Frequency Specification Testing of Analog Filters Using Wavelet Transform of Dynamic Supply Current

Swarup Bhunia, Arijit Raychowdhury and Kaushk Roy
Department of Electrical and Computer Engineering
Purdue University, IN
{bhunias,araycho,kaushik}@ecn.purdue.edu

ABSTRACT

Wavelet transform has the property of resolving signal in both time and frequency unlike Fourier transform. In this work, we show that time-domain information obtained from wavelet analysis of supply current can be used to efficiently test the frequency specification of analog filters. The pole/zero locations in the frequency response of analog filters shift due to change in component values with process variations. It is essential to test the filters for the shift in frequency response and fix it, during production test. Wavelet analysis of supply current can be a promising alternative to test frequency specification of analog filters, since it needs only one ac stimulus and is virtually unaffected by transistor threshold variation. Simulation results on two test circuits demonstrate that we can estimate pole/zero shift with less than 3% error using only one measurement, which requires about 18 measurements in the conventional technique.

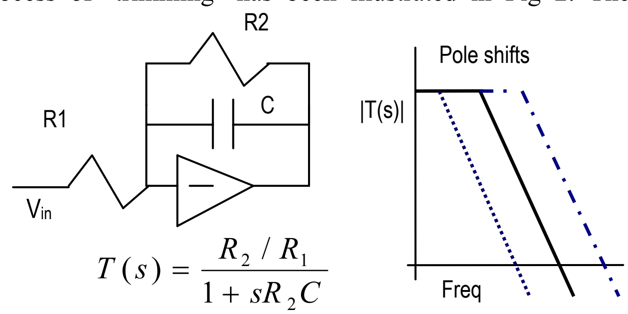
Index Terms: Wavelet Transform, Analog Filer, Trim Bit, Dynamic Supply Current (IDD).

1. INTRODUCTION

The steady growth of the digital circuit industry has made computation inexpensive and fast. However, all systems that involve interaction with the outside world demand data converters that can convert the analog signals of the outside world to binary digital signals for faster computation and also convert the processed digital data back to analog form for transmission. Thus, an analog front end is a prerequisite for all communication systems, medical instruments and signal processors, to name a few. This analog front end would ideally have an analog to digital converter (ADC) and a digital to analog converter (DAC). To add to this all the analog signals need to be amplified/ attenuated so that the ADCs and DACs can operate with full input and output voltage swings. Furthermore, analog signals need to be properly filtered in order to maximize the signal-to-noise ratio (SNR). Analog filters and amplifiers thus constitute an essential part of any system design.

Analog front-end filters provide a narrow pass-band to the signal and attenuate the unwanted noise in the stop-band. Based on the application, they can be low-pass, high-pass or band-pass, and can be of different topologies and filter orders [1-3]. In the state of the art VLSI design, all filter components are internal to the chip and hence R-C filters are the obvious choice. However, owing to process variation, the values of the resistors and the capacitors vary around their nominal values from one chip to another. Generally, the R-C values change systematically and hence, the ratio of one resistor (capacitor) to another on a particular chip remains unaltered due to process variations. As a result, although the gain of a filter (which is generally a ratio of resistors or capacitors) does not change, but the filter time constant (product of an R and a C) varies from one chip to another. This changes the location of poles (or, zeroes) of the filter and the overall frequency response changes. Fig 1 shows how the pole of a first order butterworth filter [1, 3] changes due to changing RC product.

However, it is essential to keep the frequency response and hence the time constants unaltered in any analog filter. This requires time constant 'trimming' during the production test. Conventionally, an input voltage of all frequencies is given to the filter under test and the output response measured. This output frequency response will be different form the desired frequency response because process variation would have changed the time constants of the filter. Depending on the pole (or, zero) location of the tested filter, a capacitor or resistor is 'trimmed' such that after trimming the time constant becomes equal to the desired time constant. The process of 'trimming' has been illustrated in Fig 2. The



$$T(s) = \frac{R_2 / R_1}{1 + sR_2C}$$

(a) (b)

Figure 1. Effect of process variation on analog filter parameters

capacitor C of the filter shown in Fig 1 has been provided with a bank of resistors. Once the changed time constant of the filter has been identified, the new value of C that would bring the time constant back to the desired value can be calculated. Let C' be the value of the capacitor C, that is needed to restore the time constant. The next step would be to realize C' from a capacitor bank network, as shown in figure 2. This is done by closing the appropriate switches in the capacitor network.

In this paper, we propose an efficient technique for detecting, prior to the time constant trimming step, the modified RC time constant of an analog filter due to systematic process variation. In contrast to the conventional technique, which requires an input voltage sweeping across all frequencies, here the input is a sinusoidal wave of a single frequency (namely the pole or zero frequency of the filter). The corresponding IDDT of the filter is measured. Since the phase response of any filter changes extremely fast at the pole or zero frequency, this phase change can be easily detected using wavelet transform of the IDDT. The phase information from the wavelet transform can be directly correlated to the amount of shift in the pole or zero location of the filter. Once this is known, it can be 'trimmed' as desired.

It can also be noted that the amplifiers that are used to build analog filters have bandwidths typically ten times higher than the filter bandwidth. Thus transistor process variations change the amplifier bandwidth but have no impact on the overall filter transfer function. Hence the phase response of the system remains unaffected, which makes the wavelet based method of time constant trimming robust to transistor process variations.

The biggest advantage of the method is that it requires an input of a single frequency, which reduces valuable testing time. Secondly, it is unaffected by transistor process variations and is robust and accurate. Further, the technique is 'non-invasive'. By this is meant that only the input current of the filter needs to be measured and no output voltage response is required. Thus contrary to conventional technique where extra pins need to be provided to measure the filter output response, this technique reduces the pin count.

The paper makes the following two contributions. First, it shows that information about the output frequency response of analog filters is contained in its supply current. For mixed signal circuits, where external connection to the filter's supply is more likely to be available than the filter's output, this provides a way for detecting time constant shift. Next, it presents a wavelet-based signal processing technique to extract the information about time constant efficiently. The proposed technique saves test cost and time, since it requires a single stimulus as opposed to set of stimuli over a range of frequency to be applied in the conventional method.

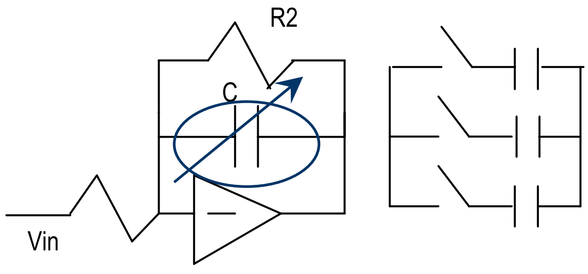


Figure 2. Trimming the time constant of an analog filter

The rest of the paper is organized as follows. Section 2 presents basic ideas about wavelet transform. Section 3 describes our technique for estimating RC shift. Section 4 presents the simulation results and deals with some issues associated with the trimming process and section 5 concludes the paper.

2. OVERVIEW OF WAVELET TRANSFORM

Fourier analysis has a serious drawback since it transforms signal in frequency domain losing all information on how the signal is spatially distributed. Wavelet transform of a signal, on the other hand, decomposes signal in both time and frequency domain [6-8], which turns out to be very useful in fault detection. In wavelet transform we take a real/complex valued continuous time function with two main properties - a) it will integrate to zero; b) it is square integrable. This function is called the mother wavelet or wavelet. Property (a) is suggestive of a function, which is oscillatory or has wavy appearance and thus in contrast to a sinusoidal function, it is a small wave or wavelet (fig 3(a)). Property (b) implies that most of the energy of the wave is confined to a finite interval.

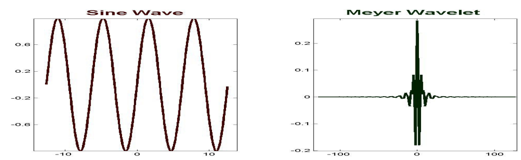
The CWT or the Continuous Wavelet Transform of a function *f(t)* with respect to a wavelet $\psi(t)$ is defined as:

$$W(a,b) = \int_{-\infty}^{\infty} f(t)\Psi_{a,b}(t)\text{, Where } \Psi_{a,b}(t) = \frac{1}{\sqrt{|a|}}\Psi\left(\frac{t-b}{a}\right)$$

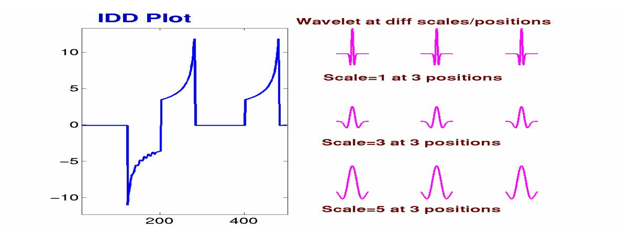
Here *a, b* are real and * indicates complex conjugate. *W(a,b)* is the transform coefficient of *f(t)* for given *a, b*. Thus the wavelet transform is a function of two variables. For a given *a*, $\psi_{a,b}(t)$ is a shift of $\psi_{a,0}(t)$ by an amount *b* along time axis. The variable *b* represents time shift or translation. Since *a* determines the amount of time-scaling or dilation, it is referred to as *scale* or dilation variable. If $a>1$, there is stretching of $\psi(t)$ along the time axis whereas if $0<a<1$ there is a contraction of $\psi(t)$ (fig 3(b)). Each wavelet coefficient *W(a,b)* is the measure of approximation of the input waveform in terms of the translated and dilated versions of the mother wavelet. Fig 3(a) compares the basis signals of DFT and wavelet transform. The mother wavelet shown in fig 3(a) is called *meyer* wavelet [9]. Fig 3(b) shows the translated and dilated mother wavelet used to approximate an IDD waveform of a test circuit.

3. PROPOSED PROCEDURE

In this section, we present details of the process of trimming analog filters to adjust shift in frequency response. As mentioned in section 2, frequency shift is caused by change in R and C component values, which cannot be precisely controlled during manufacturing. While change in RC time constant is related to the shift in pole-zero location in filter's frequency band diagram, its relationship with dynamic supply current (IDD) has not been studied before, to best of our knowledge.



(a)



(b)

Figure 3. a) Comparison of basis functions between Fourier and Wavelet b) Mapping an input signal with a translated and dilated basis function

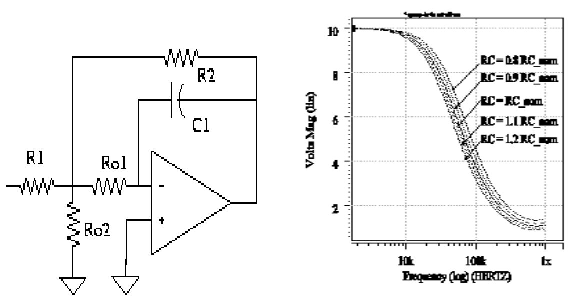
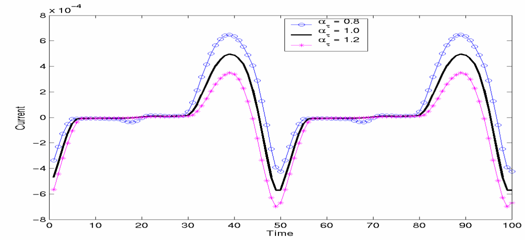


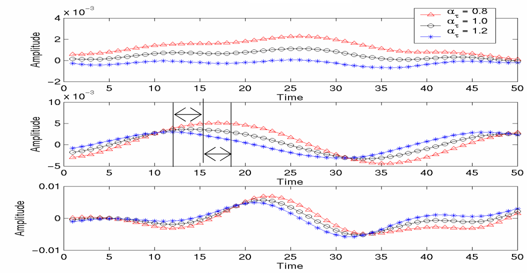
Figure 4. a) A test circuit b) frequency response plot for 5 different RC values

3.1 Relationship with Dynamic Current

We use the simple filter in fig 4 to demonstrate the relationship of pole-zero location with time-domain representation of supply current. The filter is a MOSFET-C filter [1], and has a single pole at 40KHz. Schematic of the filter and its frequency response is shown in fig 4(a) and 4(b) respectively. Frequency response is plotted for five different values of time constant – nominal value, +/-10% and +/-20% of nominal value. Fig 5(a) plots the dynamic supply current waveform for three different value of RC with response to an ac stimulus at a frequency equal to the desired pole frequency.



(a)



(b)

Figure 5. a) Dynamic supply current of the filter in figure 4 for 3 different time constants b) plot of wavelet transform coefficients for the corresponding currents

It can be observed that the current responses have fairly uniform shape, but they are translated in time axis. This can be clearly traced if we follow a particular peak point in the current waveform of nominal case and see how it moves in time for the other two cases. Furthermore, the direction at which the peak moves has direct correspondence with the direction of the RC shift. Thus, the peaks for two test circuits (one with +20% and -20% RC variation respectively) move in opposite direction with respect to the peak for nominal case. The shift in time axis is, at the same time, proportional to the change in RC value, which indicates that by observing the shift we can easily determine the direction and value of RC change.

3.2 Wavelet Analysis of Supply Current to Detect RC Shift

Since wavelet transform can resolve signals in both time and frequency domain simultaneously, it can be effectively used to localize points in time domain with sharp discontinuity in frequency. We use this property of wavelet to determine RC shift from supply current waveform. wavelet transform also helps to avoid *aliasing* problem [11] which may be present in pure time-domain analysis. First, we perform wavelet transform of the supply current at several scales. Then, we choose an appropriate scale for wavelet decomposition and choose a peak point in the plot of wavelet coefficients of the nominal current. In the next step, we measure the shift of this particular peak across test circuits.

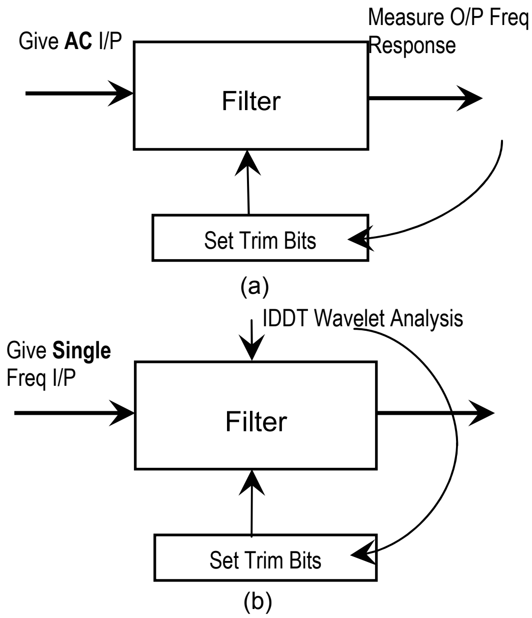


Figure 6. Comparison of test setup between conventional and proposed technique, a) conventional technique, b) IDD based technique

Fig 5(b) plots the wavelet coefficients of corresponding currents in fig 5(a), at three different scales with basis function *db2*. It can be observed that even though translation of the wavelet plot in time axis is reflected in all the scales, it can be more clearly computed for scale 2. It is also important to choose the right basis function, since closeness of approximating a signal with its wavelet components depends on the particular basis used. The process of selecting a basis wavelet and an appropriate scale can be done before the testing process based on simulation results of the filter circuit or measured waveform from a manufactured reference filter. The process of characterizing a filter circuit with a basis wavelet and a scale is simple process and needs to be performed once for each filter circuit. One advantage of wavelet transform is that the number of possible basis functions is unlimited and if for a particular circuit, we cannot find a reasonably good basis from the set of popularly known basis functions e.g. *db* family, *morl*, *mexhat*, *meyr* etc., we can compose a new basis function using wavelet toolbox.

3.3 New Setup for Trim Bit Setting

Fig 6 shows the new test setup for checking and fixing frequency specification of analog filter. Conventional method, shown in fig 6(a), requires a large number of test stimuli across a frequency range to be applied, and checking the frequency response to determine the trim bit values. On the other hand, test setup for the proposed method using IDD, shown in fig 6(b), requires only one ac input at the pole or zero frequency of the circuit, monitor dynamic current at the external supply pin and computing wavelet coefficients of the current. The coefficients are then compared with the nominal case to make decision about trim bit values. It can be noted that a filter may have multiple poles/zeros in frequency response, but a realistic filter design will have the poles/zeros at the same frequency. Hence, we need to apply only one stimulus at the input, which can substantially save the test cost and time.

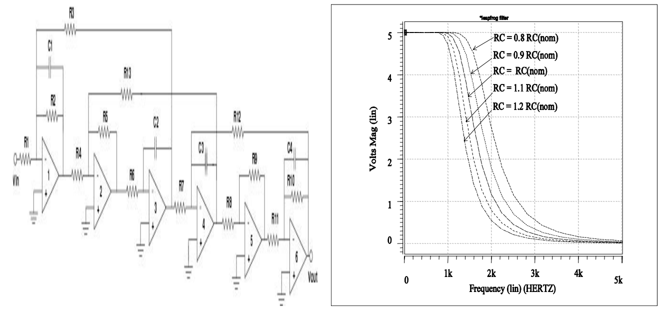
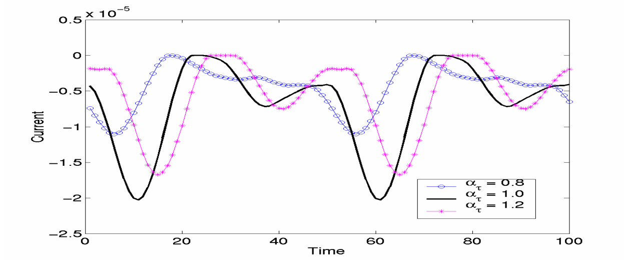


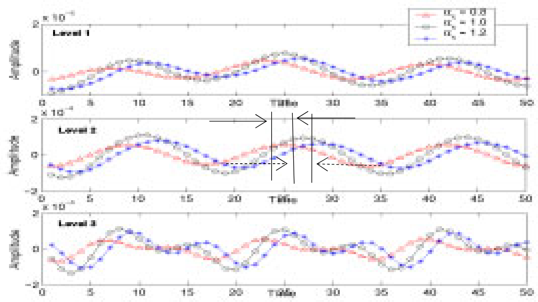
Figure 7. a) A leapfrog filter (ITC'97) b) frequency response plot of the filter for 3 different RC values

4. SIMULATION RESULTS

In this section, we present simulation results for two test circuits and discuss issues associated with accuracy of time constant estimation. One of the test circuits, referred as MOSFET-C filter is described in fig 4 along with its frequency response. We have also discussed the time constant estimation process for this filter with the proposed method. The other test circuit is a low-pass leapfrog filter with double zeros at frequency 1.4 KHz. It is taken form ITC'97 set of analog benchmarks [5] and has reasonably complex structure, as shown in fig 7(a). Fig 7(b) shows the plot of frequency response of the filter at the nominal value of time constant (bold line), +/-10% and +/-20% of the nominal value (dashed lines). The filters are modeled at TSMC 0.25µ technology node. All simulations are performed with *Hspice*. We have used matlab wavelet toolbox [9] for performing continuous wavelet transform (CWT) for multiple scales. The sampling frequency used for monitoring current waveform used was 50 MHz.



(a)



(b)

Figure 8. a) Dynamic supply current of the filter in figure 7 for 3 different time constants b) plot of wavelet transform coefficients for the corresponding currents

Similar to the process performed for the MOSFET-C filter, we applied test stimulus (ac signal at frequency of the zeros) to the leapfrog filter and obtained current response at three different RC values. Fig 8 (a) shows the plot of the current waveforms for the nominal case (no variation in RC due to process) and for +/-20% value of RC. The current waveforms were then subjected to wavelet decomposition. Fig 8(b) shows the plot of wavelet coefficients for the current waveforms in fig 8(a) at three different scales for basis function *db2*. It can be observed that RC time constant shift is clearly reflected in the plot of wavelet coefficients for several scales.

Once the translation in time domain is computed from wavelet coefficients, we convert it to an estimate of actual RC shift from a simple table. The table needs to be generated out of a simple calibration process during test design by observing the translation of wavelet coefficients with different values of RC. Fig 9 plots the estimation error with different RC shifts for the both filters. The maximum error in both cases is 3%, which proves the effectiveness of the method.

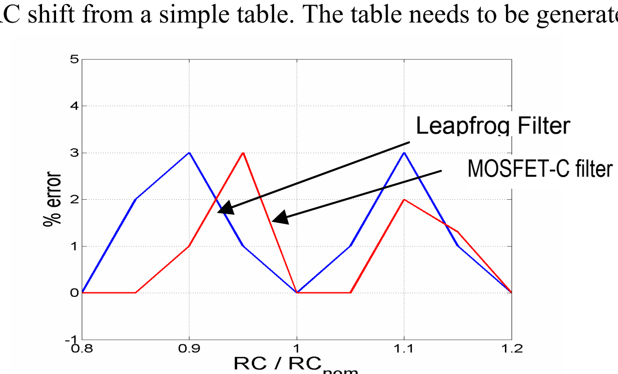


Figure 9. Accuracy of estimation of RC shift with IDD based technique

We can present a simple estimate of test cost saving in terms of number of stimuli required. Let x be the nominal RC constant for the filter. Hence, the nominal pole/zero frequency is at $1/x$. With +/-25% process variation, pole/zero position shifts from $1/0.75x$ to $1/1.25x$. So, $\Delta f = (1/0.75x - 1/1.25x) = 8/15x$. If we trim to an accuracy of $a\%$ then, number of required measurements at different frequencies is: $\Delta f/a = (8/15a)*100$. As an example, for accuracy of 3%, we need to take about 18 measurements in conventional technique, while we need to make only one measurement with the proposed method.

4.1 Impact of Transistor Threshold Variation

It is very likely that transistor parameters (length, width, oxide thickness etc.) also vary with process along with R and C components. The circuit diagram of the first stage of the OPAMP that has been used in all the simulations has been given in fig. 10. Variation of the transistor parameters in the OPAMP can be modeled reasonably well as transistor threshold (V_{th}) variation [10]. We performed simulations on both the test circuits to observe impact of transistor process variations on the RC estimation. Fig 11 and 12 plot IDD waveforms (11(a) and 12(a)) along with the corresponding wavelet coefficients (11(b) and 12(b)) for three different V_{th} values (nominal and +/-10% over nominal) for both the filter circuits. It can be noted that wavelet coefficients varies only in amplitude and not in time across V_{th} variations. Thus, RC estimation from supply current is not affected by variation of transistor parameters with process.

4.2 Miscellaneous Issues Affecting Estimation Accuracy

Power distribution network of an analog circuit can be

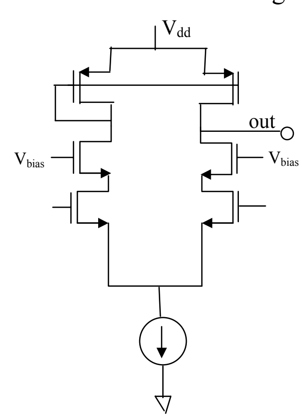
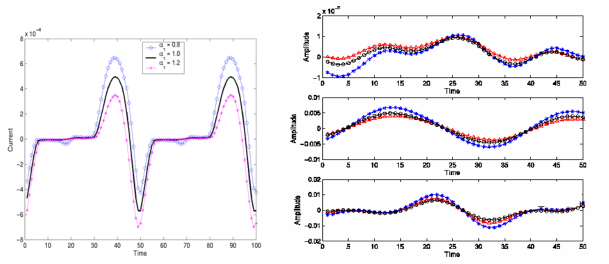


Figure 10. First stage of the OPAMP used in our simulations. The second stage is a conventional push-pull amplifier.



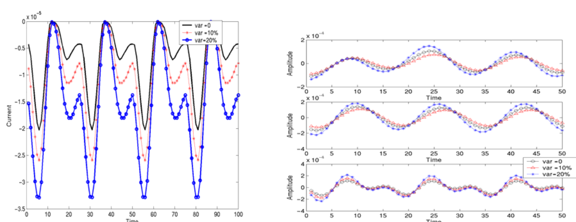
(a) (b)

Figure 11. a) Dynamic supply current of the filter in figure 4 at 3 different transistor threshold values b) plot of wavelet transform coefficients for the corresponding currents

modeled with distributed RLC components. Depending on the values of the RLC components and the network topology, any external current measurement hardware will experience some attenuation in the amplitude and frequency of the supply current. Since analog circuits generally consist of a small number of transistors, analog power-grids are very small in size and are expected to cause negligible attenuation in the measured current. For mixed signal circuits analog parts have separate power grid [3], which makes the external measurement of current easier.

Measurement hardware can also have impact on the IDD response. In most cases, external current sensor acts as a low-pass filter and eliminates the high frequency components. Since we sample the current waveform at a very low frequency (50 MHz) and then perform wavelet analysis at lower frequency range (dictated by the choice of scale), it is very unlikely that estimation error will significantly increase due to error in measurement hardware.

Impact of supply grid and measurement hardware, however, can be reduced by taking the nominal case from a measured response instead of simulated behavior. At the onset of trimming process, we need to pick up a nominal or reference filter by measuring its frequency response with the conventional method. The current waveform measured with same hardware for the test circuit will experience similar distortion as the nominal current and delta change in the time-domain information of current is likely to have little impact.



(a) (b)

Figure 12. a) Dynamic supply current of the filter in figure 7 at 3 different transistor threshold values b) plot of wavelet transform coefficients for the corresponding currents

For mixed signal circuits, where analog filters are used in the front and/or back end of the digital parts [2], external pin for analog filter outputs may not be available and hence, it is not always possible to apply conventional trimming technique for filters. Proposed method can be very effective in these cases since analog power-grid is generally connected to separate external supply pin, making measurement of supply current very feasible. If the analog filter is part of a bigger analog circuit in an IC, we can use simple gating logic at supply or proper input vector selection method to turn off certain sections in the circuit from supply line.

5. CONCLUSIONS

In this paper, we demonstrate that pole/zero locations in the frequency response of analog filters can be determined from analyzing dynamic supply current with acceptable precision. The technique can be used to observe and compensate pole/zero shift in analog filters arising from change in component values with process fluctuations. It has the advantage of requiring just one ac stimulus compared to a large number of inputs necessary in conventional pole/zero determination process. Coupled with defect oriented supply current testing using wavelet transform, which we have shown in our earlier work [4], proposed method can provide a complete test environment for analog filters.

Acknowledgement: This research was supported by the MARCO under Gigascale Silicon Research Center (GSRC).

REFERENCES

1. I. A. Johns, Ken Martin, *Analog Integrated Circuit design*, Wiley International Edition, 1996
2. Mark Burns and Gordon W. Roberts, *An Introduction to Mixed-Signal IC Test and Measurement*, Oxford University Press, 2001.
3. A. V. Oppenheim and R. W. Schafer, *Digital Signal Processing*, Chapter 5, Prentice Hall, 1975.
4. S. Bhunia and K. Roy, *Dynamic supply current testing of analog circuits using wavelet transform*, VLSI Test Symposium, 2002, pp. 302-307.
5. B. Kaminska, K. Arabi, I. Bell, P. Goteti, J. L. Huertas, B. Kim, A. Rueda and M. Soma, *Analog and mixed-signal benchmark circuits - first release*, International Test Conference, 1997, pp. 183-190.
6. I. Daubechies, *Ten Lectures on Wavelets*, CBMS-NSF Regional Conference Series in Applied Mathematics, SIAM, 1992.
7. R. Rao and A. Bopardikar, *Wavelet Transforms: Introduction to Theory and Applications*, Addison-Wesley, 1998.
8. C. Burrus, R. Gopinath, H. Guo, *Introduction to Wavelets and Wavelet Transforms*, Prentice Hall, 1998.
9. *Matlab Wavelet Toolbox*, version 2.0, **http://www.mathworks.com**.
10. P. Yang, E. Hocevar, P. Cox, C. Machala, and P.Chatterjee. A*n integrated and efficient approach for MOS VLSI statistical circuit design*, ieee transaction on cad, 5(1): 5-14, 1986.
11. K. Muhammad and K. Roy. *Fault detection and location using idd waveform analysis. ieee design and test of computers*, 18(1):42{49,) Jan –Feb 2001