S_2 and S_3. These sets of consecutive unused slots may be maintained as a linked list; any set that is replaced is removed from the list. Whenever there is a new CBR stream, the largest set of consecutive unused slots will be selected from the linked list, the middle slot of that set will be the first slot allocated for that stream, and the linked list will be updated accordingly.

Results: Four other algorithms [7] are used for comparison:

(i) Random algorithm: use a slot chosen at random

(ii) Sequential algorithm: always use the leftmost available slot

(iii) Binary algorithm: identify the leftmost set of consecutive available slots; use the slot that is midway in this leftmost set

(iv) Alternate binary algorithm: for one request, identify the leftmost set of consecutive available slots; use the slot that is midway in this leftmost set; for the next request, identify the rightmost set of consecutive available slots; use the slot that is midway in this rightmost set; alternate between the leftmost and rightmost set of consecutive available slots for each new request.

Simulations were carried out for different slot occupancies, assuming a fixed number of slots per stream. Fig. 3 shows the case for 10 slots per stream. Percentage occupancy is the total number of allocated slots as a percentage of the total number of slots in the scheduling frame.

It is found that the performance of the five algorithms converge as the percentage occupancy of the scheduling frame increases beyond 80%. At such high occupancies, the frame has few unoccupied slots. There is little choice as to which slots to use for any new connection request. The allocation algorithm correspondingly has little role to play. At lower occupancies, the largest-first algorithm performs better than the other algorithms. Similar results were obtained for different numbers of slots per stream.

Conclusion: In an ATM switch that uses a TDM scheduling frame to allocate time slots for CBR traffic, the scheduling algorithm has a major impact on other traffic types. For allocating slots to mono-rate CBR traffic, simulation results have shown that the new largest-first algorithm gives the most even distribution of unused slots compared to four other algorithms. These slots can be used by other traffic types which will therefore encounter the least delay and jitter.

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Re-optimisation of LPC filters for multi-pulse coded excitation

D.P. Palomar, M. Price and M. Sandler

A method for optimising LPC filters in linear prediction based speech coders is described. The optimisation process compensates for errors incurred through coding the excitation signal, providing an improvement in the quality of the decoded speech, with no increase in bit rate.

Introduction: For the linear predictive coding of speech, LPC filters are designed to minimise the energy of the residual (error) signal [1]. Hence, they are optimised for recovering speech with the residual as the excitation.

In practice, however, coding schemes use a coded excitation signal to minimise the bit rate [2], resulting in a loss of some of the information within the residual signal. Hence, the original LPC filters are not optimal for recovering speech. Although recent articles suggest that work is continuing to enhance the quality of standard speech coding methods [3, 4], this particular issue has not yet been addressed directly.

We therefore propose a coding scheme whereby the LPC filters are re-optimised for recovery of the original speech from the coded excitation signal, prior to transmission to the decoder. We are specifically interested in multipulse excitation coding, but the scheme could be equally applied to other coding methods.

The scheme is depicted in Fig. 1, where $y(n)$ is the original speech signal, $\tilde{e}(n)$ is the coded excitation signal, and $\tilde{a}_i$ represents the re-optimised LPC filter coefficients.

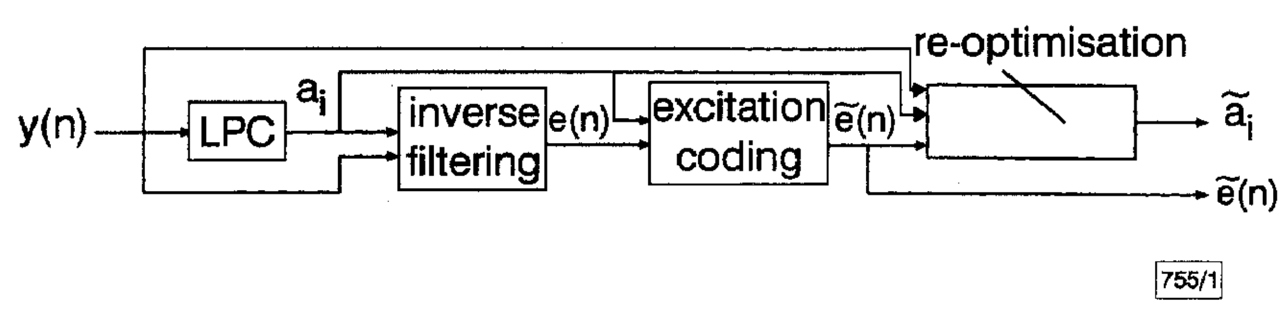

Fig. 1 *Block diagram of coding scheme*

Description of problem: The linear prediction analysis stage of a speech coder generates the coefficients a_i, which form the LPC filter $H(z)$, as defined in eqn. 1:

$$H(z) = \frac{1}{A}(z) = \frac{1}{1 - \sum_{i=1}^{p} a_i z^{-i}} \qquad (1)$$

The error signal $e(n)$, is generated by *inverse filtering* the speech signal as follows:

$$E(z) = YA(z) \qquad (2)$$

where the original speech signal $Y(z) = Z\{y(n)\}$, and the error $E(z) = Z\{e(n)\}$. Hence, the LPC filter and the error signal could be used in the decoder to recover the original as follows:

$$Y(z) = EH(z) = \frac{E}{A}(z) \qquad (3)$$

However, the error signal $e(n)$ is usually coded, yielding a compressed error signal $\tilde{e}(n)$. Hence, eqn. 3 becomes

$$\tilde{Y}(z) = \tilde{E}H(z) = \frac{\tilde{E}}{A}(z) \qquad (4)$$

where $\tilde{Y}(z) = Z\{\tilde{y}(n)\}$ is the recovered speech signal.

To re-optimise the LPC filters for $\tilde{e}(n)$, we must minimise $\Sigma_n(y(n) - \tilde{y}(n))^2$ as follows:

$$\sum_{n=j}^{F-1}(\lambda(n)\tilde{y}(n-j)) = \sum_{i-1}^{p}\tilde{a}_i \sum_{n=j}^{F-1}(\tilde{y}(n-i)\tilde{y}(n-j)) \quad \text{for } j = 1, 2, ..., p \qquad (5)$$

where F is the length of the analysis frame, $\lambda(n) = y(n) - \tilde{e}(n)$, and $\tilde{a}_i$ are the re-optimised coefficients. As $\tilde{y}(n)$ and $\tilde{a}_i$ are mutually dependent unknowns, this is a recursive problem, and it is therefore difficult to solve analytically.

Re-optimisation algorithm: To solve the re-optimisation problem, we propose iterative refinement of the LPC coefficients a_i, to form the re-optimised coefficients $\tilde{a}_i$. We define the operations of the kth iteration as the computation of

$$\tilde{y}^{(k)}(n) = \tilde{e}(n) + \sum_{i=0}^{p} \tilde{a}_i^{(k-1)} \tilde{y}^{(k)}(n-i) \qquad (6)$$

and the solution of

$$\sum_{n=j}^{F-1} \left(\lambda^{(k)}(n)\tilde{y}^{(k)}(n-j)\right) = \sum_{i-1}^{p} \tilde{a}_i^{(k)} \sum_{n=j}^{F-1} \left(\tilde{y}^{(k)}(n-i)\tilde{y}^{(k)}(n-j)\right) \quad \text{for } j = 1,2,...,p \qquad (7)$$

where $\lambda^{(k)}(n) = y(n) - \tilde{e}(n)$.

As the convergence of this procedure is not guaranteed, we can adjust the coefficients $\tilde{a}_i^{(k)}$, by computing a weighted average with the coefficients of the previous iteration $\tilde{a}_i^{(k-1)}$, according to eqn. 8:

$$\tilde{a'}_i^{(k)} = \alpha \cdot \tilde{a'}_i^{(k-1)} + (1-\alpha) \cdot \tilde{a}_i^{(k)} \quad \text{for } i = 1,2,...,p \quad (8)$$

where α is a constant value between 0 and 1.

Hence, at each iteration of the above procedure, the performance of the resulting LPC filter is compared with that of the currently best-candidate filter, (we have used SNR in our study). If there is an increase in performance, then the new coefficients $\tilde{a'}_i^{(k)}$ are stored as the new current best-candidate filter. On termination, the current best-candidate filter coefficients are used as the re-optimised coefficients $\tilde{a}_i$.

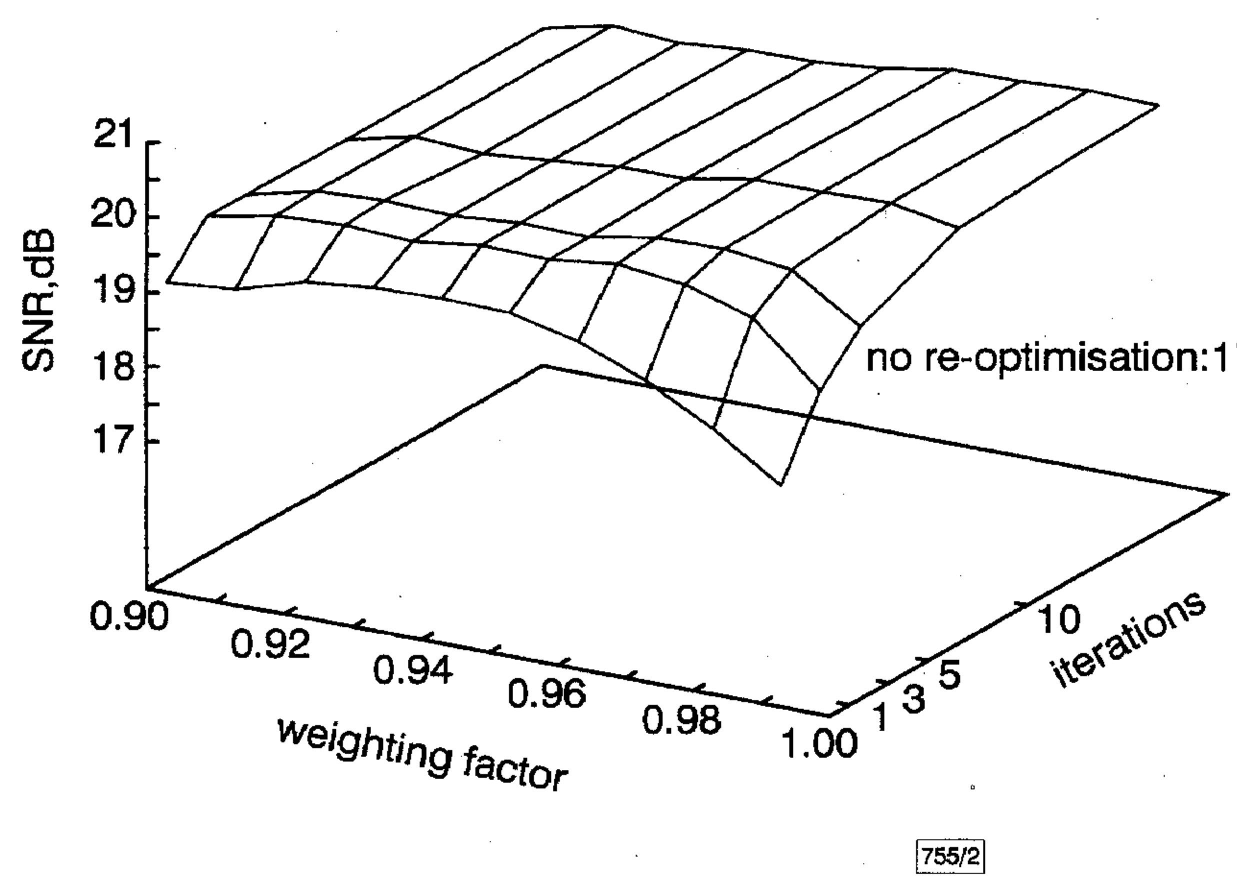

Fig. 2 *Re-optimisation surface for 'AM1'*

Performance advantage of re-optimisation: We measured the performance advantage gained from LPC filter re-optimisation with the proposed algorithm, using a 'standard' speech coding scheme with 'multipulse excitation' coding. We used a set of five speech sequences (two male adults, two female adults, and one female child), each sampled at 8kHz. These sequences were chosen to provide a wide variation in voice characteristics [5].

The LPC analysis was applied with a frame length of 160 samples and an order of 16, and the multipulse excitation coding was applied with 56 pulses per LPC frame. These values were chosen quite arbitrarily within the range of valid settings for narrowband speech to highlight the improvements possible. Note also that the LPC coefficients and the multipulse amplitudes were unquantised. Segmental SNR [1], computed over N frames, was used to measure the improvement obtained with re-optimisation.

Although the speech sequences gave a wide range of voice characteristics, the re-optimisation surfaces of the speech sequences had very similar features. This suggested that the re-optimisation process was largely independent of the speech data, i.e. near-optimal re-optimisation parameters for one speech sequence are likely to be near-optimal for any other speech sequence.

We also observed that the largest improvement was achieved with the first iteration. Furthermore, the largest improvement with a single iteration was always achieved with $\alpha \simeq 0.94$, yielding an increase in segmental SNR of > 2dB in each case.

A single re-optimisation iteration represented an increase in complexity approximately equivalent to adding an extra LPC analysis stage. To quantify this increase in complexity, we compared the CPU time per LPC frame of each of the test speech signals for the coder with (single iteration) re-optimisation against that for the coder with no re-optimisation, both implemented on a Sun Microsystems 'Ultra 5'. On average, re-optimisation increased the CPU time from 67.8ms per frame to 71.8ms per frame, which represents an increase of just under 6%

Table 1: Segmental SNR, in dB, for α = 0.94

Sequence	0 iterations	1 iteration	5 iterations	10 iterations	20 iterations
'AM2'	17.9054	20.188	20.6686	20.7036	20.7088
'AF1'	18.3897	20.6745	21.0224	21.0698	21.0698
'AF2'	18.1003	20.9136	21.1233	21.1233	21.1233
'CF1'	16.2349	18.444	18.9587	18.9587	18.9587

The re-optimisation surface for one of the speech sequences is plotted in Fig. 2. The results for all of the other sequences, with α = 0.94, are given in Table 1.

Conclusions: In this Letter, we have described a method for re-optimising LPC filters in a linear prediction based speech coder with multipulse coded excitation. Our results suggest that quite a significant improvement can be achieved with a relatively small increase in the overall complexity of the coder, zero increase in complexity of the decoder, and zero increase in bit rate.

As LPC re-optimisation is applicable irrespective of the coding parameters or the type of excitation coding, a more extensive investigation is highly recommended.

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RZ-Gaussian pulses reduce the receiver complexity in wireless infrared links at high bit rates

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The authors conclude that the on-off keying (OOK) format with return-to-zero Gaussian (RZ-Gaussian) pulses provides better performance than other signalling formats, more frequently proposed in reported works, allowing the adoption of a low-complexity receiver, and obviating the need for baseline restoration and equaliser filters. The above conclusion is validated by the corresponding bit error rate (BER) computation.