to increase the frequency of ARQ transmissions, hence it is necessary to determine whether this increase in the use of ARQ can accommodate the delay constraints of the LDD service. The slot interleaving strategy is based on the assumption that a mobile will occupy several slots in each frame simultaneously which is in line with the provisions of the GSM high speed circuit switched data (HSCSD) service recommendations. When multiple slots in each frame are utilised in this fashion, interleaving can be less effective due to the possibility of correlated errors between adjacent timeslots. The channel models provided in the COSSAP stream driven simulator can accommodate this correlation and have been used to produce the results presented in this Letter.

Performance measures: The performance measures of direct interest are throughput and delay. The delay is expressed in terms of the requirement that 95% of all transmitted frames should experience a delay of < 30ms. A major contribution to the delay is produced by the ARQ mechanism which depends on the frame error rate. The results presented in this Letter are for a typical urban channel at a mobile speed of 50km/h (TU50) and are based on a 'selective' repeat ARQ technique in which the transmitter retransmits only frames containing errors. Frame error rate is thus a good indicator of the frequency of use of the ARQ mechanism. Normalised throughput is defined as number of frames successfully transmitted per frame transmission interval.

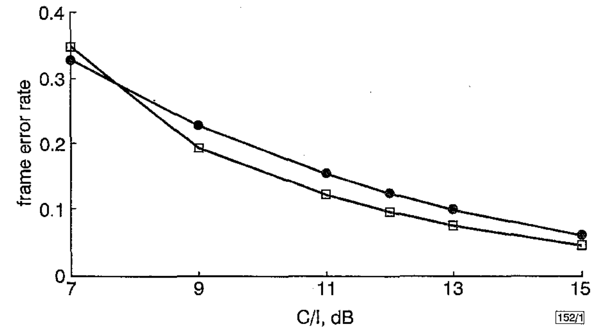

Fig. 1 *Frame error rates for GSM with frame and slot interleaving depth = 8*

—●— slot interleaving
—□— frame interleaving

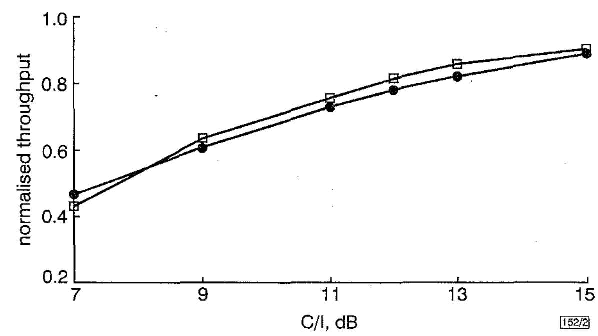

Fig. 2 *Throughput for GSM data channels, interleaving depth = 8*

—●— slot interleaving
—□— frame interleaving

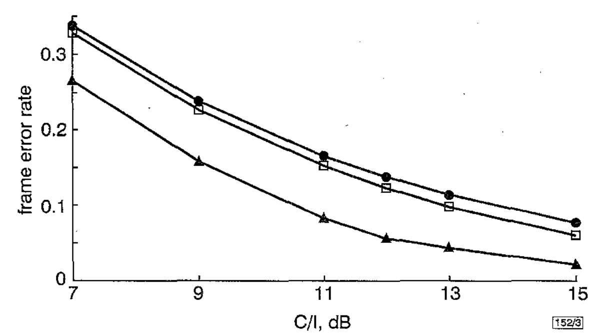

Fig. 3 *Frame error rates for different slot interleaving depths*

—●— slot interleaving = 4
—□— slot interleaving = 8
—▲— slot interleaving = 22

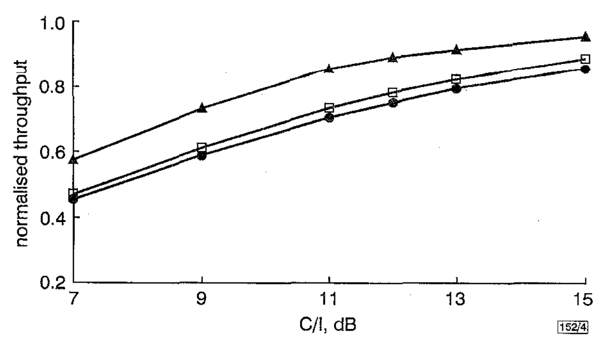

Fig. 4 *Throughput for different slot interleaving depths*

—●— slot interleaving = 4
—□— slot interleaving = 8
—▲— slot interleaving = 22

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Simulated performance results: Fig. 1 shows the frame error rate for a frame and slot interleaving depth of 8. The slot interleaving curve is slighty more accurate, having been obtained over a higher number of transmitted bits. Frame interleaving has a marginally better performance in terms of FER over most of the C/I range. However, due to the more rapid response to ARQ the slot interleaving scheme actually has a higher normalised throughput (Fig. 2).

A comparison of the effect of varying the *slot* interleaving depth on FER and normalised throughput is shown in Figs. 3 and 4. The FER is smallest for an interleaving depth of 22 slots and hence the throughput is greatest at this depth of interleaving.

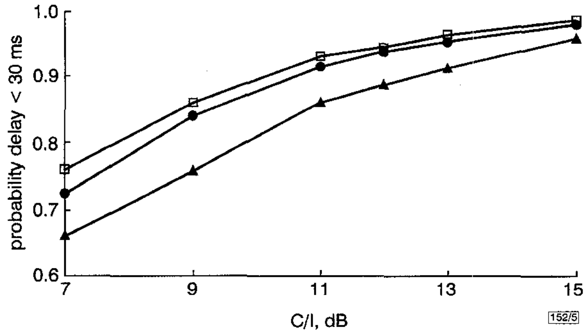

Fig. 5 *Probability of transmitted GSM data frames having a delay < 30ms*

– – – – threshold = 0.95
—●— slot interleaving = 4
—□— slot interleaving = 8
—▲— slot interleaving = 22

The most important characteristics, from the viewpoint of the LDD service, are shown in Fig. 5 which shows the probability of obtaining a delay of < 30ms. If the delay requirement is that at least 95% of transmitted frames should experience a delay < 30ms, a slot interleaving depth of 8 gives the highest probability of meeting this constraint.

Conclusions: This Letter has illustrated that GSM may support the most demanding of the UMTS bearers, i.e. the low delay data bearer. To accomplish this it is necessary to vary the standard interleaving scheme employed in GSM data channels. This Letter has illustrated that an acceptable performance may be realised in terms of delay and residual bit error rate if a slot interleaving scheme is implemented in conjunction with a selective repeat ARQ mechanism.

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References

1 URIE, A., STREETON, M., and MOUROT, C.: 'An advanced TDMA mobile access system for UMTS', *IEEE Personal Communications*, 1995, **2**, (1), pp. 38–47
2 LIN, S., and COSTELLO, D.J.: 'Error control coding – fundamental and applications' (Prentice-Hall Inc., Englewood Cliffs, NJ, 1983)
3 GSM Technical Specification: 'Digital cellular telecommunications system (Phase 2+) Channel coding (GSM 05.3)'. Version 5.2.0, August 1996

Virtual path long-term bandwidth allocation algorithm for ATM networks using simulated annealing

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The long-term virtual path (VP) bandwidth allocation process requires the use of heuristics or artificial intelligence to find the number of virtual circuits that can multiplex by each VP to maximise the quality of service, which is defined here in terms of the call blocking probabilities. The authors propose a new technique, based on simulated annealing, for performing the allocation process. It substantially improves upon previous proposals.

Introduction: There are two types of management functions that can be exercised over virtual paths (VPs): the definition of their routes, and the definition of their bandwidths. The latter, which is the object of this study, adjusts the VP bandwidths to small changes in traffic to achieve a quality of service (QoS) that is as good as possible [1]. A centralised management is chosen for this work [2]. The problem can be formalised in the following terms. Given: (i) a set of switching nodes, a set of links of known bandwidth and the physical topology of the network; (ii) the traffic matrix, i.e. the traffic between each pair of nodes; and (iii) the VP routing table at each node. Minimise: a cost function, which depends on the call blocking probabilities (CBPs) of the VPs. Choosing: the bandwidth assigned to each VP. And satisfying the constraint: the sum of the VP bandwidths in a link cannot exceed the maximum link capacity.

The CBPs are computed by the well known Erlang loss model. With no loss of generality we have assumed that the equivalent bandwidth required to support a VC is not dependent on the number of VCs multiplexed on the same VP [3].

Classical solving methods cannot be used to solve this problem, but heuristics and artificial intelligence propose innovative alternatives, some of them based on natural phenomena, such as simulated annealing [4]. The final solutions obtained by these techniques are not the absolute optimum but are close-enough in a limited period of time.

Simulated annealing: A crystal represents a state of minimum energy at a given temperature. The thermodynamic model of the formation of a crystal is known as the Metropolis algorithm. It was applied to optimisation problems by S. Kirkpatrick in the 1980s. A simulated annealing search starts with an initial feasible solution. A new point in its neighbourhood is generated and the increase in the cost, and difference in quality, is calculated. If the cost is minimised, as intended, the new solution is always admitted, but in the opposite case, it is accepted with a probability given by $p = \exp(-\Delta cost/T)$ The difference in cost plays the same role as the energy increase in the thermodynamic model, and T acts as the temperature, which is slowly and gradually reduced as iterations are performed. The higher the value of T, the greater the ability of accepting non-improving solutions in order to avoid being trapped at local minima. As T is reduced, the exploration turns into the exploitation of a zone of the search space to provide the final minimum in that promising part. It is necessary to define the way in which a new solution is generated from another one: the initial and final values of T, the number of iterations per temperature step (each reduction in temperature), the number of those steps, and a function to decrease T. Several cost functions have been tested in this study, for example: the worst call blocking probability (WCBP), the average CBP, and the mean of both or the traffic carried by the network.

Algorithms: The technique of simulated annealing acts as a metaheuristic procedure to guide the search, but an internal mechanism which provides a new solution from an existing one must be defined: this is the heuristic part of the algorithm.

When the WCBP is used as the cost function, the execution of the heuristic part is defined by the following steps: *A1* Parameter initialisation. *A2* Introduction of traffic changes. *A3* Find the VP with the WCBP. *A4* Calculate the minimum free bandwidth avail-

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