

# Signal shaping dual modular redundancy for soft error tolerant finite impulse response filters

P. Reviriego, C.J. Bleakley and J.A. Maestro

A technique to protect finite impulse response (FIR) filters against soft errors is presented. The approach is based on the use of two copies of the FIR filter. In one of the copies, preprocessing of the input and a postprocessing of the output are added. In the event of a soft error, the outputs of the filters differ or mismatch for one or more samples. The additional processing introduced in the second copy of the filter ensures that the mismatch patterns are unique to each copy. Hence, the copy in error can be identified and the output of the other copy selected as the final error protected filter output. The proposed scheme can efficiently correct isolated soft errors at lower cost than general techniques, such as triple modular redundancy.

**Introduction:** Soft errors are a major concern for electronic circuits [1]. When a soft error occurs, it causes a transient error that can affect combinational and sequential circuit elements. As process geometries decrease, the probability of soft errors increases significantly. Therefore, the design of soft error tolerant circuits is becoming increasingly important [2]. Protection of circuits against soft errors can be tackled at various levels. For example, the manufacturing process can be modified to make the devices more resilient against soft errors [2]. Alternatively, the circuit design can be modified to incorporate redundancy such that errors can be detected and corrected by voting, as in triple modular redundancy (TMR) [2]. In addition, in the case of signal processing circuits, the algorithmic properties of the relationships between the inputs and outputs can be used to detect and correct errors [3].

Finite impulse response (FIR) filters are among the most commonly used signal processing circuits [4]. Many techniques have been proposed to protect them from soft errors. For example, in [5], the use of reduced precision replicas to detect and correct errors in FIR filters is proposed. In [6], the use of dual modular redundancy (DMR) employing a different structure for each copy (transpose and transpose-cascade) is proposed. Errors are detected and corrected by identifying the patterns of mismatches between the filter outputs.

In this Letter, we propose a novel technique for protecting FIR filters from soft errors. As in DMR, two copies of the filter are used. The first copy has a transposed direct form structure. Assuming that there is no logic sharing between multipliers, a single soft error hitting the filter causes an output error in only one sample. The second copy has the same transposed direct form at its core but the input signal is filtered using a simple FIR filter and the output signal is filtered using its inverse. In the event of a single soft error, the inverse filter causes an output error in several samples. Thus, by counting the number of samples in error, we can determine which filter copy suffered the soft error. If one mismatch occurs, the error is in the first copy, whereas, if more than one consecutive mismatch occurs, the error must be in the second copy. The technique is somewhat similar to that presented in [6]; however, the proposal herein has the advantage that the same FIR filter structure is used in both copies. This simplifies implementation and is more suitable for implementation of adaptive filters.

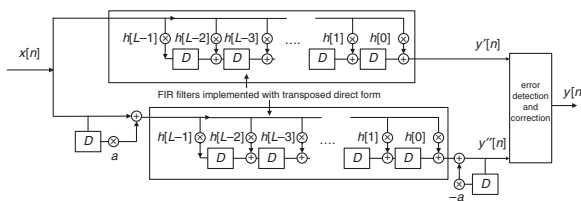


Fig. 1 Signal shaping dual modular redundancy implementation

**Signal shaping dual modular redundancy:** An FIR filter with impulse response  $h[i]$  performs the following operation on an input signal  $x[n]$  to obtain the output  $y[n]$ :

$$y[n] = \sum_{i=0}^{L-1} h[i]x[n-i] \quad (1)$$

The proposed signal shaping dual modular redundancy technique is illustrated in Fig. 1. It can be observed that two identical copies of the FIR filter are used but the input to one of them is preprocessed (shaped) and its output postprocessed (de-shaped). The shaping filter is a one-tap FIR filter that implements the following transfer function in the  $z$ -domain:

$$S[z] = 1 + az^{-1} \quad (2)$$

Correspondingly, the de-shaping is an infinite impulse response (IIR) filter that implements the following transfer function in the  $z$ -domain:

$$D[z] = \frac{1}{1 + az^{-1}} \quad (3)$$

Therefore their combined transfer function is an identity such that, in the absence of error, the outputs from the two copies  $y'[n]$  and  $y''[n]$  are equal.

When an error affects the copy that is not shaped, the output  $y'[n]$  will be in error for one sample. However, when an error affects the copy that is shaped, the output  $y''[n]$  will be in error for more than one sample. This is because the de-shaping implements an IIR filter that propagates the error to subsequent samples. Errors in the shaping or de-shaping elements would also produce errors that affect more than one sample of output  $y''[n]$ .

The different error patterns can be used to correct the soft errors by implementing error detection logic that compares outputs  $y'[n]$  and  $y''[n]$ . If the outputs differ for a single sample but are equal for the previous and next sample then the error occurred in  $y'[n]$  and  $y''[n]$  is selected as the final output. Conversely, if the error affects more than one consecutive sample, then the error has occurred in  $y''[n]$  and  $y'[n]$  should be selected as the final output. This logic can be implemented efficiently using the circuit shown in Fig. 2. In this implementation, an isolated single soft error event in the error detection logic will not change the final output  $y[n]$ . Therefore there is no need to add redundancy to protect the correction logic. Since soft errors are rare, we assume that they are isolated, that is, only a single soft error affects the circuit at any one time [1].

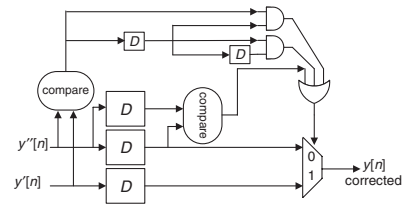


Fig. 2 Implementation of error correction logic

Although the two copies implement exactly the same FIR filter, the input signals to the filters are different and therefore the quantisation errors will be not be the same; this leads to small differences between the outputs even in the absence of errors. To alleviate this problem, a threshold is used in the comparison of  $y'[n]$  and  $y''[n]$ . This means that errors below the threshold are not detected. For an error  $e$  at the input to the de-shaping filter, the output will have an error of  $e$  in the first sample,  $ea$  in the second sample,  $ea^2$  in the third sample and so on. Therefore, for a threshold  $t$ , errors smaller than  $t/a$  will not be corrected. From an implementation perspective the value  $a = 0.5$  is attractive as it simplifies the shaping and de-shaping circuits. In this case, errors smaller than  $2^*t$  will not be corrected.

Another issue is that the shaping of the input introduces a gain of up to  $1 + a$  that can cause saturation in the filter. In many cases, the errors introduced by saturation will be treated as soft errors and corrected. However, saturation can be completely avoided by dividing the signal by two after shaping and multiplying it by two after de-shaping. This introduces some additional quantisation noise.

**Evaluation:** The signal shaping DMR technique has been implemented in Matlab using fixed-point quantisation of 16 bits for the input signal, coefficients and outputs. A value of  $a = 0.5$  has been used for shaping. The threshold used in the comparison is  $t = 0.001$ , which is above the quantisation noise for all filters. To avoid saturation the signal is divided by two after shaping and multiplied by two after de-shaping. Two different filters have been tested: a lowpass and a highpass filter. For each filter type, filters of orders 8 and 16 have been evaluated.

Two input signals have been used in the experiments: white noise and white noise plus a low-frequency sinusoid. The input signals were in the range  $-0.5$  to  $+0.5$ .

For each of the configurations, one million errors were randomly inserted. Each error was separated by 100 samples from the previous errors. The error magnitude was also randomly distributed in the range  $-0.5$  to  $+0.5$ . For the unprotected filter, the maximum error at the output was in all cases close to 0.5 as the  $h[i]$  coefficients are smaller than one in all filters. For the filter protected with the proposed technique, the maximum error at the output was logged and the results are shown in Table 1. It can be observed that the maximum error is between 0.0025 and 0.0031. This is close to the value  $t/a = 0.002$  predicted by the theoretical analysis presented in the preceding Section. The discrepancies between the results and the value of  $t/a$  are due to the different quantisation noise in the two copies of the filter. This means that an error that is slightly larger than  $t/a$  may cause a difference between  $y'[n]$  and  $y''[n]$  that is smaller than  $t/a$  and is not corrected. When the quantisation noise is considered, the experimental results are in line with the theoretical analysis and show that the proposed scheme can efficiently correct isolated soft errors.

**Table 1:** Maximum error observed at output for different configurations tested

	White noise	White noise plus sinusoid
Lowpass, order 8	0.0025	0.0025
Lowpass, order 16	0.0029	0.0031
Highpass, order 8	0.0026	0.0026
Highpass, order 16	0.0027	0.0028

For FIR filters of moderate/large order and bit-width, the overhead of the proposed technique is small. This is because when  $a = 0.5$  no multipliers are needed for the shaping and de-shaping processing. This is an advantage when compared with the technique proposed in [6] where additional multipliers were needed. The correction logic in Fig. 2 is also negligible compared to an FIR filter with programmable multipliers. Therefore, the implementation cost of the proposed technique would be slightly greater than two times that of the original, unprotected FIR filter.

Finally it is worth mentioning that the FIR filter implementation is the same in both copies. This simplifies implementation of the technique, particularly for hard-macro filters. For adaptive filters, the use of two identical copies is an advantage since decomposition of one of the

transposed directly form to a transposed-cascaded structure, as proposed in [6], can be difficult to implement for coefficients that vary continuously.

*Conclusions:* In this Letter, a technique to protect finite impulse response filters against soft errors is presented. The proposed technique uses a modified dual modular redundancy structure, thus reducing the implementation cost compared to general techniques, such as triple modular redundancy. The technique can efficiently detect and correct isolated soft errors as shown in the evaluation. The proposed scheme can be combined with reduced precision redundancy by using reduced precision in one of the copies. This would further reduce the implementation cost. The study of such a combination is left for future work.

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