A NON LINEAR FILTER; THE DESIGN OF AN ANALOG OTA CURRENT MODE MEDIAN FILTER.

ARTURO PRIETO FUENLABRADA.*, BORIS ESCALANTE RAMÍREZ, JOEL GARCIA DELGADO*.
División de Estudios de Posgrado de la Facultad de Ingeniería,
Universidad Nacional Autónoma de México.
*Facultad de Ciencias de la Electrónica,
Benemérita Universidad Autónoma de Puebla.
MÉXICO.

Abstract: - This article presents a non linear filter design, an Analog Median Filter design based on transconductance operational amplifiers OTA's and Current elements. This implementation corresponds to an ideal macromodel based on VCCS elements and current mirrors. The analysis, control variables and limitations are presented, a brief comparison between other reported filters is also made.

Key-Words: - Non linear filter, analog Cmos median filter, current mode filter.

1 Introduction

In many signal processing applications the suppression of unwanted components can be achieved by linear filters, but in signals with sharp edges corrupted by noise, linear filters also smoothes out signal edges (information) and in addition, impulsive noise cannot be sufficiently suppressed. Another type of filters, non linear or adaptive were proposed, filters that preserves edges while suppressing impulsive noise. The non linear Median Filter has had a good performance in such cases, it replaces the input signal value at each point by the median of the signal value in a neighborhood around that point [1,2,3]. These filters are used in Speech processing, video signals and other applications like Neural Networks [4]. Some digitally [3,5,6] and analog implementations [4,6,7,8] are reported.

2 Problem Formulation

Lee and Kassam, based on a Maximun Likelihood Estimators theory, proposed an algorithm to get the median [2]. It is stated as : The output yk of the median filter M is defined as the solution of the equation :

$$\sum_{k=1}^{N} \underset{\lim e \to 0}{f\left(x_k - M\right)} = 0 \qquad (1)$$

If f(x) is a linear function f(x)=ax, the media is obtained. If f(x) is no linear approaching the hard limiter as shown in Fig. 1, the median is obtained.

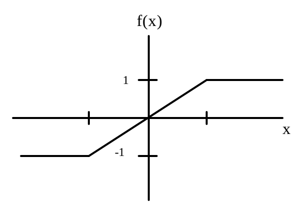


Fig.1. Function to get the analog median filter.

Based in this algorithm, this article proposes an analog system to solve this equation. A novel conceptual behavioral macromodel is presented and a real implementation emanated from this model is presented using CMOS technology, with Vdd=2.5V, Vss=-2.5V, λ=1.2μ. , results are simulated in SPICE .

2 Problem solution

A novel macromodel to solve the Lee and Kassam algorithm is shown in the next figure (fig.2), it should be noted that this diagram corresponds to the conceptual implementation in OTA current mode.

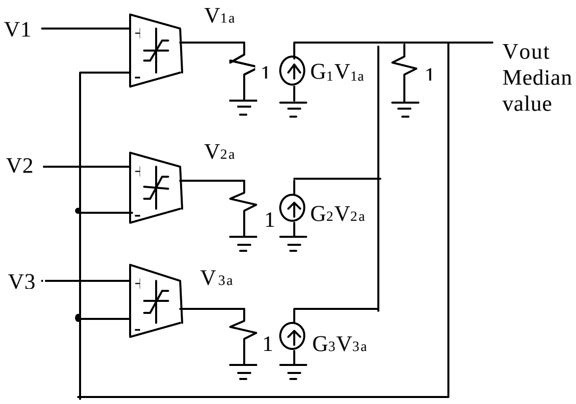


Fig. 2. Block Diagram to implement the median value of three inputs.

The block to make the hard limiter function was replaced by a linear amplifier followed by a clipping section as shown in fig.3. It should be mentioned that the three parameters can be adjusted, the gain of the amplifier; the higher the best and in the case is infinite the function becomes a sign type function, and the limiting voltages V1 and V2.

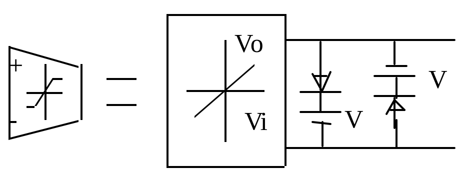
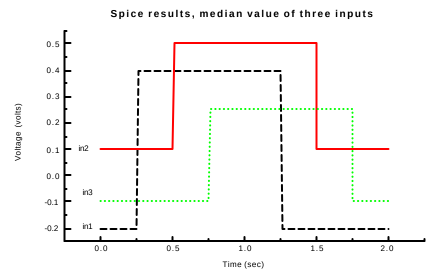
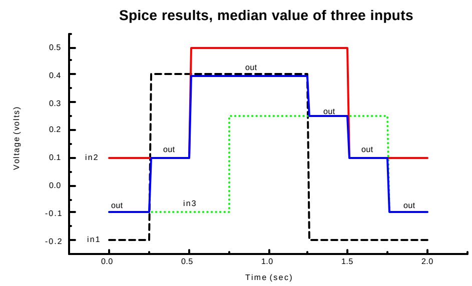


Fig.3 Hard limiter function.

With this macromodel some different SPICE simulations show the way this algorithm works. The ideal simulated results are shown in fig.4, where it can be seen how the output follows the median of the three input values, in a) the three inputs are shown and in b) the median superimposes the signals. For example at interval 0.0 secs. to 0.25 secs., input1 has a value of -.2V, input2 has a value of 0.1 V and input3 has a value of -0.1V, the median of these three values is -0.1V and superimposes input3 in that interval as shown in Fig 4.b.



a) three input signals.



b)median output of three signals.

Fig.4 . Simulated results of Lee and Kassam algorithm with ideal blocks. a) inputs b) median output.

Some important points should be noted, it is a non linear system, it gets the median value from the three input values if the limiting function is hard enough and in the limit it implements the sign function. With gains of the amplifiers (in the first OTA or in the VCCS element) greater than 100 the median value is obtained. Higher gains make no significant differences in the median value. Some other functions like the soft limiting one has exactly the same results as the hard limiting ones except that it smoothes the sharp transitions. When the gains are lower the output value tends to be the Media instead of the Median.

This ideal macromodel was the basis for the design implementation, it is advisable for other applications to handle this macromodel and its variables prior to the design steps. An iterative process follows between this macromodel and real design parameters. An example would be the finite gain of each OTA putting a real limit to the hard limiter function and as pointed above the media instead of the median is obtained; parasitic capacitances in order to check the frequency response; the values of the VCCS elements and load resistances; it should be advisable to add these constraints to the macromodel and make tradeoffs between them to get optimum results.

This block is the basis of an analog median filter, the value of the signal is substituted by the median value in a neighborhood, so the input signal is delayed (Z^{-1}) and compared to get the median as shown in Fig.5.

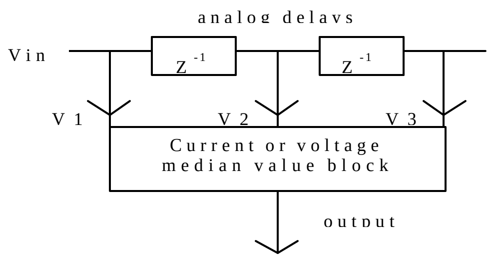


Fig .5. Block diagram of an Analog Median Filter.

4 RESULTS

A novel OTA Current Mode analog filter Implementation based on this macromodel was designed using a CMOS technology of 1.2μ.

The novel OTA current mode implementation of the block to get the median value between three input signals is shown in figure 6, where a value of Vdd=2.5 V, Vss=-2.5 Volts, where used.

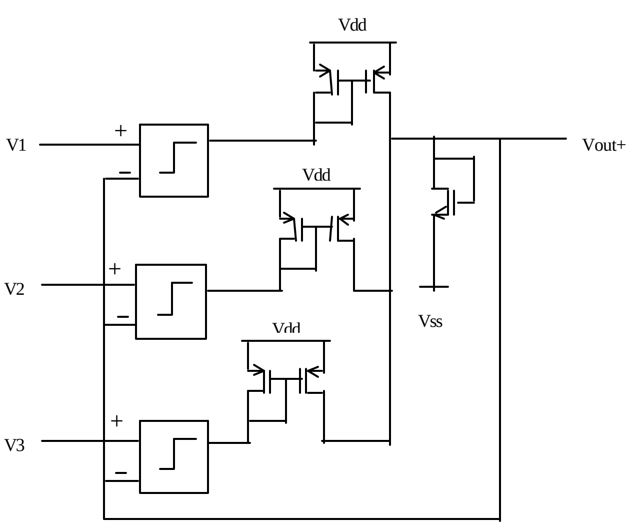
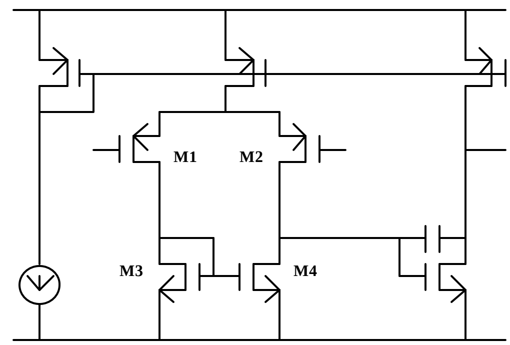
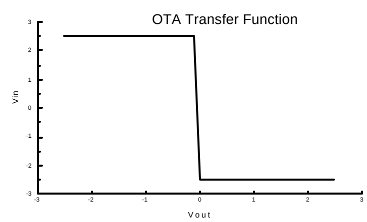


Figure 6. The circuit to get the median of three values..

As a hard limiter block the well known CMOS Miller OTA [9] was used with an IB= 2.5 μA, it is shown in Fig.7.a, together with the simulated Spice results of the limiting function in Fig.7b.



a) Circuit of OTA.



b) Transfer function.

Fig.7 a) CMOS Miller OTA, b) Simulated results showing the limiting function.

To delay the signal, an all pass filter [9] like the one shown in Fig.8 was realized, its transfer function is :

$$H(s) = -\frac{s - \frac{Gm_1}{C}}{s + \frac{Gm_2}{C}} \qquad (2)$$

Where Gm1, and Gm2 are the transconductances of the OTA´s, they are similar to the one of Fig.7.

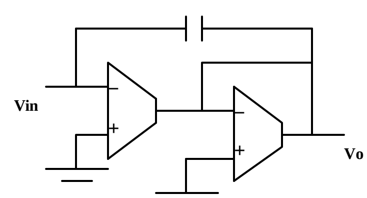
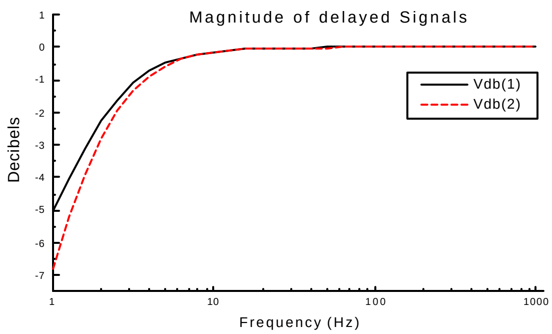
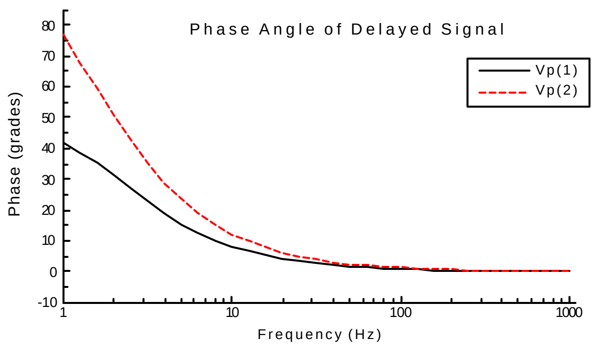


Fig. 8 .OTA based all pass filter.

Two of these all pass filters where connected in series, in order to obtain the signal and two delay versions of itself, the simulated magnitude and phase Spice results are shown in Fig.9a and Fig.9b. The transient response of a sinusoidal input and delayed twice is shown in Fig.10.



a) Magnitude



b) Phase

Fig.9 All pass filter simulated results a) Magnitude and b) Phase characteristics.

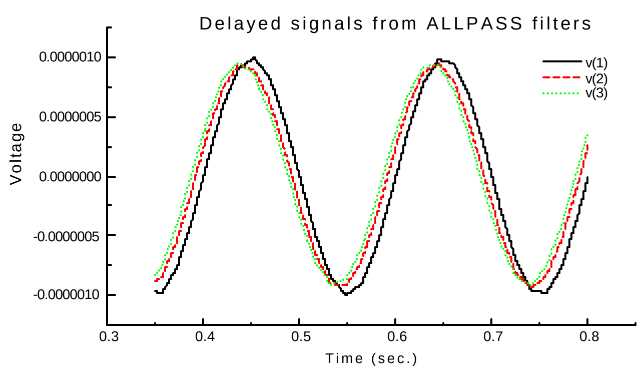
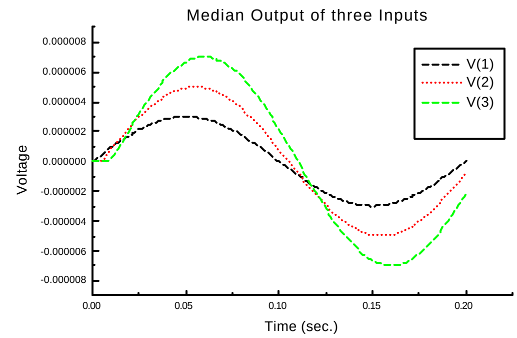
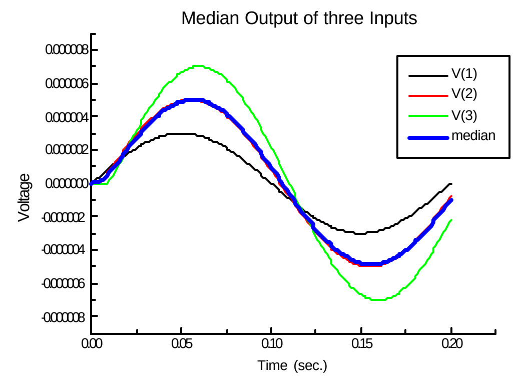


Fig 10 . Delayed signals from a two all pass sections.

The results of the circuit shown in Fig.8 without the delays, to get the median output of three signals are shown in Fig. 11.



a)three input signals



b) median output of three signals.

Fig. 11 . The median of three inputs, results from circuit of fig.8.

The total response of the proposed Analog Median Filter with a sinusoidal input contaminated with an impulsive noise is shown in Fig.12, it should be noted how the noise is almost suppressed, with signals with sharper transitions no such good results are obtained ,because of capacitive matchings, non ideal frequency responses, high DC impedance nodes, non linear effects etc. Some authors recommend linears filters together with Median or other non linear filters to achieve the best results. The control variables from the model and its real equivalents variables in the designed Median filter like , OTA gains, gain of current mirrors, delay of signals, input range etc. can be adjusted in each stage. Tradeoffs between control variables should be made in each application to obtain the best results. A comparison between different results from the literature are shown next.

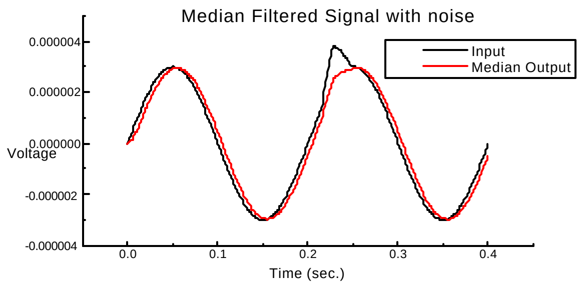


Fig. 12. Output of median filter corrupted with noise

The digital Median Filters get the median , although in general they need a lot of hardware to implement different algorithms , they need memory and an ALU, and of course A/D and D/A converters. Three different ways to get the median are discussed in reference [3], another with a recursive algorithm in reference [5].

The analog ones, in general do not make the difference between the mean and median value, some get the median [6,8] using a sign type function [8], with a cascode (gain 70Dbs) and a comparator [6]. In [7 they get the mean and only in the limits the median. Most of the articles show how to get the median between three values , but the filter with the delay sections and its limits are not shown.

5 CONCLUSIONS

A non linear filter was designed, an analog median filter with Gm current mode elements based on Lee and Kassam's algorithm. An ideal macromodel simulates the algorithm. The design and analysis of its different parts was made. The prototype was implemented using CMOS technology of 1.2µ , power supplies of ±2.5 V and Ib=2.5 µA. Good results in the presence of impulsive noise are shown.. From the model the control variables were determined and tradeoffs between them have to be made to optimize these non linear filters. Simulated results from Spice are presented.

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