Efficient Analog Test Methodology Based on Adaptive Algorithms

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Abstract

This papers describes a new, fast and economical methodology to test linear analog circuits based on adaptive algorithms. To the authors knowledge, this is the first time such technique is used to test analog circuits, allowing complete fault coverage. The paper presents experimental results showing easy detection of soft, large-deviation and hard faults, with low cost instrumentation. Components variations from 5% to 1% have been detected, as the comparison parameter (output error power) varied from 300% to 20%.

1 Introduction and motivation

Faults in analog circuits can cause different symptoms at the circuit outputs, from short and opening of components to slow deviations of the operation point caused by a degradation in passive or active component characteristics. Presently, there are some reported works that generate a minimum excitation set in order to detect these type of faults [1], [2]. The general idea is to test the response of the circuit to a given frequency. Deviations on circuit parameters caused by any soft or hard fault will affect the output response, either in its module or phase. The test engineer must supply a minimum number of excitation frequencies in order to detect various components faults.

In our methodology, the idea of exciting the circuit with different signals is maintained, by observing the output of the analog circuit and comparing it with an expected output. However, instead of exciting the circuit with a minimum set of frequencies in order to cover hard and soft faults (like in [1]), we excite the circuit with infinite frequencies at the same time (white noise). This way, we not only guarantee complete coverage of faults, but we are also able to detect in an easy way fine grain details like phase or amplitude variations at the output. Deviations of 1% from nominal value of the circuit components have been detected. Any analog function that can be represented in the s-plane by a pole-zero polynomial fraction N(s)/D(s) can be tested.

This paper is organized as follows: section 2 presents the overall idea of our approach, with mathematical explanation of its principles. Section 3 analyses a set of examples, were we show the success of the approach by detecting faults which are not easily detected. In section 4 we present some actual results of the methodology, followed by some limitations (section 5). Finally, section 6 presents our conclusions and future work.

2 Testing with adaptive procedures

2.1 The Adaptive Tester

In [3] and [4] DSP (Digital Signal Processing) techniques had been used to detect misbehaviors of AD converters. Basically, the Fast Fourier Transform was used to identify the presence of harmonics showing signal degradation. Recently, Spina and Upadhyaya ([5]) have used white noise to train an Artificial Neural Network to detect fault patterns in analog circuits. Although we use a similar technique (adaptive filters being related to neural networks as sum of products sequence of operations), we do not try to characterize all circuit patterns, nor we require any knowledge about the system under test. In our approach we first recognize the circuit under test as a plant, with all its pole-zero characteristics, like an analog signature. Then, we apply specific algorithms to detect any deviations from this first obtained analog signature.

The theory regarding linear plant recognition is quite settled [6]. Recently, with the advent of fast DSP microprocessors and boosted by digital communication problems, adaptive filtering took place as a good mathematical framework to solve not only line equalization problems, but plant recognition as well. The idea of an analog plant recognition by the use of an adaptive algorithm is shown in figure 1. A plant is submitted to an excitation, the same being done with the adaptive filter. Their output is compared, and the amount of error helps the adaptive filter change its coefficients in order to track plant performance.

After some time, all filter coefficients have stabilized, and the plant has a twin plant implemented in the filter. Now, one has only to change the output of the error signal in order to transform this adapted plant to a tester, as shown in figure 2. Once we substitute the plant by a circuit to be tested, the previously adapted filter will output an error signal showing how much the new plant is equal to the previous (fault free) one. It is interesting to notice that this explains the robustness of the methodology. The basic idea is an analogy to the detection of variations in a bridge circuit. Any fault in the analog circuit,

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being it soft, hard or catastrophic, will change the plant under test. In other words, any change in the plant signature (its pole-zero characteristics) will increase the power of the error signal. The tester will detect this change and compare it to a previously defined threshold, according to the tolerated variation of the circuit.

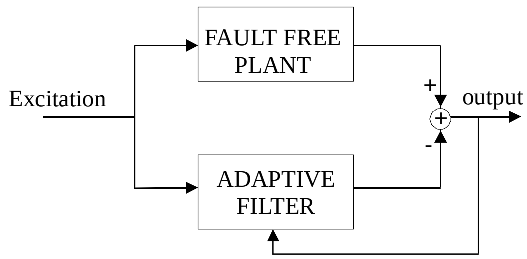


Figure 1. Basic plant recognition structure

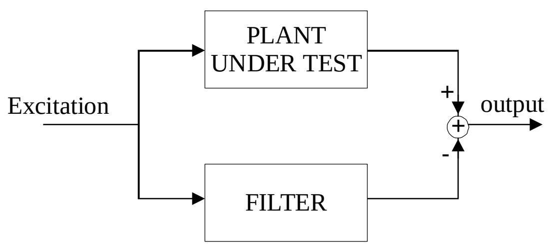


Figure 2. Plant being tested with filter

2.2 The adaptive algorithm and filter topology

A finite impulse response (FIR) digital filter ([7]) can be represented by a polynomial on its input signals delayed by a fixed amount of sampling times, like

$$Y(n) = a_0 * X(n) + a_1 * X(n-1) + ... + a_{m-1} * X(n-m+1) \qquad (1).$$

Y(n) is the output at discrete time **n**, while **m** is the number of filter taps. In an adaptive digital filter all coefficients (a_0 ... a_{m-1}) are determined at runtime. The questions to be answered are how to discover the correct set of coefficients, so that the output of the filter performs the desired functions and, also, the depth **m** of the filter. In case of an adaptive filter, the output should exactly mimic the output of the plant. There are available different algorithms ([8-10]), and their basic trade-off regards the amount of computations that must be performed against the speed of coefficient convergence.

The filter adapts to any plant, provided it has enough taps (pair coefficient-sample) to represent all poles and zeros of the original plant. Actually, the adaptive filter matches the impulse response of the plant. To decide which kind of signal to be used in order to excite the couple filter-plant, one must note that this signal must last long enough for the filter to converge (the error signal must go below a certain threshold), and must be rich regarding its frequency components, in order to provide excitation of the plant at all frequencies of interest.

In our case, we choose one of the simplest adaptive algorithms, the Least Mean Square or LMS, showed in figure 3.

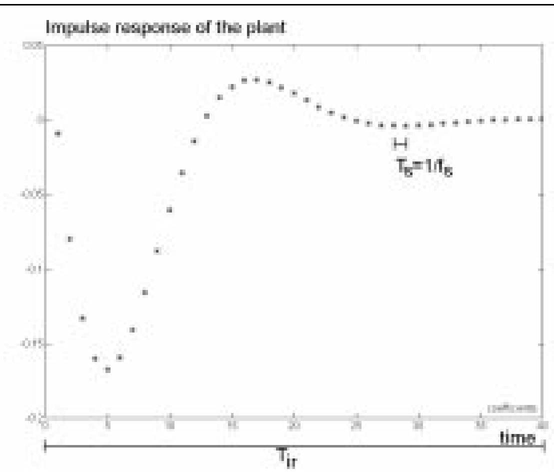
```
begin
ai(0)=0, i = 0...m-1
loop
    Y(n)=SUM(ai(n)*X(n-i)) , i = 0...m-1        /* filter */
    error(n) = d(n) - Y(n)                      /* error */
    ai(n+1) = ai(n) +  µ*error(n)*X(n-i)        /* coefficients */
end
```

Figure 3. LMS algorithm

Once one knows the plant to be mirrored, the sampling frequency should be determined by an evaluation of the plant fastest response to be tracked by the test. In other words, the frequency response of the plant must be under the Nyquist limit, fs/2. The maximum sampling frequency is a function of the data acquisition system.

The designer can now define the number of taps to be used by a set of relations shown in figure 4. To define the number of taps, one should know the sampling frequency (fs) and have an estimate for the duration of the plant impulse response (Tir). In figure 4 we show an example impulse response (after convergence), approximated by a filter with 40 taps.

The µ parameter can be chosen inside its upper and lower bounds ([8,9]), which are related to the power of the input signal applied to the system. The µ parameter can be a trade-off between speed of convergence (large µ) or precision in the error signal (smaller µ).



Ntaps = Tir/Ts
Ts=1/fs

Figure 4. Relations to define the minimum number of taps

2.3 Excitation Signal and the Tester

In order to excite the plant conveniently to detect the largest number of faults, the input signal should be such that all poles and zeros of the plant are properly excited. Any variation in this set of poles and zeros caused by any kind of fault will change the basic plant, being detected. The best signal would be one having all frequencies represented in its spectrum.

Our first approach was to use the impulse response of the system, for in an impulse signal all frequencies are present. The impulse response of a circuit applied to circuit diagnosis was partially used in [11]. The approach of using the impulse response of the circuit, however, is quite dangerous. An impulse can move some linear circuits out of their linear region of operation. Also, defining the correct finite amplitude of a theoretically infinite impulse signal while maintaining linearity could be quite tricky. A better input signal was needed.

White noise has, by definition, an equally distributed power in its frequency spectrum. White noise can be easily generated using a random number generator and a DA converter. Moreover, one can be quite sure about its energy frequency spectrum definition, since we can sample it as it is being fed to the circuit under test and measure its frequency distribution. This way, white noise was the second natural choice.

An AD converter, a PC and a DA converter can tackle the test to be carried constituting a low cost tester. White noise is generated in the PC, and the plant signal is sampled and processed in the same PC. The test engineer can take two tracks. The first one is to define a good plant at an abstract level based on his/her design, compute filter coefficients by any mathematical tool (Matlab, Matcad, etc) and to use this as the reference plant. Another strategy is to develop of a fault free prototype, and then use the system to adapt the filter and save the discovered coefficients to be used in the test procedure. It is important to notice that the choice of white noise guarantees that all frequencies are conveniently excited. This causes that all frequency dependent faults are detected, without the need of complex circuit analysis to define the minimum set of test frequencies.

3 Worked examples

In order to better explain the proposed test methodology, two test circuits will be used. The first one is a simple integrator, followed by a biquad filter, an example a little more complex, also chosen because published data was available in order to make comparisons. This circuit was built and tested for different variations in some of its components.

3.1 Integrator

The integrator used is shown in figure 5. The transfer function of this circuit can be found from elementary circuit analysis to be:

$$H(s) = \frac{-1}{R_1 C} \cdot \frac{1}{S + \frac{1}{R_2 C}}$$

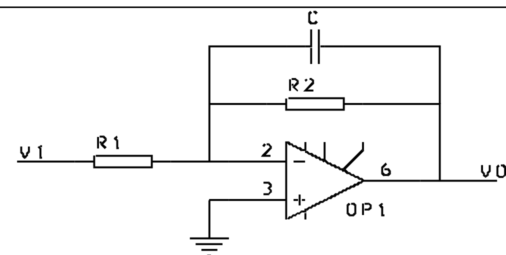
For the simulation of this plant in the digital domain we take its z-transform ([6,7,9]), in series with a zero-order hold:

$$H(z) = \frac{-R_2}{R_1} \cdot \frac{1 - e^{-kT}}{Z + e^{-kT}}$$

$$k = \frac{1}{R_2 C}$$

where T is the sampling period. From this equation we have obtained the recursive equations for the simulation of the plant, using the values for the circuit components presented in figure 5.

The system was simulated using Matlab. We have used the rand function to generate the input to the circuit, and a FIR adaptive filter with 60 taps was adapted for 3000 samples. After conversion, the filter coefficients were stored and a new plant was defined, changing the value of C from 1.0µF to 0.95µF (a -5% variation). The resulting output error of the two cases is shown in figure 6 (for the identified system) and in figure 7 (-5% variation). The estimated output error power ratio has been found to be 13.3475 for this case. Both figures are at the same scale.



R1=1e3; R2=1e3; C=1e-6; T=0.1e-3;

Figure 5. Simple Integrator

3.2 Biquad filter

The biquad filter was used to validate the methodology and a prototype circuit was build from discrete components. The circuit is shown in figure 8, with the nominal values that were used. We have used a PC with a DSP board and stereo audio analogue interface as the test system. The sampling frequency used was 16KHz. The DSP system generated the input signal for the system, and a 2-channels AD sampled the input and the output of the plant.

We have used 40 taps for the adaptive filter, and adapted the system during 3000 samples. All data processing after acquisition was made in the Matlab environment. Table 1 presents the output error power ratio obtained for variations in components R2 and C2. The error power ratio shows how easily detected is a fault. A ration of 1 means an exact match between the plant and the adaptive filter. We cover soft, hard and catastrophic faults like a short or open circuit. All results were generated from real acquired data using variations introduced in the original system. For the case with no variations, the same data file was used, but at different time intervals.

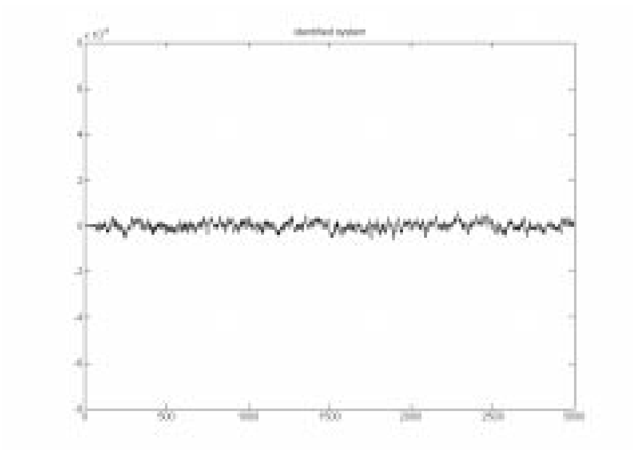


Figure 6. Error output of a good plant (identified system)

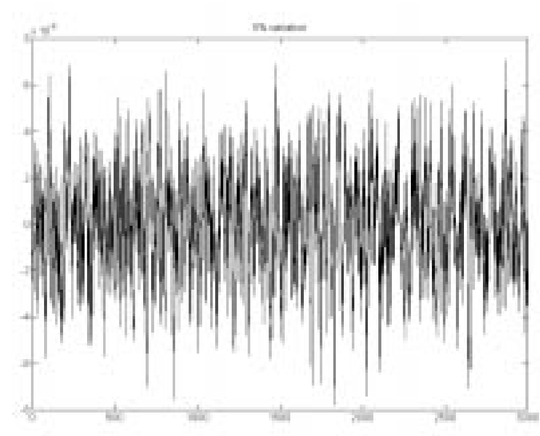
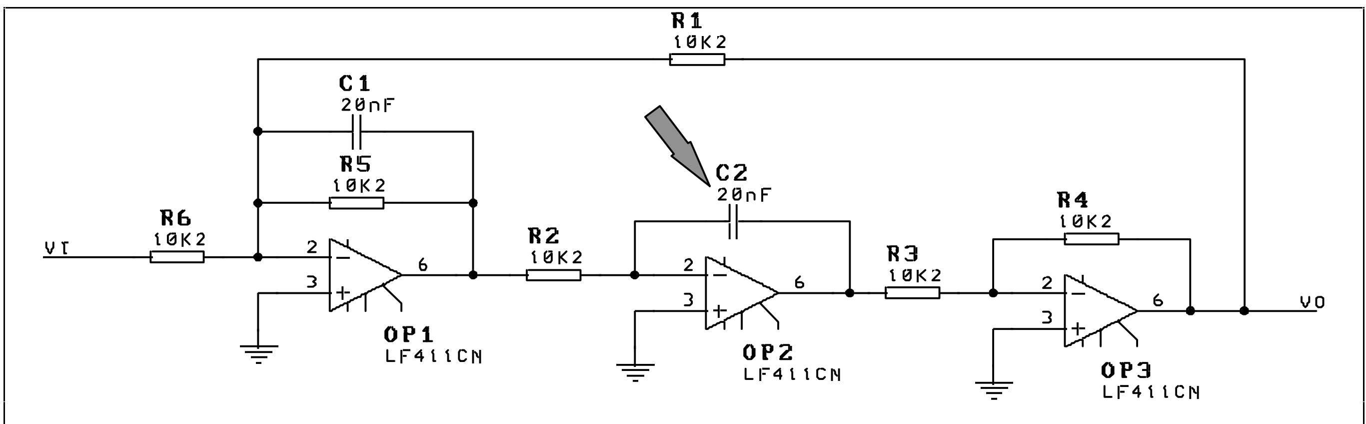


Figure 7. Error output of the integrator with a variation of 5% in C



R1,...,R6=10K2 C1,C2=20nF

Figure 8. Biquad schematic

All data presented used 30000 samples to evaluate the output error power ratio. It should be notice that a small 1% variation on R2 caused a relatively large (19%) variation in the test variable, that is, the output error power ratio. This shows why faulty plants are easily recognized.

3.3 Comparison with other approaches

In case of the Biquad all tests of the proposed methodology detected smaller component deviations when compared to the same example using other approaches. According to the results presented in [2], only variations greater than 28.9% in R2 or in C2 could be detected, forcing the comparison parameters out of its 5% tolerance border. In the proposed method, we were able to detect 1% variations in both components, with comparison parameter variations of 20%, well beyond the 5% tolerance border.

One should also mention that the test is inherently simple and fast: there is not the need to choose the correct set of input frequencies, neither to verify more than one variable like the gain or phase. Acquisition time was 6s, and processing time was 30 seconds inside the Matlab environment, in a 66MHz 486. With a dedicated program and a faster machine test time can be further reduced.

4 Real-life examples

In this section we present the robustness of the method by developing some real-life examples, including on-line testing of systems, the sensitivity of circuits to open-loop gain and bandwidth as well as fault detection at the presence of complex signals like voice.

4.1 Testing of a 4th-order filter

We applied the proposed testing method to switched capacitors circuits, using the MF10 chip as a test vehicle [12]. This chip is a programmable filter with a state-variable approach. Filter programming is done by the use of 4 external resistors. In this test, a 5% variation in one of the resistor caused a 80% variation in the error power ratio. A variation in the clock input frequency was also detected. When we moved the clock from 200KHz to 50KHz the output error ratio was 50.89 times higher than the normal case (4989% variation).

	R2	C2
-50%	73.33	74.08
+5%	4.47	4.11
+1%	1.19	1.21
short	126.00	72.06
open	72.06	125.40
no variation	0.9905	0.9905

Table 1. Output error power ratio

4.2 Testing a Biquad with non-ideal opamp

Although most works regarding the testing of analog circuits focus on the testing of small deviations of components, the opamp itself is not generally addressed. Let us take as an example a basic integrator composed of a resistor R and a capacitor C. If the differential gain (Ad) of the opamp is considered to be finite, than the transfer equation is

$$H(s) = \frac{1}{R \cdot C \cdot s + \dfrac{1 + R \cdot C \cdot s}{Ad}}$$

The above equation clearly shows that, in case the opamp has some malfunction that diminishes its open-loop gain, then the transfer function will reflect this property. In our methodology this is easily detected, for the gain shortage will mean a difference in feedback impedance, changing the plant transfer function. Table 2 shows the detection of opamp non ideal behavior, like limited gain or limited bandwidth.

opamp characteristic	value	error variation
open loop gain	1000v/v	35%
open loop gain	50v/v	818%
gain-bandwidth product	1MHz	50%
gain-bandwidth product	0.01MHz	159%

Table 2. Detection of opamp non ideal behavior

4.3 On-line testing

The presented test methodology can be also used on-line, that is, while the circuit is operating. In this case, a general DSP processor or a simple dedicated adaptive FIR integrated circuit must be used. The adaptive phase can be developed at power-on, or alternatively at any time during the operating life of the equipment, to take into account slow varying parameters like temperature.

Training of the digital filter is done in much the same way. Although the training is done with a frequency rich signal like white noise, one might wonder what will ever happen in case the signal of interest has itself many frequency components. That is, a signal with frequencies spread all over the spectrum could make the adaptive filter loose its selectivity against what is a fault or not. However, this is not the case. The linear adapter has shown itself robust, as shown in figure 9. The scope screen presents the operation of a filter trained with white noise. The first wave is a voice signal, the second the error signal of the training phase with white noise. The third wave presents the error signal at the presence of the voice signal. As the plant is working without faults, the relative power of the error is very close to 1. In figure 10 the same filter has a 50% variation in one of the capacitors. As it can be seen, the RMS error signal with some voice input is much larger.

5 Limitations of the approach

Although conceptually simple and easy to use, the methodology is based on some assumptions that must be fulfilled. The first is that an AD converter with enough resolution and speed is available. This, however, regarding specific niches, is readily achieved. For example, for audio applications, a simple acquisition board at extremely low costs connected to almost any personal computer can be used.

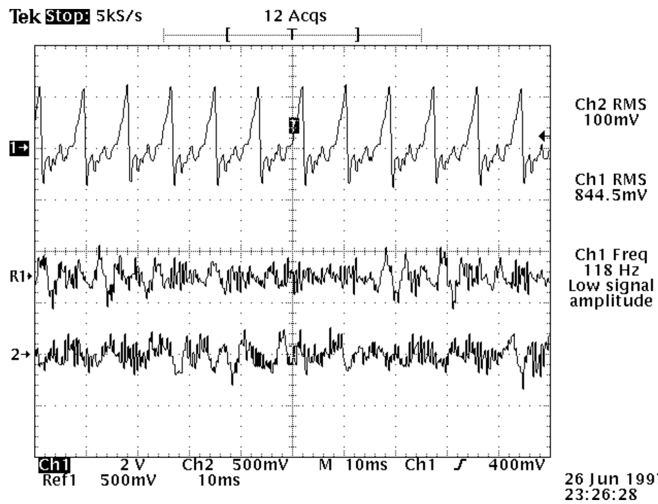


Figure 9. Good plant trained with noise

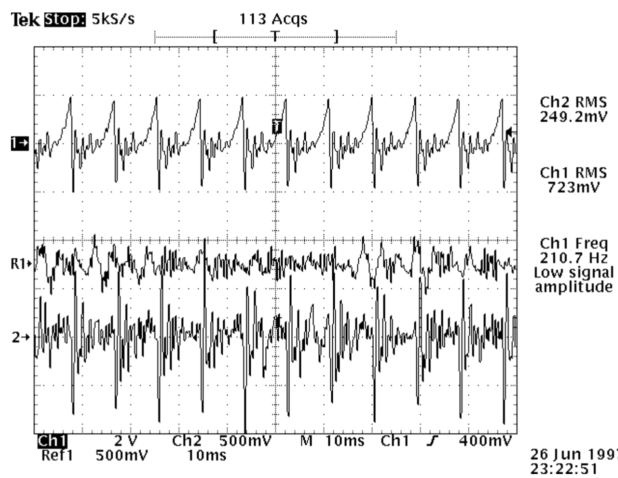


Figure 10. Error signal with voice input

The performance requests of the DA are somewhat relaxed. This is because since we need white noise, a large number of bits is not required, but the linear pass band must be higher than the one of the plant under test.

Testing time is quite fast, since although the filter might have a large number of coefficients to be settled, this is done only once every time a circuit is to be tested. Testing during production can be done with a number of excitation states which is proportional to the number of filter taps, guaranteeing full frequency sweep. For example, testing the Biquad took only 36 seconds in a 66MHz 486 PC inside the Matlab environment, and used part of a 6 seconds long data acquisition file.

Since adaptive algorithms are mathematically based on the Z transform, they can have a DC component. In our case, however, we still can not detect gain faults separately, since they could be mixed with some low frequency pole or zero variations.

This test is not able to identify the faulty component. This does not limit its range of use, since production testing is not intended to determine faulty components. The testing methodology can also serve as a high precision mechanism to evaluate the degree of similarity between mathematical models and prototype during design.

Finally, it should be mentioned that for circuits working at very small frequencies one would have to have a huge number of taps. To avoid this, the test engineer must use a smaller sampling frequency. Also, at high frequencies, one might have troubles with the acquisition system. In this case, the acquisition AD and the white noise DA converter must have enough bandwidth to excite all high frequency poles. If processing is not developed at real time, it can be done at any PC without problems. If on-line testing is required, power consumption of the adaptive filter must be taken into account.

6 Conclusions and future work

This paper has shown a new test strategy able to precisely detect soft, hard or catastrophic faults in an analog circuit. The method has low cost and high precision, and is straightforward to implement in any personal computer with a data acquisition board, or in a DSP board for on-line operation.

In comparison with published work, the method is quite robust, and is able to detect minor components variations without problems, up to less than 1%. Also, the error signal is 1 to 2 orders of magnitude greater than the traditional approach, being easier to detect. Since the methodology does not assume any set of particular frequencies, it can be integrated to any analog circuit described as a transfer function.

In our future work we intend to expand the methodology to include AD and DA converters, as well as DC signal components. Finally, when the original plant is known by the test engineer, the possibility of detecting the specific faulty component by a modification in the convergence will be investigated.

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