Digital Implementation of a Tail Cancellation Filter for the Time Projection Chamber of the ALICE Experiment.

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Abstract

In the ALICE TPC (Time Projection Chamber), the readout chambers are conventional proportional chambers with cathode pad readout. The pad signal has a rather complex shape, which depends on the details of the chamber and the pad geometry, characterised by a long tail due to the motion of the positive ions. Since the zero-suppression has to be done before the data transfer, the high channel occupancy calls for a very precise tail cancellation. In order to be compatible with the required *dE/dx* resolution, a suppression to 0.1% or better of the maximum pulse height, is required. We present a digital implementation of a shortening digital filter based on the approximation of the tail by the sum of exponential functions. The hardware implementation of the filter is described and the results analysed.

I. Introduction

The ALICE TPC [1], [5] is a gas detector, of cylindrical shape, with a sensitive gas volume of about 80 m^3 subdivided into drift spaces by a central high voltage plane. A charged particle passing through the gas volume creates electrons by ionisation. The electrons drifting in the electrical field towards the readout chamber, where they are amplified in the field of the sense wires. The signal is coupled to the pads (570 000), which are at few mm distance behind the sense wire-plane. The charge collected on each pad, integrated and subsequently shaped, is sampled at 5.66MHz over the 88μs drift time.

A single readout channel is comprised of three basic units: a charge sensitive preamplifier/shaper (PASA); a 10-bit 5.66 MHz low-power ADC; and an ASIC that contains a shortening digital for the tail cancellation, the baseline subtraction and the zero-suppression circuits [2]. The large granularity of the TPC (2.85×10^8 samples), leads to event sizes of 356Mbyte/event and a data volume in the front-end electronics which is far beyond of present data-handling techniques. To achieve the necessary rate capability – about 200 events/s written to permanent storage – data has to be pre-processed directly in the front-end before it is transferred to the DAQ system. One way to compress the data stream is to discard “zero” data. According to Montecarlo simulations the zero-suppression in the front-end will reduce the data throughput by a factor of 4. However, owing to the high channel occupancy (up to 40%), the traditional zero-suppression algorithms are very inefficient due to pile-up effects. The minimisation of such effects requires a very precise tail cancellation, 1‰ or better of maximum pulse height, which implies the design of a very accurate digital filter.

A technique to cancel the tail was developed for a proportional chamber and implemented in a analogue circuit based on discrete components proposed in [3]. The same technique underlies the tail cancellation scheme used in the preamplifier/shaper circuit developed for the readout of the CERES TPC [4]. The accuracy of a shortening filter realised by discrete analogue network is limited by the tolerance of the components. Owing to the poor precision in the matching of the passive elements, provided by the current integrated circuit technologies, an analogue filter would not reach the 1‰ accuracy if it has to be built in an integrated circuit with no external tuneable components. On the other hand, individual tuning would not be feasible considering the number of circuits to be implemented. A digital circuit allows better control of the requirements by choosing the word length and type of arithmetic. Flexibility is also given by the possibility of reconfiguring the digital signal processing operations by changing programmable coefficients. With a digital circuit, we can cope with different signal shapes due to the chamber geometry and gas mixtures

In section II of this paper, the basic idea of approximating the tail by the sum of three exponential functions will be addressed. Section III describes the VLSI implementation of the circuit performing the digital filtering operations and the comparison of several structures according to area and error versus word length. The results of such implementation for measured signals using two different amplifiers are given on section IV. Section V concludes the paper.

II. Basic Concept

The tail of the signal received from the TPC analogue readout, *is(t)*, can be approached by the sum of *n* exponential functions:

$$is(t) = I_0 \times \sum_{i=1}^{n} A_i \times e^{-\frac{t}{a \cdot t_i}} + r(t) \qquad t_1 << t_2 << ... << t_n \quad \sum A_i = 1$$

Where r*(t)* is a residual function due to the error of the approximation. The sum of the gains A_i should be one so that input and output have the same height. The input signal can be expressed in the *Z* domain as:

$$Is(z) = I_0 . \sum_{i=1}^{n} \frac{A_i}{1 - \exp(T / a\, t_i).z^{-1}} + R(z)$$

The signal is passed through a linear digital network that cancels all but the fastest of the exponential terms. The *n-1* pole-zero network has a transfer function that expressed in the *Z* domain is:

$$F(z) = \frac{(1 - \exp(T / a\, t_2 . z^{-1}))..(1 - \exp(T / a\, t_{n-1} . z^{-1}))}{(1 - L_1 . z^{-1} + L_2 . z^{-2} + .. + L_{n-1} . z^{n-1})}$$

The numerator of *F(z)* will perfectly cancel all the poles of *Is(z)* except one. The constants L1, L2,..,Ln-1 are chosen such that the numerator of the expanded form of *Is(z)* disappears. The response of this linear network to the incoming signal is the convolution in the time of the impulse response function of the filter and the signal itself:

$$is(t) * f(t) = I_0 \exp(-t / a t_0) + r(t) * f(t)$$

One can easily observe from this expression that the performance of the tail cancellation is strongly related to r*(t)*. The remaining fast exponential is a constraint of the system and can be chosen such that:

$$\exp(-t / a t_0) < 0.1\% \qquad t \geq 1ms$$

The Tail Cancellation scheme can be shown in a block diagram:

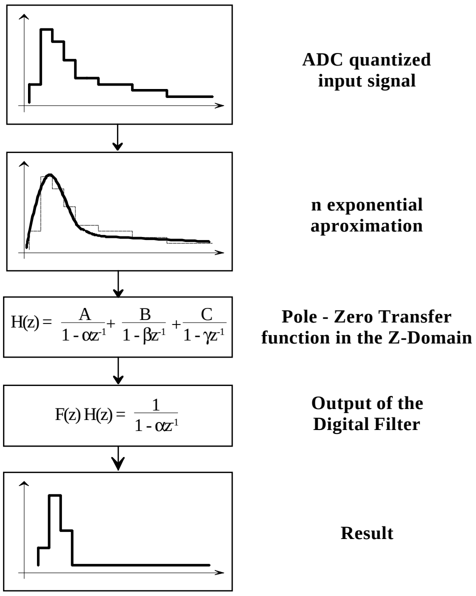

Figure1: The Tail Cancellation scheme

The digital signal from the ADC is approached by *n* exponential functions. From the coefficients (gains and exponents), one can derive the *Z* transform, *H(z)*, impulse response function of the input signal. The coefficients of the filter *F(z)* are chosen so that the product of *F(z)* and *H(z)* is the *Z* transform of the fastest exponential. It should be noted that as the coefficients of the filter depend on the approximating exponential functions (gain and exponent), some sets of parameters generate unstable filters. It is common on the cases analysed that the filter poles tend to be quite close to the unit circle and stability problems occur if care is not taken when coding the coefficients [7].

III. HARDWARE IMPLEMENTATION

In case of the ALICE TPC signal shape, approximations with 3 and 4 exponential functions were considered, generating IIR filters of order 2 and 3 respectively. There are numerous architectures for implementing such filters [6], [8]. In this section three different structures were analysed and compared in terms of area for a given word length, and error as function of the number of bits. Both cases use fixed-point arithmetic. Figure 2 shows the three different structures analysed. Filter A is a Direct Form realisation, Filter B is a Lattice-Ladder structure and Filter C two first order filters cascaded. These filters were synthesised using a 0.35-micron CMOS process. The comparison in terms of area is shown is table 1 for 18-bit filters. Figure 3 shows the Error as function of the number of bits for the three structures.

Table 1: Silicon area for Filter A, B and C for 18-bit

	Filter A	Filter B	Filter C
Area (mm^2)	1.024	1.901	1.068

From these values, one concludes that Filter C reaches the highest accuracy for a given number of bits. Its area is not considerably higher than Filter A. When implementing Filter C, one has access to the poles and zeros directly via the registers

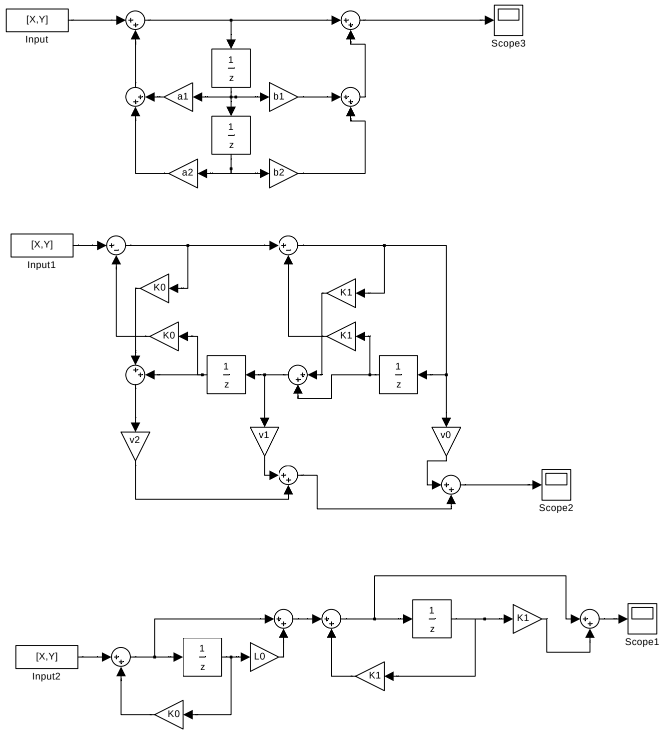

Figure 2: Filter A (top), B (middle) and C (bottom)

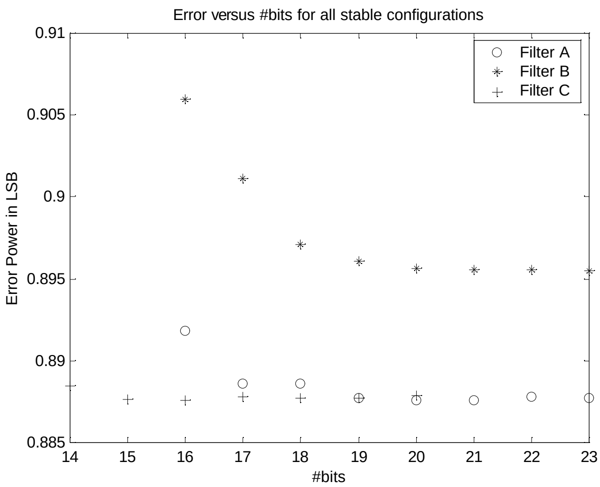

Figure3: Sum of the squared errors expressed in LSB as function of the number of bits.

IV. RESULTS WITH MEASURED SIGNALS

Measurements were performed using a TPC prototype with the same cathode plane layout as the ALICE TPC, operated with ArCH4 95% - CO2 5%. Two different analogue amplifiers were used: the NA35/ALEPH and the NA49-FTPC Preamplifier/Shaping Amplifier [9]. The two amplifiers produce signals with different shapes therefor have to be treated using different approximations.

A. *NA35/ALEPH Preamplifier/Shaping Amplifier*

The input of the Digital Filter is shown in figure 4. We can see that this amplifier induces an initial undershoot before the beginning of the tail itself at 1 microsecond. Figure 5 and figure 6 show respectively the approximation with 3 and 4 exponential functions and the error committed by each one of these fittings. Clearly the 4 exp. approximation fits best the tail. The coefficients (exponent and gain) for each of the exp. were computed in such a way that before 1 microsecond the approximation is relatively rough, but quite accurate afterwards. Such weighted least-mean square approximation scheme allows the error to be within 1‰ after 1μs. The approximation can be decomposed in the elementary exponential function. As an example, figure 7 shows the case for the 4 exponential approximation.

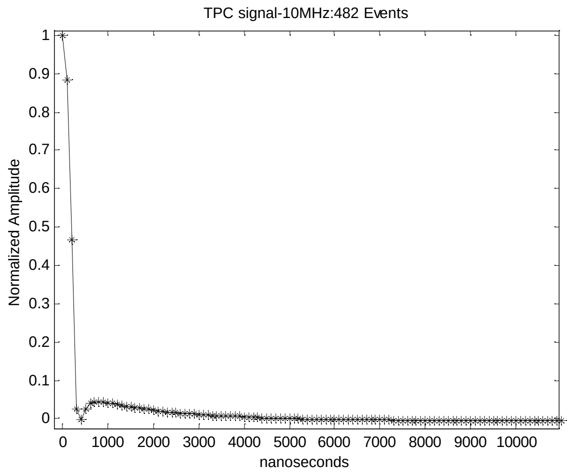

Figure 4: The filter input signal (from NA35/ALEPH preamp/shaper)

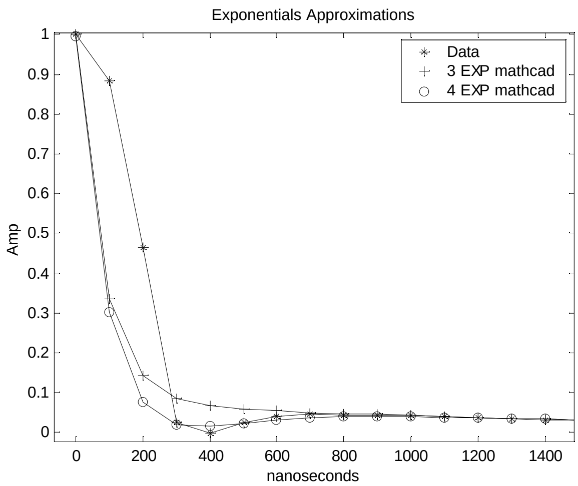

Figure 5: The Exponential Approximations

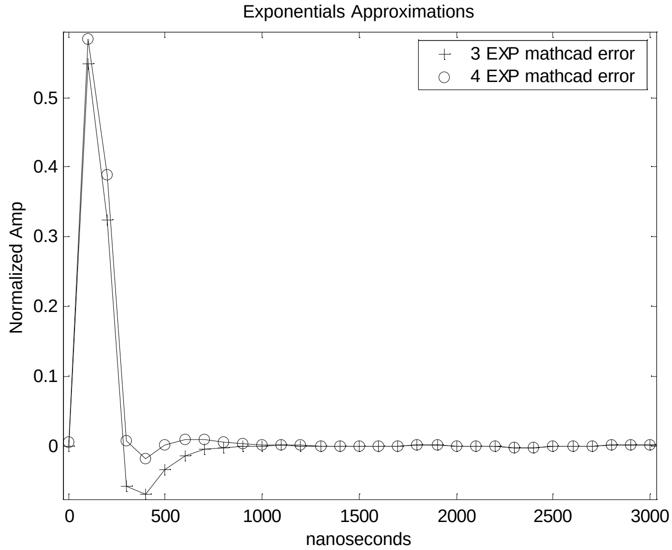

Figure 6: Error for 3 and 4 exponential functions

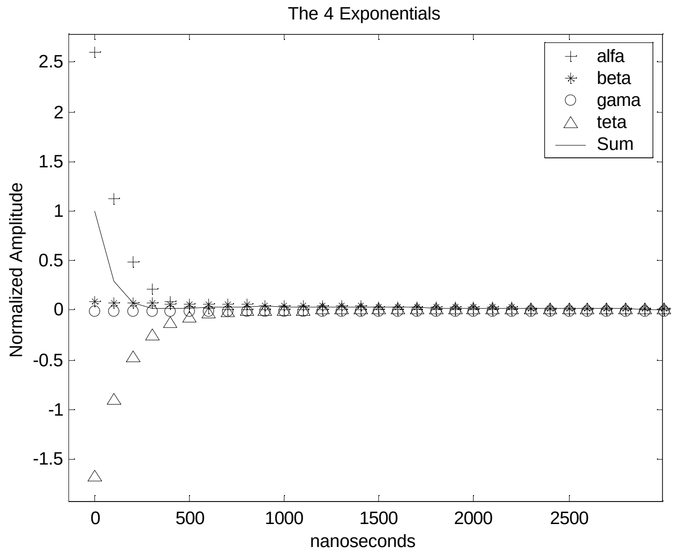

Figure 7: The 4 exponential functions and their sum.

Two of the exponential functions have negative gain and one has gain superior to one. This configuration is required in order to best fit the initial undershoot. Performances for both filters, one of second order and one of third order (3EXP and 4EXP) are shown in table2 and figure8.

Table 2: Comparison between the input and the output of a 2nd order and 3rd order filters.

	Input	3EXP	4EXP
Time to 0.1% (us)	40	2	3
Max. Undershoot (%)	0.5	9.5	4.5

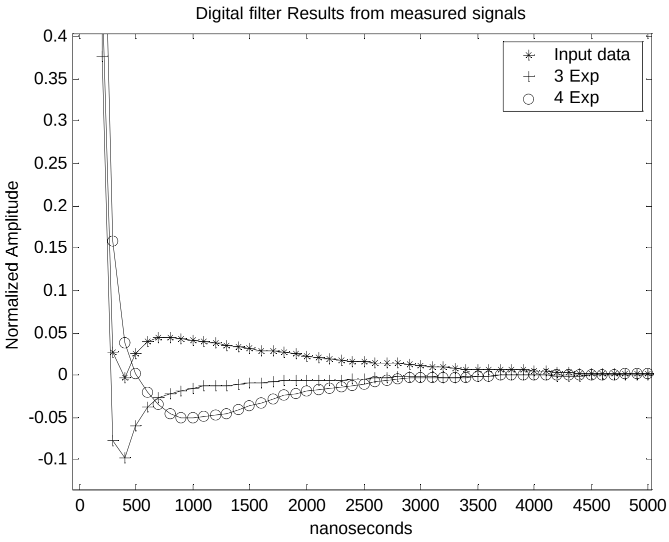

Figure 8: Input, output of the 2nd and 3rd order filter.

B. *NA49/FTPC Preamplifier/Shaping Amplifier*

Figure 9 shows the input signal, figure 11 and figure 10 present the error of the approximation and the approximation itself. Figure 12 and table 3 show the output of both filters. This amplifier is bipolar and a slight mismatch of poles and zeros on the analogue circuit creates an undershoot. In this case the tail tends towards zero from the negative values. For this reason the gain of two of the 4 exp. should be negative. The principle exposed in section II is general enough and the tail cancellation is achieved after 1.2 microseconds. We can observe that the 4 exp. approximation is more accurate before 1 microsecond and as good as the 3 Exp. afterwards. In order to test the proposed filter Filter under more realistic conditions, 20 different measured clusters were distributed in 50 microseconds according to the occupancy given by Montecarlo simulations (table 4). The measured signals were taken at a rate of 10MHz, and 505 samples were collected for each single event in a drift time of 50 microseconds. The Montecarlo statistics assumed a 5.66MHz sampling rate and a 80 microseconds drift time. The input signal is shown in figure 13 and the results of a second order digital filter in figure 14.

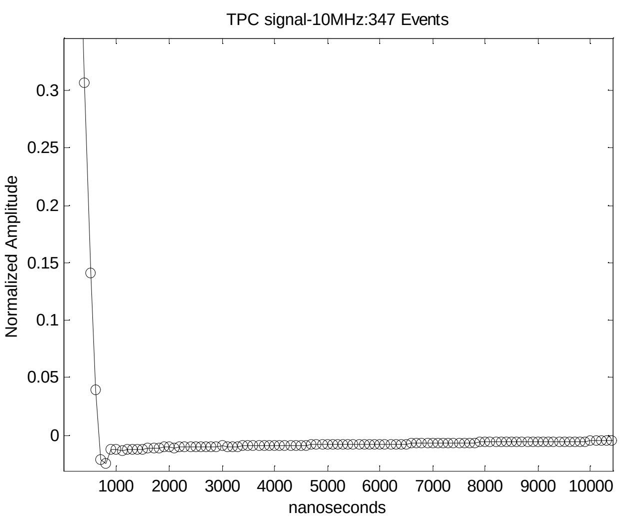

Figure 9: The filter input signal (from NA49-FTPC preamp/shaper)

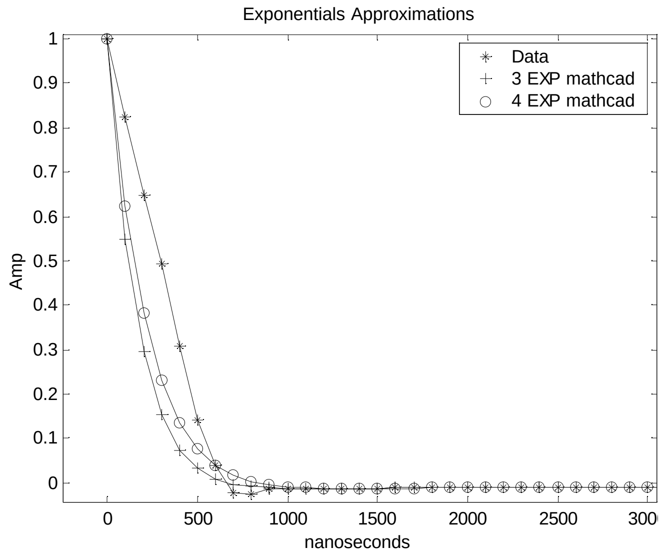

Figure 10: The Exponential Approximations

The input signal was constructed using single measured events (not the average). The input signal presents a downward baseline shift that is clearly visible in the case of multi-cluster events. We can observe that the filter cancels the baseline shift caused by the added tail of each one of the events. The linearity of the system allows the tail cancellation even in the very likely scenario of having superimposed clusters, assuring, in such way, an efficient zero-suppression.

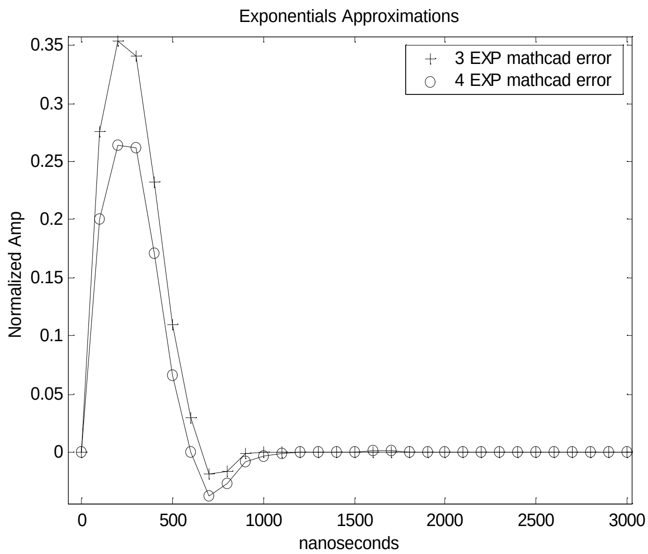

Figure 11: Error for 3 and 4 exponential functions

Table 3: Comparison between the input and the output of a 2nd order and 3rd order filters.

	Input	3EXP	4EXP
Time to 0.1% (us)	35	1.5	1.2
Max. Undershoot (%)	2.5	0	0

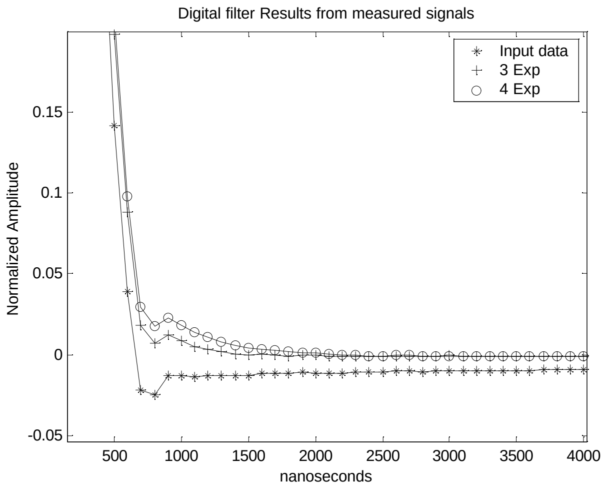

Figure 12: Input, output of the 2nd and 3rd order filter.

Table 4: Statistic data from Montecarlo simulations

Mean time between clusters	3.3 microseconds
Samples/Pad	150
Clusters/Pad	24

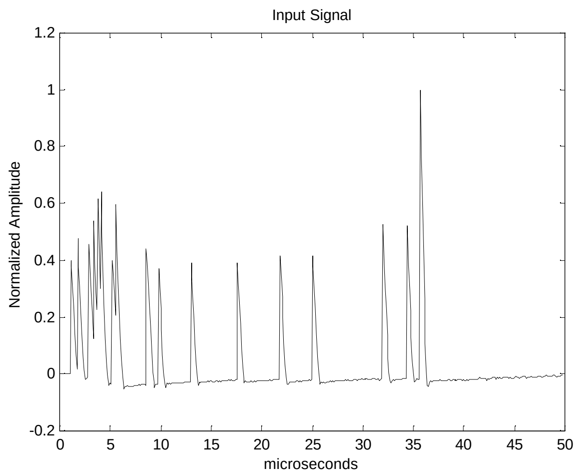

Figure 13: Input signal. 20 events in 50 microsecs randomly distributed.

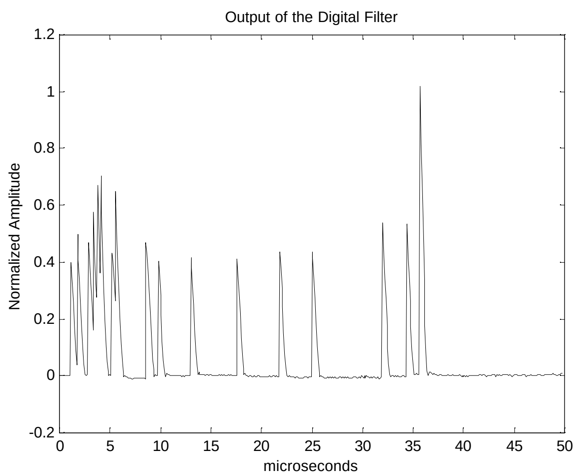

Figure 14: Output of the Digital Filter

V. Conclusions

From the results presented, the proposed scheme can achieve the desired precision of 1‰ within 1-3 μs by a manageable size Digital Filter. The performance increases with the number of approximating exponential functions, in other words, to the expanse of extra silicon. If the input from the analogue electronics presents undershoots, the performance decreases. This is due to the fact that it becomes more difficult, with a limited number of exponential functions, to fit bipolar signals. The Digital Filter can also perform under realistic conditions with several events per pad, allowing a correct zero-suppression. Different architectures of Digital Filters were simulated and compared in terms of silicon area and error vs. number of bits. All the circuits were synthesised using *Synopsys* and simulated with *Verilog*. Other problems of filtering and shaping on the time domain can be addressed using the method proposed here. Nowadays the CMOS sub-micron processes allow arithmetic operations in relatively small area and achieve an accuracy that could not be met using analogue circuits. More measurements will be performed allowing a better understanding of the physics and the correct tuning of the coefficients of the Filter.

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