From the results, we find that although the best performance is obtained with reflection coefficients and LARs, the system is more sensitive to the changes in σ and α_g based on the spectral distance measure (Fig. 3). Conversely, cepstrum coefficients yield good results with less relative sensitivity to the width parameters (σ and α_g), which means that the system is less sensitive to speech contents when using cepstrum coefficients.

Conclusion: We have proposed a new method to approximate the event functions by variable-width Gaussian functions for TD. The efficiency of the method has been shown through comparison with fixed-width event approximation using parametric and spectral distance measures. The overall results show the suitability of the adaptive approximations in TD-based speech coding. Among different parameter sets used in our experiments, cepstrum coefficients yielded the best results due to least sensitivity to the width parameters. Investigation is currently being carried out to improve the width-adaptation performance.

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Code configuration for avoiding error propagation

K.A.S. Immink

Indexing terms: Constrained codes, Error correction

A new configuration of codes is proposed for translating user information into a constrained channel sequence. The new configuration makes it possible to practically apply constrained channel codes whose efficiency approaches the channel capacity.

Introduction: In transmission and recorder systems, source data is commonly translated in two successive coding steps: (i) error-correction coding and (ii) channel (or modulation) coding. The two coding steps are illustrated in Fig. 1. Error-correction control is realised by adding extra (parity) symbols to the conveyed message. The family of Reed-Solomon codes (RS) is of major importance for recording applications. A channel encoder has the task of translating arbitrary user information plus error-correction symbols into a sequence that complies with the given channel constraints. Examples are spectral constraints or runlength constraints. A good code embodiment realises a code rate that is close to the capacity of the constrained channel, has a simple implementation, and avoids the propagation of errors during decoding [1].

A configuration of codes is developed in this Letter that avoids serious error propagation.

Bliss scheme: Bliss [2] has proposed to revert the conventional hierarchy of the error-correction code and the channel code. A block diagram of the Bliss scheme is shown in Fig. 2. The constrained codewords are treated as the binary input data of an error correcting code in the usual way. In a byte-oriented error control code (ECC), such as the popular Reed-Solomon code, the constrained codewords are grouped into bytes and the parity bytes generated are affixed to the end (or beginning) of the constrained codeword. The parity bytes thus generated do not, in general, obey the prescribed constraints and they are translated with the aid of a second channel code. Provisions have to be made for concatenating the various segments. Decoding is straightforward. We start by decoding the parity information using the second channel code decoder. We can correct the errors in the constrained sequence and then the first channel decoder delivers the source sequence.

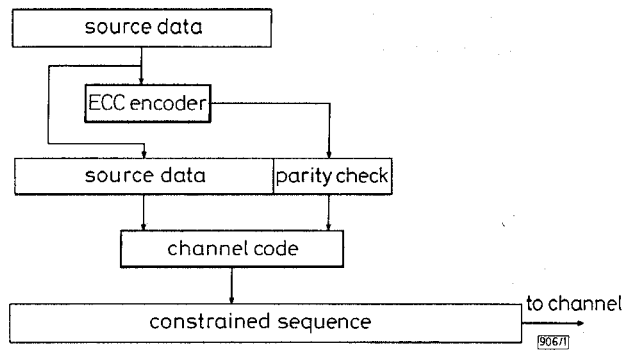

Fig. 1 *Block diagram of two coding steps in recording system*

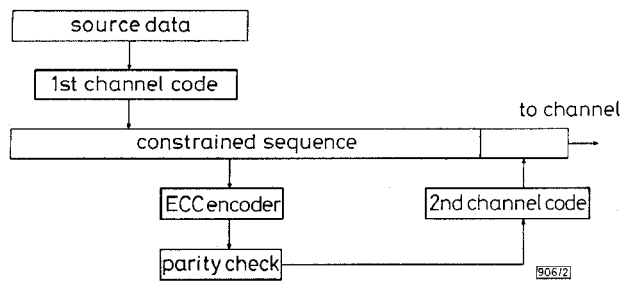

Fig. 2 *Code configuration of post-modulation or 'Bliss scheme'*

The efficiency of the second channel code is, as it is much smaller than the first channel code, lower than that of the first code. However, as the number of parity bits is normally a small fraction of the number of input bits, the efficiency of the second code has a relatively small bearing on the overall efficiency. It is of paramount importance that the error propagation of the second channel code is limited to a few bits, preferably to a single byte in a byte-oriented system.

Burst error correction: In the above 'Bliss scheme', the constrained sequence is the input of the ECC code. Clearly, the constrained sequence is a factor of $1/R_1$ longer than the source data, where R_1 is the rate of the first channel encoder. Assume that the ECC is capable of correcting error bursts of a length of q bytes. Since the ECC operates on channel bytes, the corresponding number of user bytes it can correct is reduced by a factor of R_1. For recording systems this implies that the burst error correction capability measured in geometrical units, e.g. metres, is reduced by the same factor R_1. Secondly, the length of the constrained sequence instead of the user sequence must be smaller than the maximum imposed by the RS code at hand. The above drawbacks of the Bliss scheme are so severe that in spite of its benefits, it is of limited practical usefulness in recording systems, where the correction of burst errors is a major requirement.

These difficulties can be solved by reconfiguring the codes and defining a third intermediate coding layer. The new encoding format is shown in Fig. 3. Essentially, the constrained sequence is compressed into a third intermediate sequence before it is forwarded to the ECC code. The constrained sequence is partitioned into blocks of p bits. The block length p is chosen so that the number of distinct constrained sequences of length p is not larger than N, the field size of the symbol error correcting ECC. It is then possible to define a one-to-one mapping between the p-tuples and the ECC symbols. Using a look-up table **L**, or enumeration, the p-tuples are uniquely translated into an intermediate sequence of bytes. The intermediate sequence, in turn, is used as the input of the ECC encoder and the parity check bytes are generated as

usual. It should be appreciated (see Fig. 3) that the intermediate sequence is not transmitted. As in the Bliss scheme, the parity check bytes are modulated by a second constrained code. The cascaded sequence, i.e. the constrained sequence followed by the constrained parity bytes, is eventually transmitted.

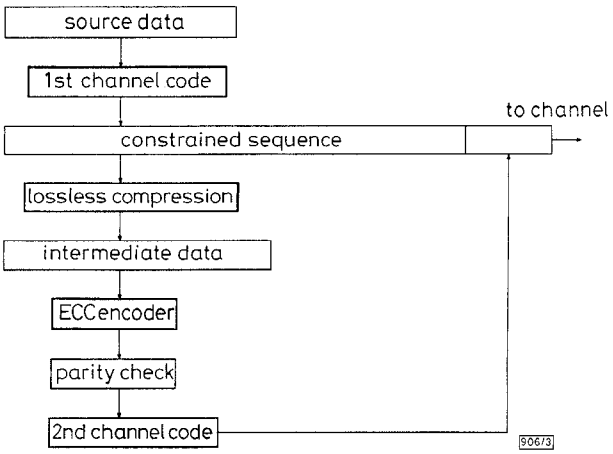

Fig. 3 *Encoder configuration of new coding scheme*

Decoding is straightforward: First, the parity symbols are found using the second channel decoder. The source bytes are found as follows. Using the look-up table **L**, the constrained sequence is translated into a sequence of bytes. They may contain errors, which can be corrected by the RS code. Then, after the ECC decoding operation, the corrected bytes are translated into the constrained sequence using the inverse of the look-up **L**. The corrected constrained sequence is decoded by the first channel decoder.

Conclusions: We have presented a new coding technique for translating user information into a constrained sequence. Serious error propagation resulting from the use of large codewords can be avoided by reversing the order of application of the error control code and the constrained code. A third coding layer has been proposed to improve the burst error correction capacity.

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Comparison of two recycle dilated banyan network with output buffers

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Indexing terms: Telecommunication systems, Switching networks, Banyan networks, Asynchronous transfer mode

Two dilated banyan networks with recirculation and deflection routing algorithm are proposed. Without input buffers, the proposed switching fabrics can fully reuse the routing capacity wasted in a pure dilated banyan network. An analytical model of two switching fabrics and a study of packet loss probability are presented.

Introduction: A large number of ATM switch proposals are based on banyan networks due to their following features: low modular hardware complexity, a self-routing property and suitability for VLSI implementation. However, a banyan network is an internal-blocking network since there is a unique path for each inlet-outlet pair. Many solutions have been proposed; dilation [1] is one which reduces the blocking and improves the throughput.

A dilated banyan network replaces each internal link by d links. Internal blocking occurs only when $d+1$ or more packets all select the same logical outlet of an SE. There are N inlets and N outlets; instead of dropping packets which lose the conflict, we can retransmit them through the network. This method allows wasted links in a pure dilated banyan network to be reused. In this Letter, we propose two switching fabrics, dilated banyan networks with recirculation, and compare their performance.

Switching fabric and routing algorithm: The recycle dilated banyan network (RDBN) is composed of a concentrator, a d_dilated banyan network, a distributor, packet filters and output buffers. The expanded recycle dilated banyan network (ERDBN) is composed of of an RDBN, except for the distributor. As an example, Fig. 1 shows the switch fabric using a 2_dilated banyan network of size 8 × 8 with two recycle links.

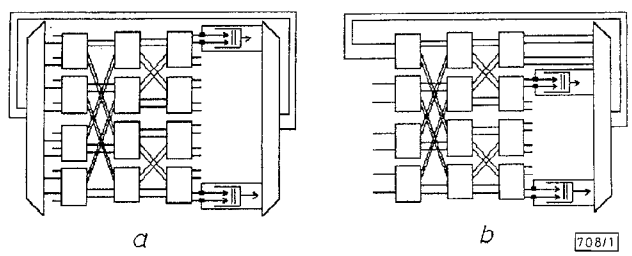

Fig. 1 *Banyan networks*

a Recycle dilated banyan network (RDBN)
b Expanded recycle dilated banyan network (ERDBN)

The new packets arriving at the inlets are transmitted directly to the network; when internal conflict occurs, the proposed switch networks use a deflection routing algorithm [2] in the operation of SE. When $i \le d$ packets are destinated for the same outlet of an SE, all of them are routed to the i links of an SE. If $i > d$, d packets are routed to the correct outlet and i–d packets are routed the wrong way with deflection tags. In the remaining stages, the deflection packets are routed randomly so as not to affect the path of normally routed packets. At the outlet of the network, the address filter examines the deflection tags of arriving packets. If tag = 1, then the deflection tags are removed and are transmitted to the concentrator. Otherwise, normal packets are transmitted to the output buffers. The concentrator has $d \times N$ inlets and T outlets (T is the number of recycle links). If $i > T$ misrouted packets are transmitted to the inlets of the concentrator, T of them are randomly choosen transmitting to the outlet and others are dropped. If $i < T$, all can be transmitted to the T recycle links. The misrouted packets can therefore multiplex T recycle links. For RDBN, the recycle links are attached to the idle inlets of SEs at the first stage, whereas for ERDBN, the recycle links are attached to the inlets of the special SEs, so $N \times N$ ERDBN only has N–T inlets to receive new packets with T inlets for misrouted packets.

Performance analysis: Consider a d_dilated network of size $N \times N$, which is constructed with $n = \log N$ stages. We assume that the traffic is uniform. For brevity, it is supposed that the output buffers are infinite so that packet loss only occurs at the concentrator.

Let $q_{k,t}(l, m)$ denote the probability of finding l normal packets, and m deflected packets at one outlet of an SE in stage k at time slot t; $q_{k,t}(l, m)$ has been derived in [3].

Thus, the probability of j packets appearing at inlets of the concentrator, $\alpha(j)$, is given by

$$\alpha(j) = \sum_{\substack{1 \cdot k_1 + \ldots + d \cdot k_d = j \\ k_0 + k_1 + \ldots + k_d = N}} \frac{N!}{k_0! \cdots k_d!} p_0^{k_0} \cdots p_d^{k_d} \qquad (1)$$

where $p_l = R(l) = \sum_{x=0}^{d-1} q_n(x, l)$, $0 \le l \le d$, $q_n(x, l)$ is the steady state value of $q_{n,t}(x, l)$; so, the probability that k packets arrive at the outlets of concentrator $\gamma(k)$ is given by