Noise reduction and Signal Feature Extraction (study) of real time RPC gas detector pulse via HARDROC2B ASIC chip

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1 Introduction

The Real time Resistive plate chamber (RPC) pulse is inserted into the input channel of the HARDROC ASIC.The amplified pulse output taken from the Hardroc Out Fsb0. The Out fsb0 signal is observed on the DSO(digital storage osciloscope). The Out fsb0 signal is very noisy and it is dificult to extract the information from it.To extract the crucial features from the Noisy signal, the Low pass butterworth filter is designed and applied to the noisy pulse(signal), then the signal is filtered and various main parametres of the filtered pulse like Rise Time, fall time, pulse width,bandwidth and peak time is measured by writing the code in matlab.The filtered signal pulses for the different charge input 10fc to 100fc is shown further in given the figures. The present study include filter design technique to reduce the noise significantly and this is further utilized for measuring the various critical time measurement parametre like Rise time , Fall time, pulse width and bandwidth of the signal.

2 RPC Raw pulse Characterstics

The Resistive Plate Chambers (RPCs) are gaseous parallel plate detectors, made of highly resistive electrodes such as glass or bakelite (phenolic resin), to track charged particles. The concept of the RPC was introduced in 1981 by R. Santonico and R. Cardarelli as an alternative to the localized discharge spark counters. The main features of the RPCs are excellent detection efficiency, good homogeneity of the sensitive medium, good spatial as well as position resolutions, wide area coverage and low production cost. In particular it has replaced plastic scintillators, whenever large detecting areas are needed in low counting rate environments.Resistive Plate Chambers (RPCs) that act as the active detector elements. RPC raw Pulse Amplitude generaly vary from 2mV to 5mV.Typical Rise time is order of 1 ns hence the pulses are extremely narrow 5 to10 ns. RPC raw pulse is simmulated using LTspice [12] as shown in fig 1.

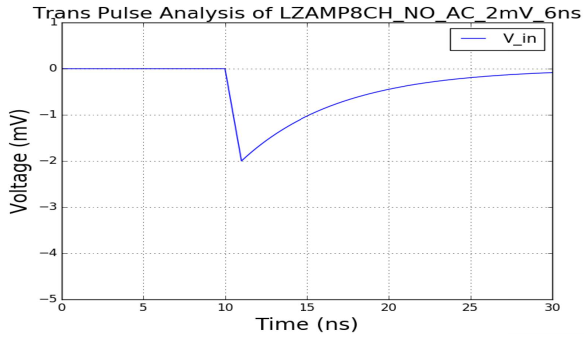


Figure 1: RPC raw pulse simmulated from LT spice

3 Characterstic of RPC pulse through hardroc2B

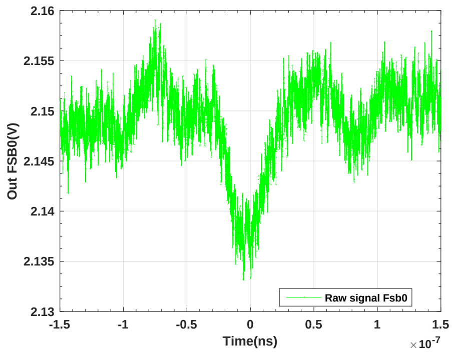
3.1 HARDROC2B ASIC

HARDROC (HAdronic Rpc Detector ReadOut Chip) is the very front end chip designed for the readout of the RPC or Micromegas gaseous detectors. The 64 channels of the 2nd prototype, HARDROC2, are made of[2] - [5].

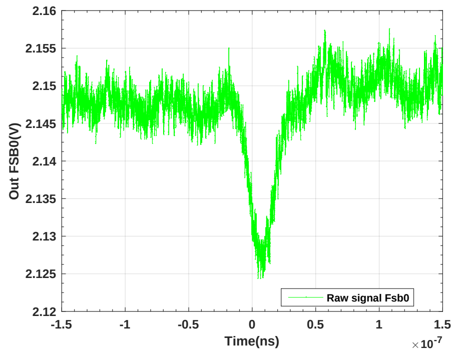
- Fast low impedance preamplifier with a variable gain over 8 bits per channel.
- A variable slow shaper (50-150ns) and Track and Hold to provide a multiplexed analog charge output up to 10pC.
- 3 variable gain fast shapers followed by 3 low offset discriminators to autotrig down to 10 fC up to 10pC. The thresholds are loaded by 3 internal 10 bit- DACs and the 3 discri outputs are sent to a 3 inputs to 2 outputs encoder
- A 128 deep digital memory to store the 2*64 encoded outputs of the 3 discriminators and bunch crossing identification coded over 24 bits counter.
- Power pulsing and integration of a POD (Power On Digital) module for the 5MHz and 40 Mhz clocks management during the readout, to reach 10μW/channel here you can show the image graph.at diffrent charge input.
- Each channel is made of a variable gain preamplifier with low input tunable impedance (50-100Ω), a low offset and a low bias current (5Amp) in order to minimize the cross talk. This variable gain allows adapting thegain depending on the detector choice, up to a factor 2 to an accuracy of 1 percentage with 8 bits. This gain tuning is also convenient to switch off a noisy channel.

3.2 Analog output of the Hardroc2b with Pulser

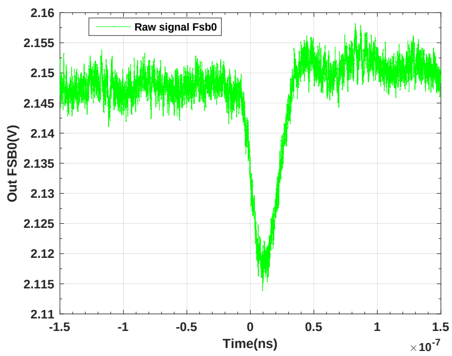
The different calibrated input charges pulses from the pulser feeded into the hardroc and Corresponding analog output (out Fsb0) is observed.which is shown below in the fig2. The Amplifier gain is set to the 128 digital gain [5] and fast shaper is set to 0100, Output signal is taken from the Out Fsb0 port of the hardroc.The out Fsb0 signal is very noisy. Different input charge ranging from 10fC to 100 fC is inserted into the hardroc channel and waveform is shown in figure2.



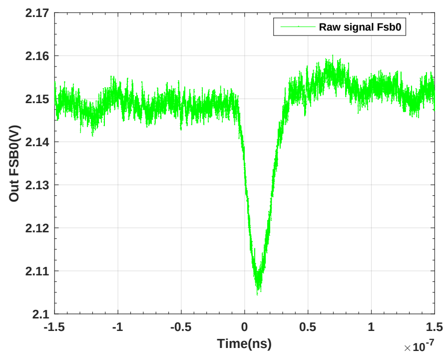
(a) For 10 fC input Charge



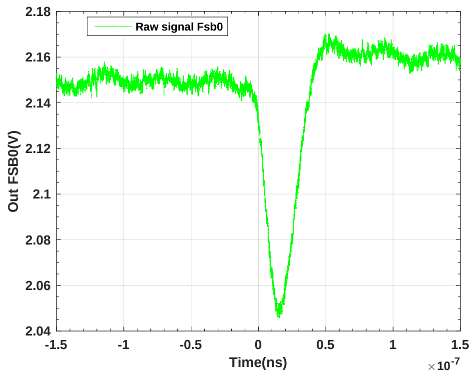
(b) For 20 fC input Charge



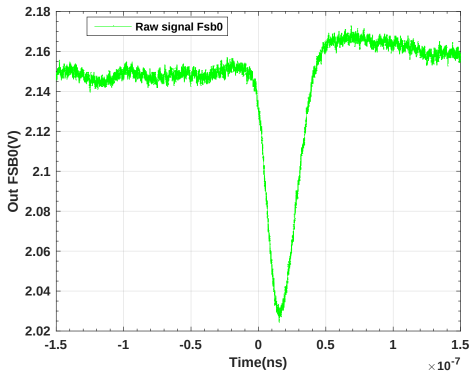
(c) For 30 fC input Charge



(d) For 40 fC input Charge



(e) For 50 fC input Charge



(f) For 60 fC input Charge

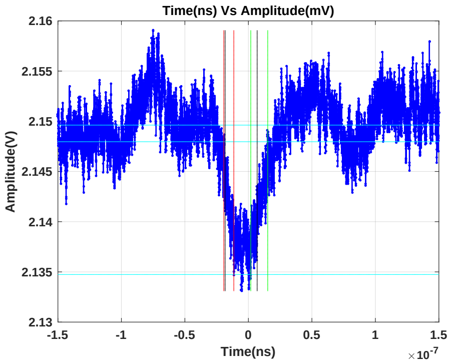
Figure 2: Analog output of hardroc ASIC for different input Charges from the pulser.

3.3 Output of the Hardroc2b for the Rpc pulse input

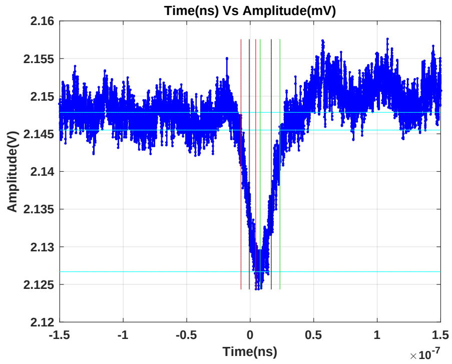
The RPC pulse signal is inserted into the hardroc input channel,The Amplifier gain is set to the 128 and fast shaper is set to 0100, Output signal is taken from the Out Fsb0 port of the hardroc. RPC pulse signal at different Voltages is inserted into the hardroc channe input. need to insert the RPC waveform

4 Motivation of doing Signal processing and designing the filter

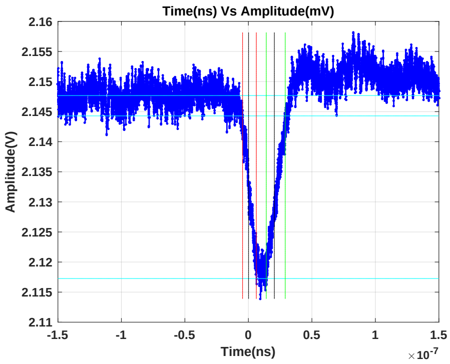
We have seen out fsb0 pulse is very noisy thus it is very much affecting the signal performance and the measurement of parameter like Time measurement [1] and muon track reconstruction by offline correcction. The noisy signals specially for low input charges the pulse noise and amplitude are comparable, some time it is difficult to identify the original pulse and noise and make the parametre measurements difficult [4] .Time measurementt parameter results are not consistant and signal to noise ratio (SNR) is very poor. We need to reduce the noise of these signal with suitably designed filter to overcome problems.



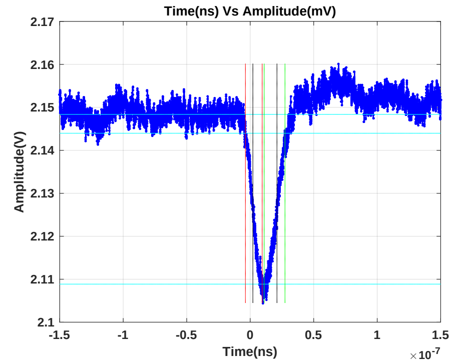
(a) For 10 fC input Charge



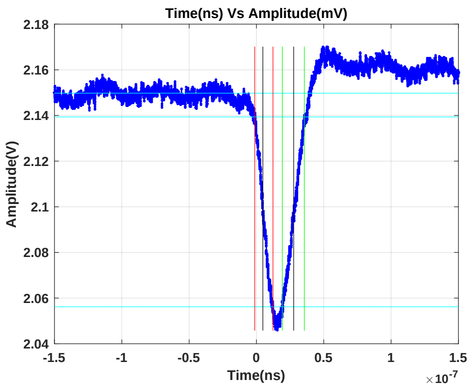
(b) For 20 fC input Charge



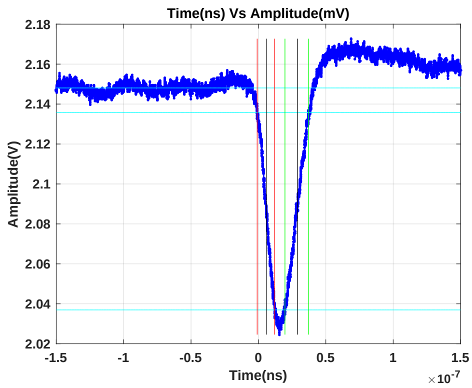
(c) For 30 fC input Charge



(d) For 40 fC input Charge



(e) For 50 fC input Charge



(f) For 60 fC input Charge

Figure 3: Measurement of parameter like Rise time Fall time peak time Pulse width of Analog output of hardroc ASIC for different Charges.

RPC Pulse Parameters

Parameter	**Description**
t_r (Rise Time)	It is the time to rise the pulse from10% to 90% of the maximum amplitude.
t_f (Fall Time)	It is the time taken by the pulse to fall from 90% to 10% of its maximum amplitude.
Pulse Width	Pulse Width is the FWHM(Full Width at Half Maximum).
Peak Time	It is the time to reach the pulse from the 5% of its altitude to maximum amplitude.
V_{max}	Pulse Amplitude is the maximum amplitude of the signal.
t_{max}	The time taken for the pulse to reach its Pulse Amplitude.
$Charge = \int(V/R)dt$	Charge is determined by calculating the area under the pulse[1]

Table 1: parameter Table

5 Experimental setup of Hardroc

The performance of the HARDROC2 has been evaluated through various calibration tests. The aim of the tests was to understand different parameters and working readout modes of HARDROC2 for operation on the single gap RPCs. The HARDROC2 ASIC was bonded with the test board along with required electronic components. The connection between the ASIC and the FPGA for readout and external connectivity has been performed via the surface mounted copper tracks engraved on the testing PCB. An input charge provided by an arbitrary function generator (AFG) is fed into the input channel of the HARDROC2 [2]. The output signal is examined using a digital storage oscilloscope (DSO). The experimental setup used in the measurements is shown in Fig.4.

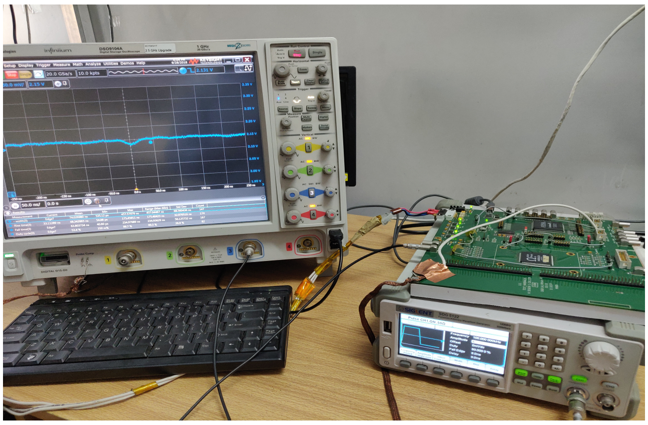
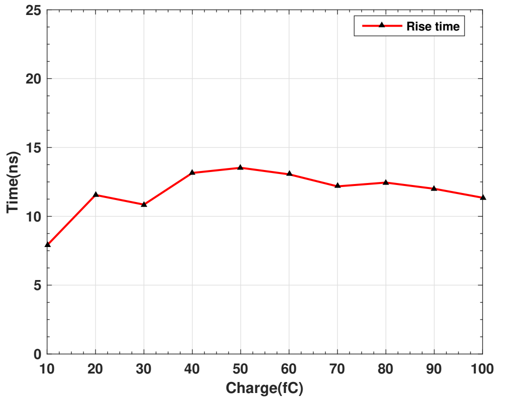


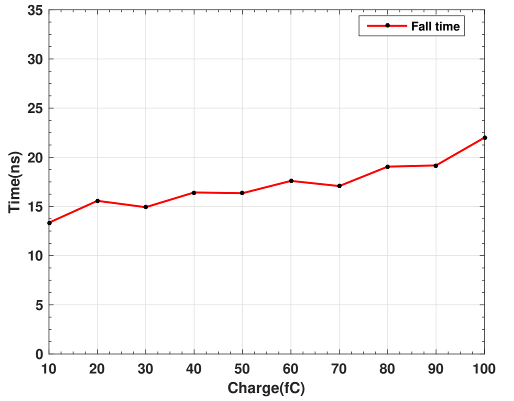
Figure 4: The experimental setup used to perform various tests on a test bench. The picture shows the HARDROC2 board connected to the waveform generators and Digital Oscilloscope

5.1 Parameter Measurements of Out Fsb signal

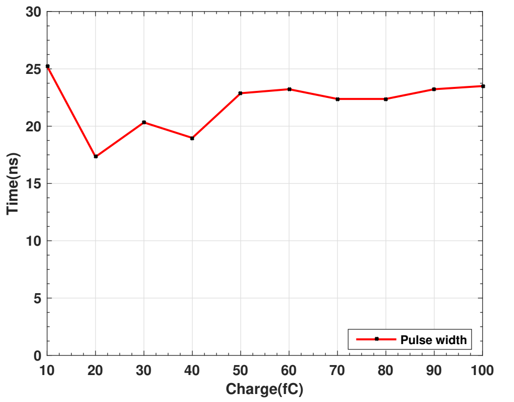
The parametres which is listed in table1 is determined according to its definition. The signal to noise ratio is not good enough to measue the parameter accurately however I have put many programming effort in matlab code to measure the parameter accurately in the existiing noisy pulse but this can not be done accurately without filtering. The measurement result is ploted with diffrent charge inputs. It will be better to reduce the noise before determine the parameter. I will try to make signal to noise ratio better using the Signal processing tool in Matlab in the next section .



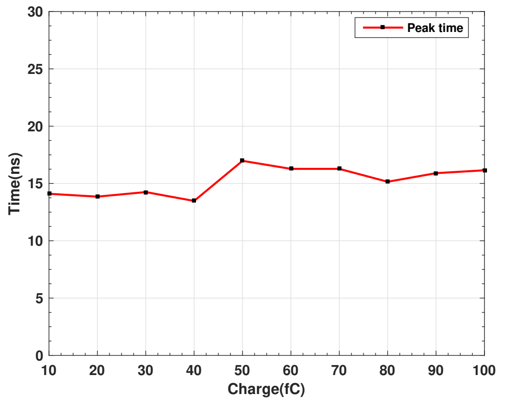
(a) Rise time with different input charges



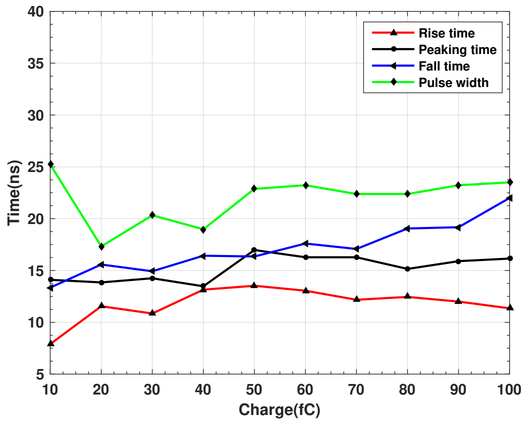
(b) Fall time with different input charges



(c) Pulse width diffrent cahrge inputs



(d) Peaking time diffrent cahrge inputs



(e) Overlapped all plots for diffrent input Charges

Figure 5: Time Measurement Result (a) Rise time (b) Fall time (c) Pulse width (d) peak time (e) Overlapped plot of all measurements of Analog output of hardroc ASIC for different Charges.

6 Signal processing and Bandwidth Extraction of RPC pulse signal via Hardroc

Any meaningful information is called the signal.The meaning information defines here the behavior or attributes of some phenomenon.The phenomena which are broadly defined such as sound,images and electronic measurements.The process of signal modification, analysis and synthesis is called signal processing.

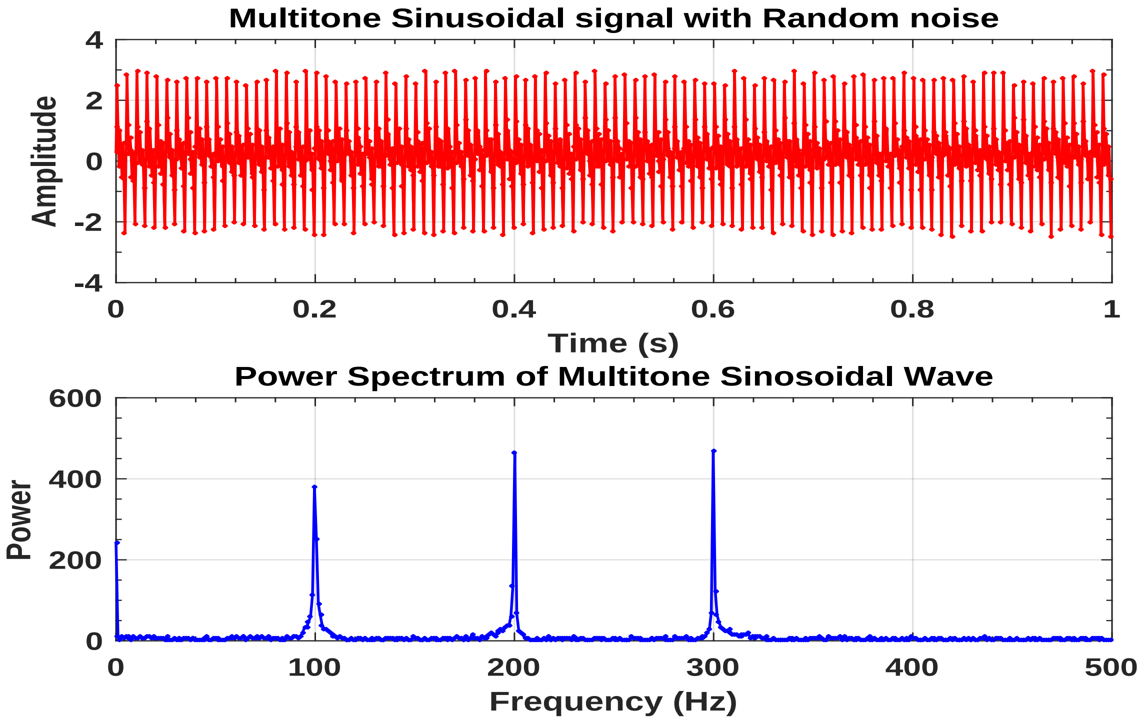


Figure 6: Signal processing example

here the signal is manipulated or modifed as we like such as filter the unwanted signal or amplify the intended signal .

The obtained data is sampled with sampling frequency of 40GHz, therefore the Sampling interval will be 25 psec.The bandwidth information of the raw pulse signal is extracted by taking the fast fourier transform [6] [11] of the signal using matlab. here fast fourier transform (fft) of the signals for the different charges inputs from 10fC to 100fC is calculted and ploted in given figure. From here we are able to calculate the bandwidth of the pulses.The pulse frequency spectrum gives information of the bandwidth of our interest signal.The approximate bandwidth of the signals is 40MHz.

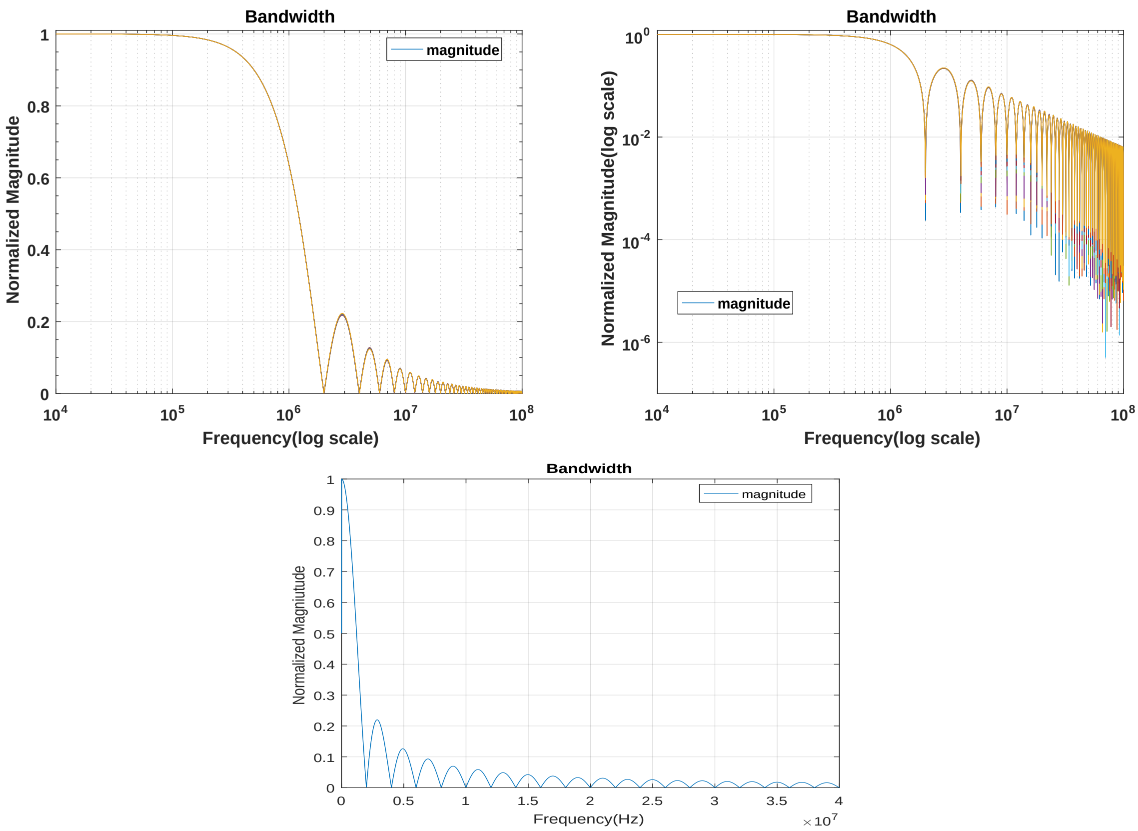


Figure 7: Signal bandwidth information

7 Finite Impulse Respone (FIR) filter Design

7.1 FIR filter design procedure

FIR stands for finite impulse response. FIR filter system is of a finite length. The term finite impulse response arises because the filter output is computed as a weighted, finite term sum, of past and present of the filter input. Fir filter is always stable, the phase can be made exactly linear, FIR is an all zero system, represented by the difference equation[7] - [11] .

$$y(n) = \sum_{k=0}^{N} b(k)x(n-k) \tag{1}$$

which is clearly a convolution of the input sequence x(n) with an impulse response b(n) for k = 0 to N. Here N is finite. The Ideal FIR low- pass filter has a response

$$H(e^{j\omega}) = \begin{cases} 1 & \text{for} |\omega| \leq \omega_c. \\ 0 & \text{for} \omega_c \leq |\omega| \leq \pi. \end{cases} \tag{2}$$

The Inverse fourier transform of the above ideal frequency response that is impluse response of the Fir low pass filter[7] , which is shown below.

$$h(n) = \frac{\omega_c}{\pi}\left(\frac{sin(\omega_c n)}{\omega_c n}\right) \quad \text{for } -\infty < n < \infty. \tag{3}$$

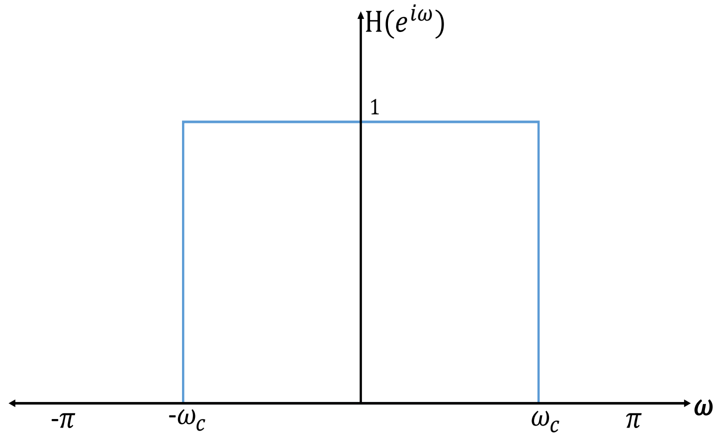
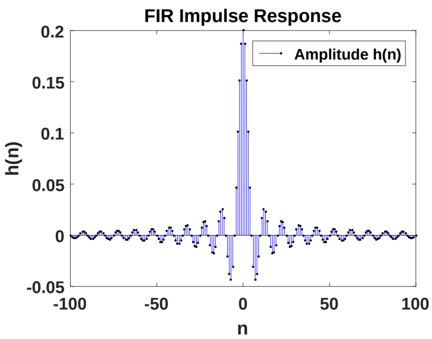
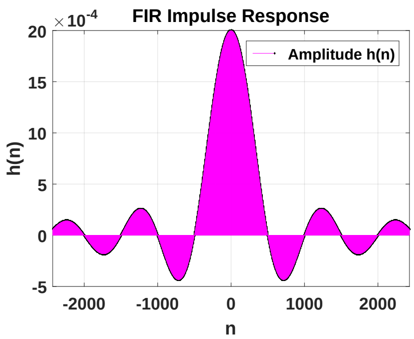


Figure 8: Frequency Response of Ideal low pass Filter$H(e^{j\omega})$

the below given figures shows the central region of the impulse response of an ideal Filter with $\omega_c = 0.2\pi$ and $\omega_c = 0.002\pi$ respectivily.



(a) Ideal FIR filter with $\omega_c = 0.2\pi$



(b) Ideal FIR filter with $\omega_c = 0.002\pi$

There are two problems observed in this impulse respone:

(a) The response is infinite in length.

(b) The response is non-causal.

To make filter Response finite length truncate the impulse response by multiplying h(n) with an even window function w(n) of finite length N+1.There are many window functions available which we can use.here we are using hamming window for our Fir filter design.

7.2 The Hamming window

This is raised cosine window with a pedestal value of 0.08 at the extremities $w(-\frac{N}{2})$ and $w(+\frac{N}{2})$. The side lobes of hamming window are much smaller than the side lobes of the rectangular window. but the main lobe of hamming window wider than the main lobe of rectangular window . As the tappernes increases, width of the main lobe increases

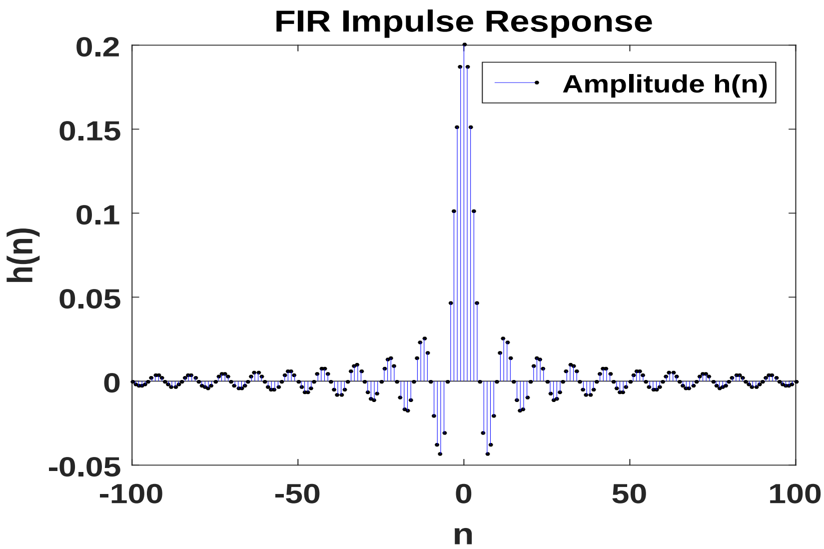


Figure 10: Ideal FIR filter with $\omega_c = 0.2\pi$

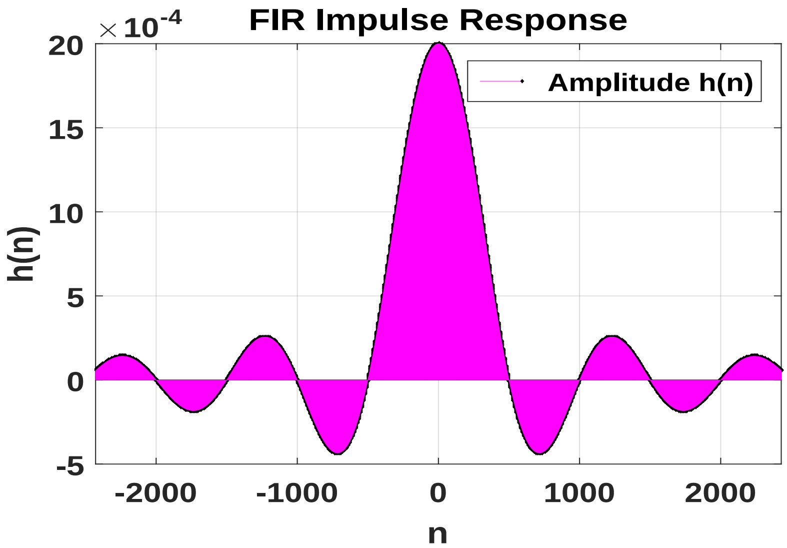


Figure 11: Ideal FIR filter with $\omega_c = 0.002\pi$

which causes the increment in the transition width but at the same time increment in taperness causes the amplitude of the side-lobes decreases more rapidly away from the main lobe, with the result that the filter stop-band attenuation is significantly increased at high frequencies[7]. The hamming window function is shown below

$$w(n) = \begin{cases} 0.54 + 0.46cos\left((\frac{2\pi}{N})n\right) & \text{for} - \frac{N}{2} \leq n \leq +\frac{N}{2} \\ 0 & \text{otherwise} \end{cases} \tag{4}$$

The hamming window shown in following figure. when the response is truncated we can

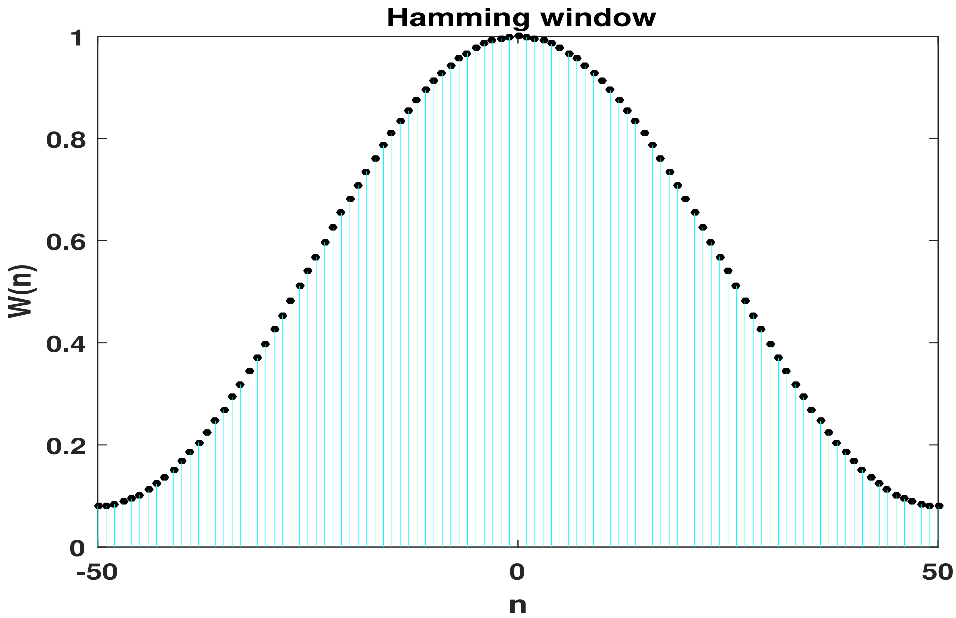


Figure 12: Hamming Window

right shift the response by N/2 sample and make the impulse response causal.

7.3 Filter Design using Windowing method

Low pass Filter Design using Windowing method: These are the only design parameters using the fixed window[7] [8].

- Low-pass cut-off frequency ω_c: First we need to determine the low-pass cut-off frequency ω_c ; here the pulse signal bandwidth 20 MHz is observed, beyond this frequency the any signal is considered as a noise signal. Therefore cut-off frequency ω_c = 20MHz.

- Choice of window type: here hamming window is choosen for low pass filter. Window is a vector of N+1 length. N+1 must be the odd Number. N will be even Number so that N/2 samples will be on the Negative time axis side and N/2 samples will be in positive time axis side.one sample will be on the origin.

- The filter length N + 1: Finite length approximations to the ideal impulse response lead to the presence of ripples in both the pass band and the stop band of the filter, as well as to a nonzero transition width between the pass band and stop band of the filter. (See Figure) The filter length is N + 1. There is an empirical formula to specify the order of the filter. This empirical formula calculates the order of the filter on the basis of the pass band, stop band ripples and transition band.

$$N = \frac{-10\log_{10}\delta_1\delta_2 - 15}{14\Delta}; \tag{5}$$

where δ_1 and δ_2 is Passband and stop band ripple respectivily. Δ is transition width.

$$\Delta = \frac{\omega_s - \omega_p}{2\pi}$$

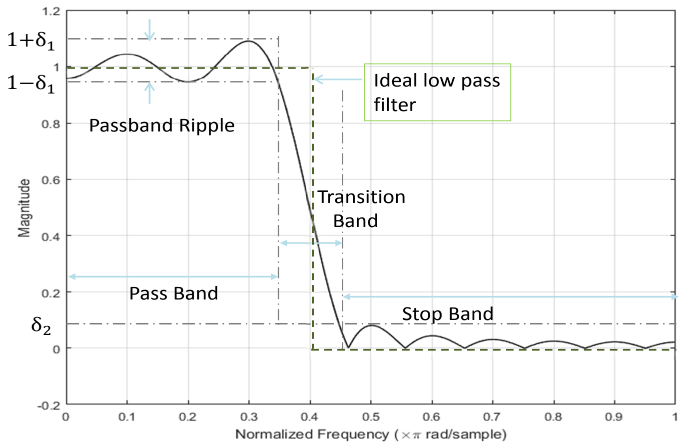


Figure 13: Characteristics of practical Fir Filter

Once these above parameters are determined, the procedure is as follows

(a) As above we have decided the Cutt-off frequency ω_c = 20MHz of ideal low pass filter. According to this the impulse response of ideal low- pass filter is derived which is given below.

$$h(n) = \frac{\omega_c}{\pi}\left(\frac{sin(\omega_c n)}{\omega_c n}\right) \quad \text{for } -\frac{N}{2} \leq n \leq +\frac{N}{2} \tag{6}$$

(b) The ideal low pass filter impulse response has infinite length. To make it finite length response multiply by finite length using hamming window of N+1 length.

$$h'(n) = h(n) \cdot w(n)$$

here $w(n)$ is window of N+1 length. here the window size N = 4858 is choosen, according to the pass band and stop band attenuation using the emperical formula in equation 5. The selected window is shown below. As we have choosen the windw w(n) for restricting the response to a finite length and multiplied the same window with the impulse response of the ideal low pass filter h(n). Finally we get the the impulse response hprime which is our designed filter response and frequency response shown below.

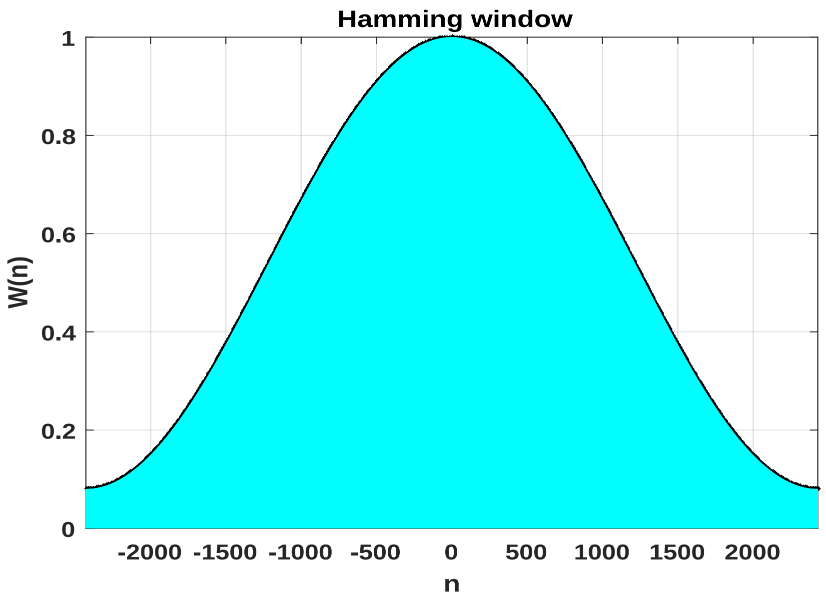


Figure 14: Hamming Window with N = 4858

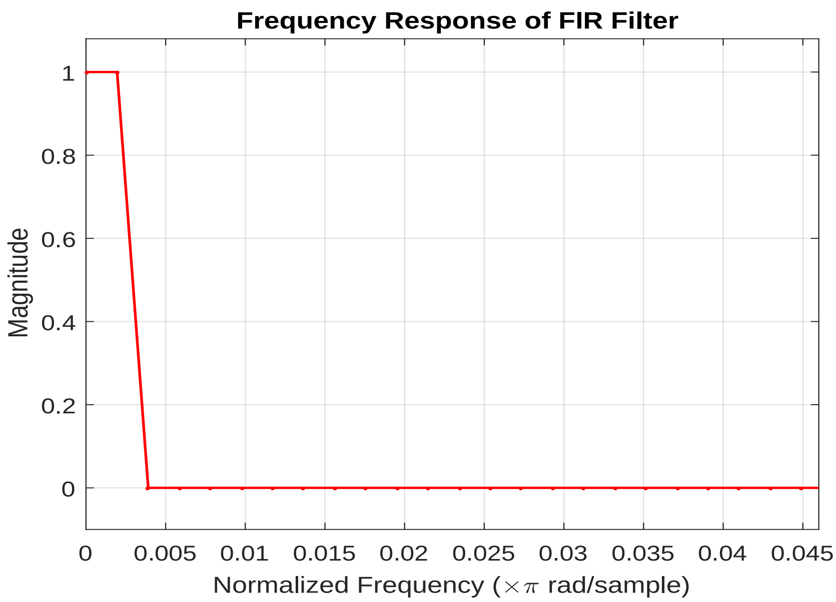
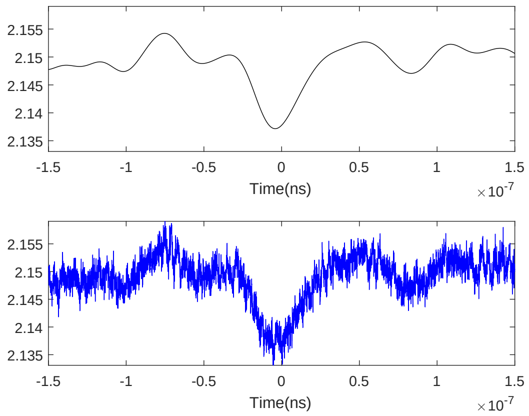


Figure 15: Frequency Response of Fir filter $h'(n)$

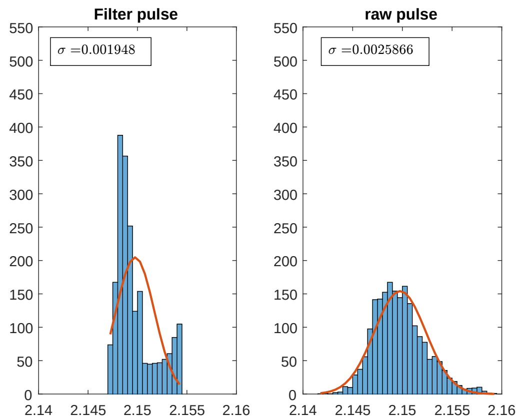
(c) For making the our designed filter impulse response causal, Shift all samples to the right by N/2 samples.

7.4 Filtered Hardroc Pulse using our optimized FIR Design

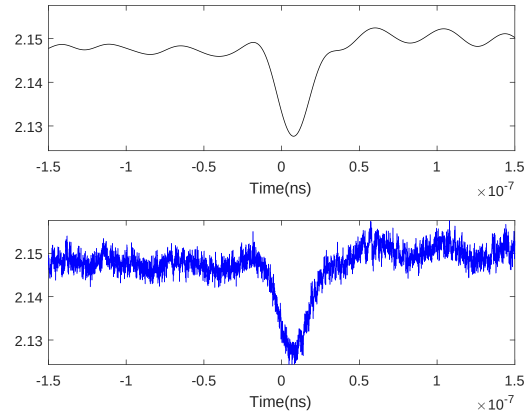
The noisy pulse signal is improved using our optimized FIR design. here Some filtered signal is shown with respect to the noisy signal. As we have used the fir filter and showed some result of improved signal.



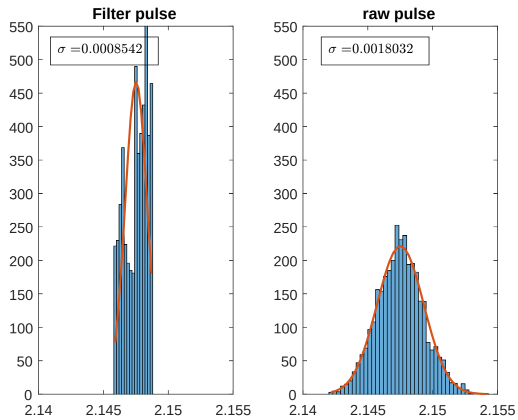
(a) For 10 fC input Charge



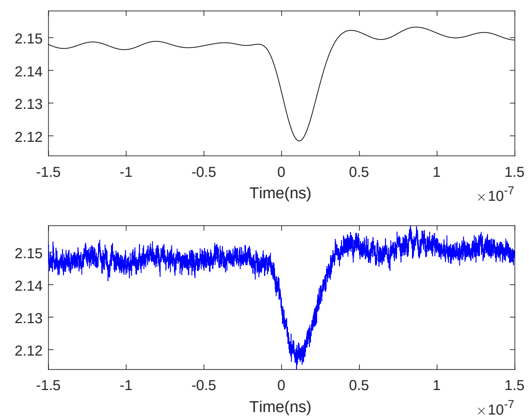
(b) For 10 fC input Charge



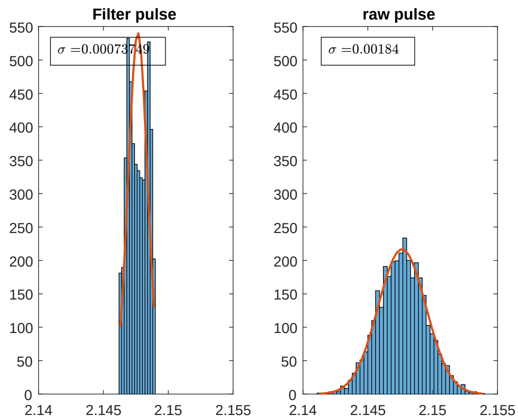
(c) For 20 fC input Charge



(d) For 20 fC input Charge

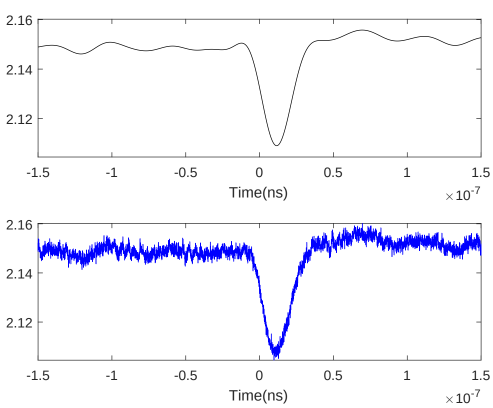


(e) For 30 fC input Charge

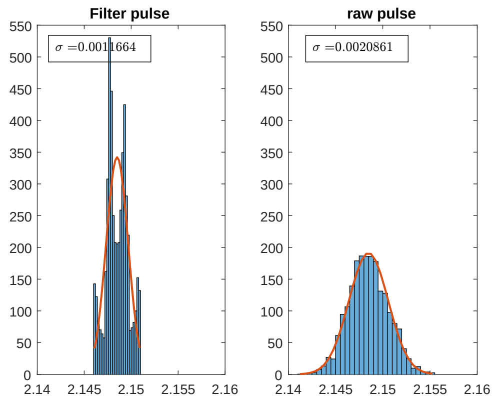


(f) For 30 fC input Charge

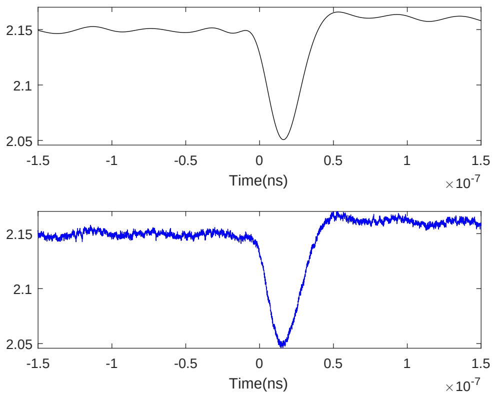
Figure 16: Filtered, Raw signal and histogram of filtered and Raw signal for 10fc to 30fc.



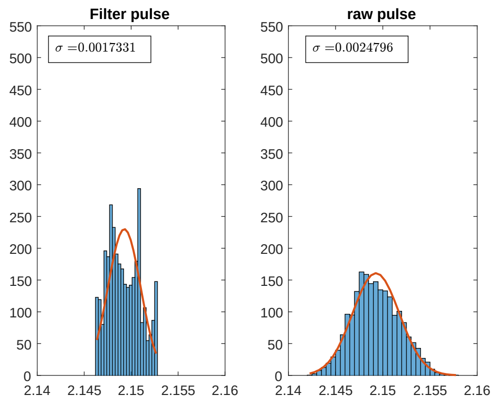
(a) For 40 fC input Charge



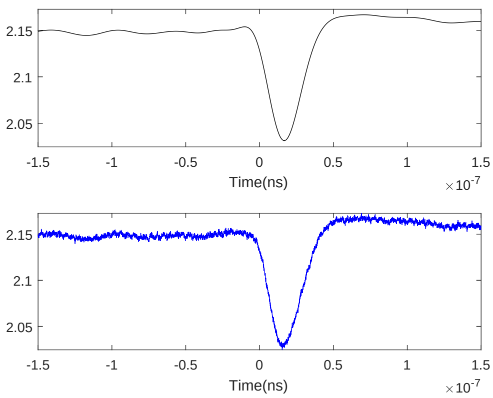
(b) For 40 fC input Charge



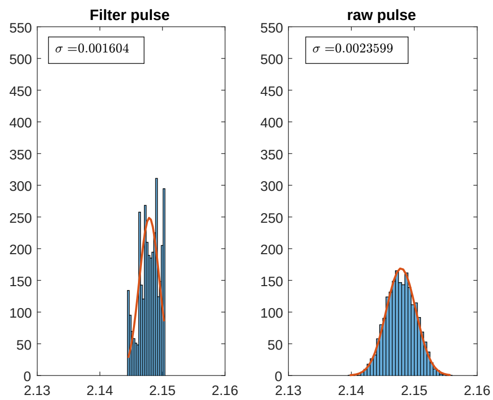
(c) For 50 fC input Charge



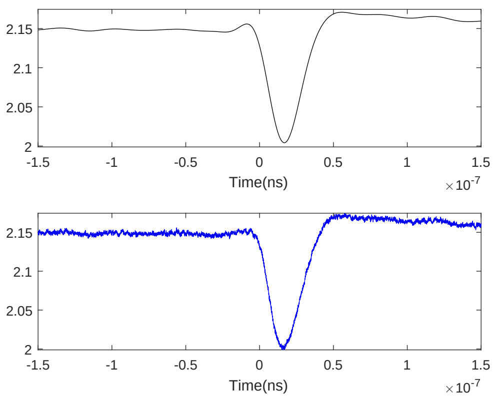
(d) For 50 fC input Charge



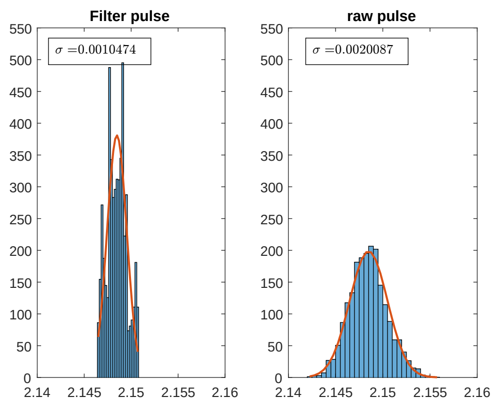
(e) For 60 fC input Charge



(f) For 60 fC input Charge

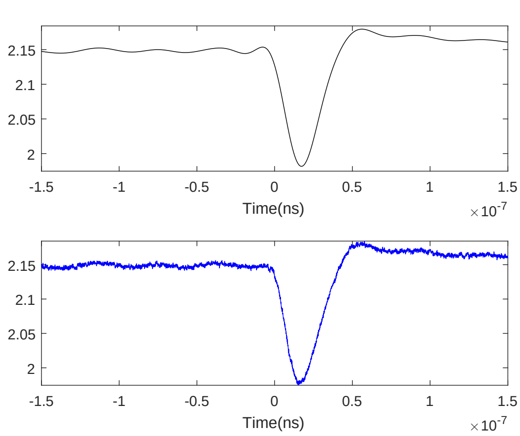


(g) For 70 fC input Charge

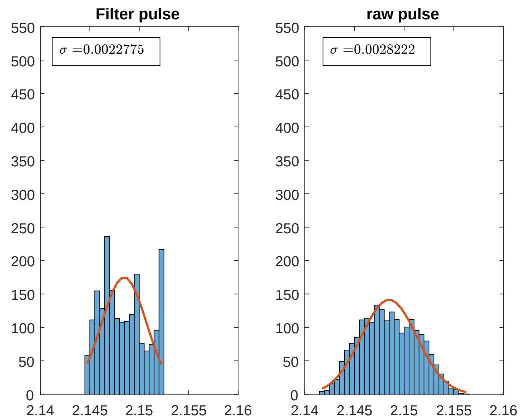


(h) For 70 fC input Charge

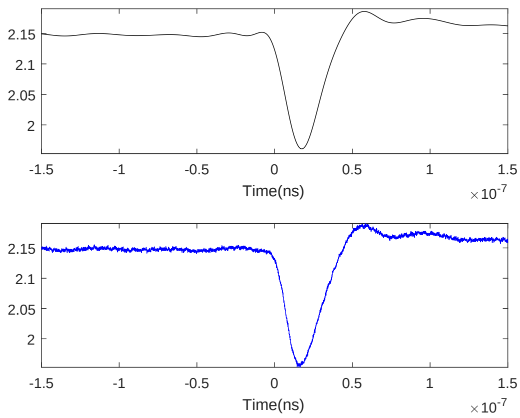
Figure 17: Filtered, Raw signal and histogram of filtered and Raw signal for 40fc to 70fc.



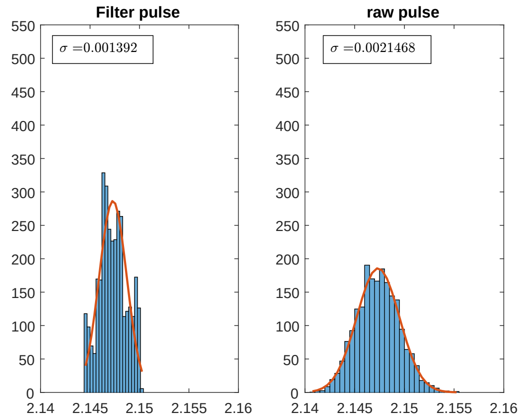
(a) For 80 fC input Charge



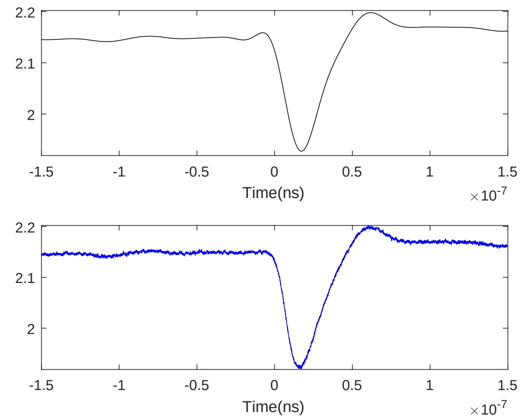
(b) For 80 fC input Charge



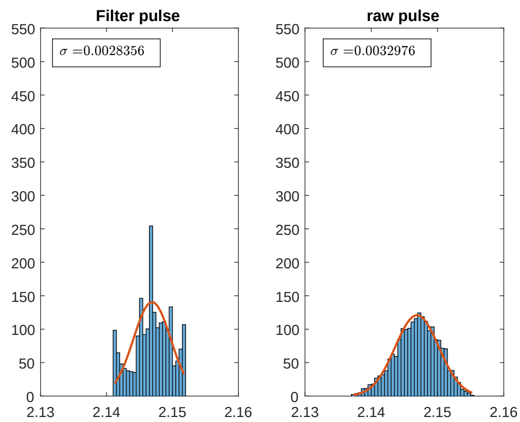
(c) For 90 fC input Charge



(d) For 90 fC input Charge



(e) For 100 fC input Charge



(f) For 100 fC input Charge

Figure 18: Filtered, Raw signal and histogram of filtered and Raw signal for 80fc to 100fc.

8 Results of FIR Filter Design

The performance of the proposed optimized Fir filter design algorithm has been studied using matlab.The standard deviation and peak to peak volatge of filtered and Noisy pulse signals is observed. The good improvement is found in the filterd signal as shown in the given figure.

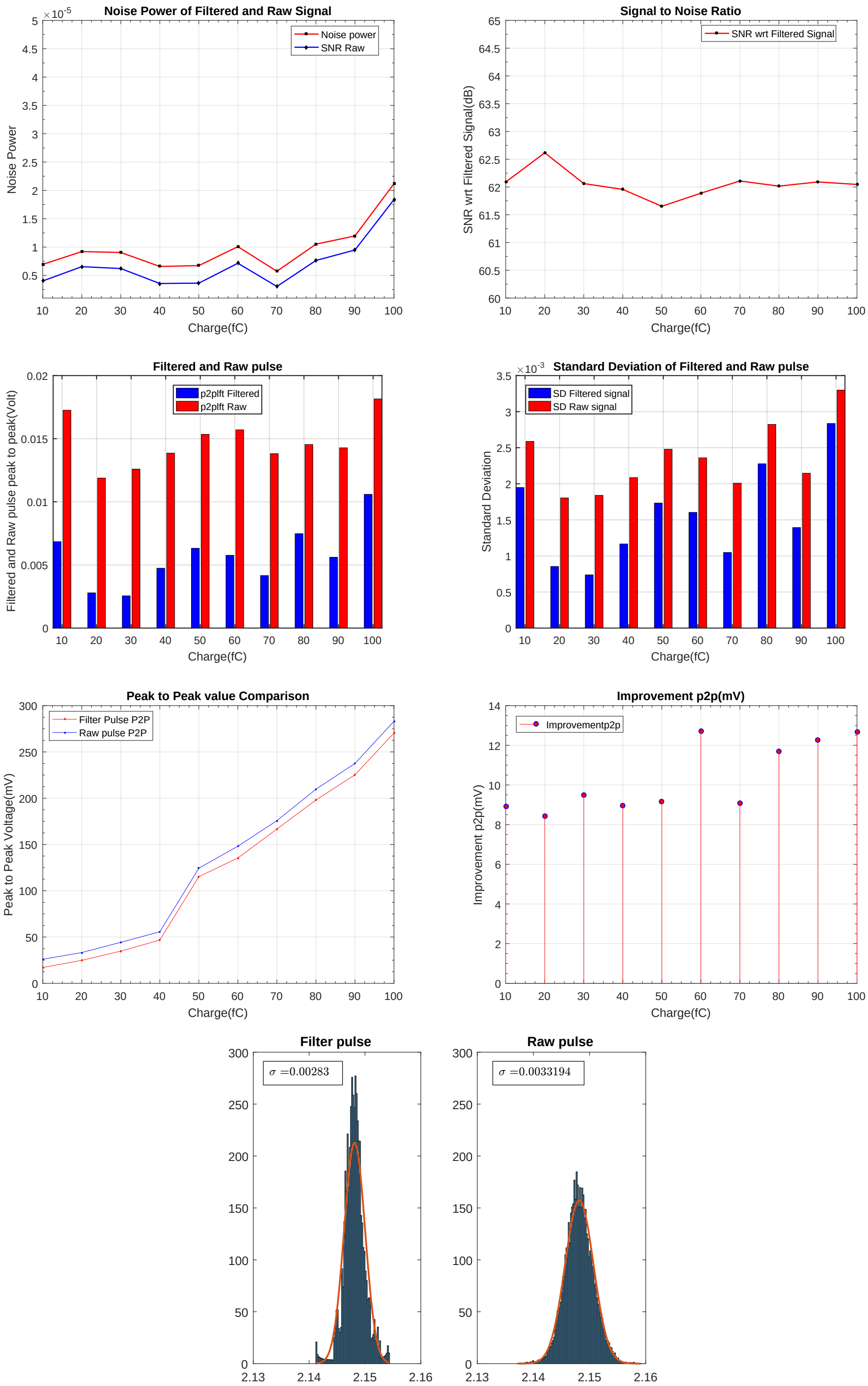


Figure 19: Quantitative comparison of filtered and Raw signal

9 IIR Filter

The IIR system is infinite impulse response system.The Present output of IIR system is depends on present input, past inputs and past outputs.An IIR filter is characterized by a recursive difference equation

$$y(n) = \sum_{k=1}^{N} b(k)y(n-k) + \sum_{k=0}^{M} b(k)x(n-k). \tag{7}$$

where y(n) is output sequence , x(n) is input sequence and b(n) is an impulse response. An unwanted signal which is present in the signal is called noise.The process of weakening the background noise and suppress the interfering signals by rejecting the some frequencies is called filtering. There are various types of filters which are classified based on various criteria such as linearity or non-linearity, time variant or time invariant, analog or digital, active or passive.

9.1 Design of IIR Filters

IIR filters have the advantage that they can give a better cut-off characteristic than a FIR filter of the same order, but have the disadvantage that the phase response cannot be well controlled. The most common design procedure for digital IIR filters is to design a continuous filter in the s-plane, and then to transform that filter to the z-plane. Because the mapping between the continuous and discrete domains cannot be done exactly, the various design methods are at best approximations. These are the some design procedure for digital IIR filters [7]-[8].

- Design by Approximation of Derivatives:
- Design by Impulse-Invariance:
- Design by the Matched z-Transform (Root Matching)
- Design by the Bilinear Transform

9.2 Low Pass Butterworth Filter Design

In 1930 physicist and the British engineer Stephen Butterworth described about a Butterworth filter in his "on the theory of filter amplifiers" paper for the first time. Hence, this type of filter named as Butterworth filter.The Butterworth filter is a type of signal processing filter designed to have a frequency response as flat as possible in the passband.Butterworth filter has a magnitude response that is maximally flat in the passband and monotonic overall.In practice however, Butterworth's ideal frequency response is unattainable as it produces excessive passband ripple.A low-pass filter (LPF) is a filter that passes signals with a frequency lower than a selected cutoff frequency and attenuates signals with frequencies higher than the cutoff frequency. The exact frequency response of the filter depends on the filter design.

The bandwith of the signal is determined in section 3.5 is 40MHz and corresponding figure shown in figure 7, The cuttoff frquency is 40MHz. so we can remove high frequency noise by using the low pass filter of 40MHz cuttoff frequency.

Matlab butter function uses these following steps to implement the our specific filter design with cuttoff frequency 40Mhz and order 6.

It uses the butterap function to determine the zeros and pole locations of the prototype low pass filter then by using the zp2tf function it forms the transfer function H(s) as shown in figure.

[z,p,k] = buttap(N); Where N is the order of filter
[NUM,DEN] = zp2tf(z,p,k);

$$H(s) = \frac{NUM(s)}{DEN(s)}; \tag{8}$$

this returns a set of zero locations in vector Z, a set of pole locations in vector P, and a gain in scalar K. Vectors NUM and DEN are returned with numerator and denominator coefficients in descending powers of s.

For digital filter design, The analog filter converted into the digital filter by using the bilinear transformation with frequency prewarping. Careful frequency adjustment enables the analog filters and the digital filters to have the same frequency response magnitude at Wn.

9.3 Frequency Response for a Butterworth Filter

here the butterworth filter passband and stop band frequency is determined according to the bandwidth of the signal. Earlier in the above section we have determined the normalized cuttoff frequency 0.002 of the filter and order of the designed filter is 6. The frequency response of the designed butterworth filter shown below.

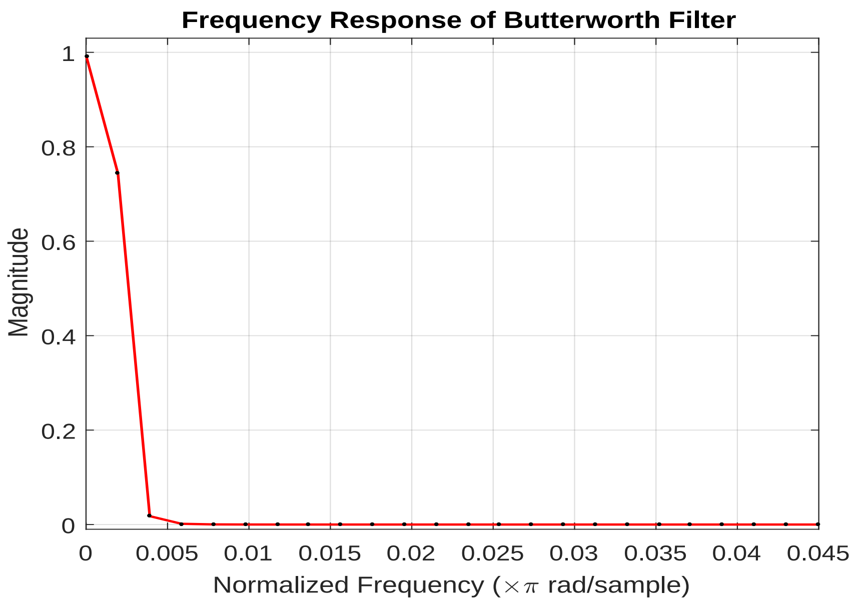
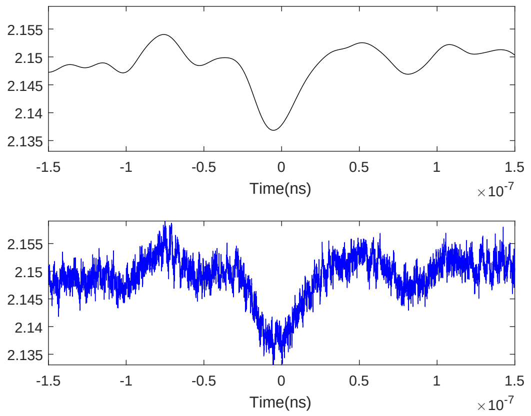


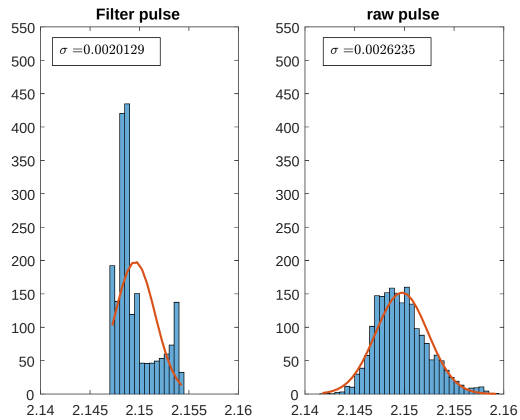
Figure 20: Frequency Response of butterworth filter

9.4 Filtered Hardroc Pulse using our IIR Filter Design

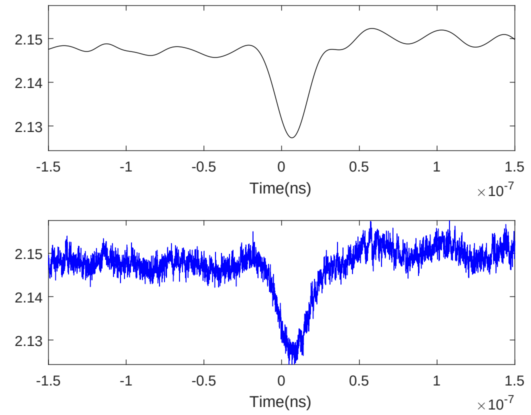
The noisy pulse signal is improved using our optimized IIR Filter design. here Some filtered signal is shown with respect to the noisy signal. As we have used the IIR filter and showed some result of improved signal.



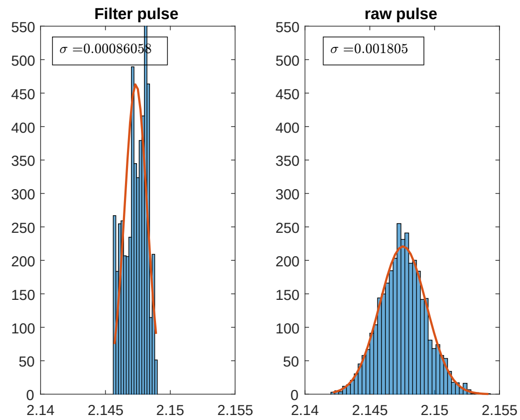
(a) For 10 fC input Charge



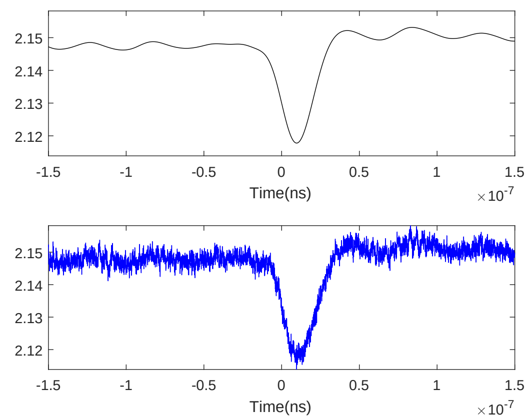
(b) For 10 fC input Charge



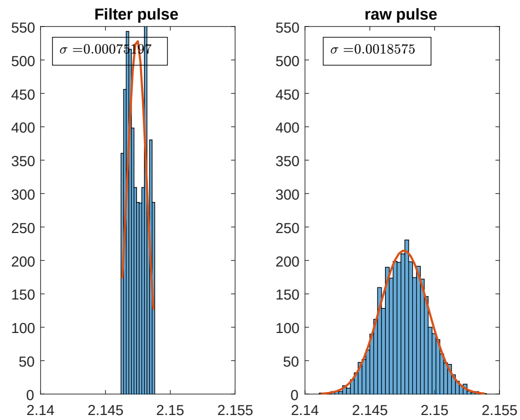
(c) For 20 fC input Charge



(d) For 20 fC input Charge

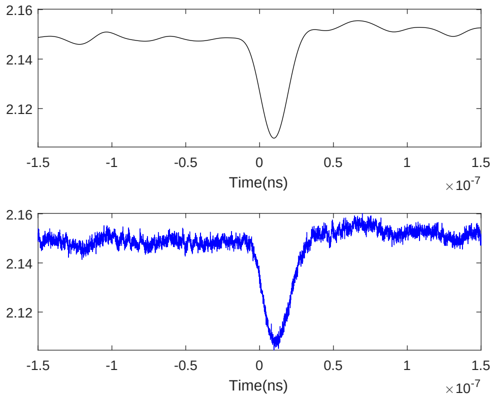


(e) For 30 fC input Charge

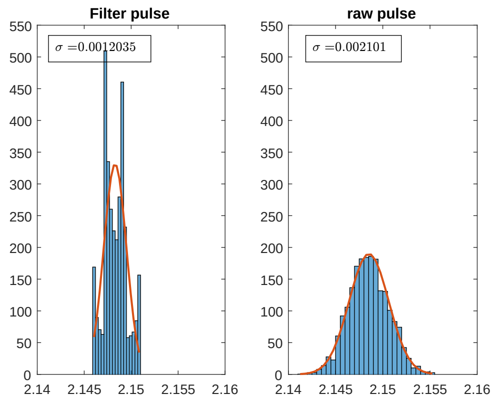


(f) For 30 fC input Charge

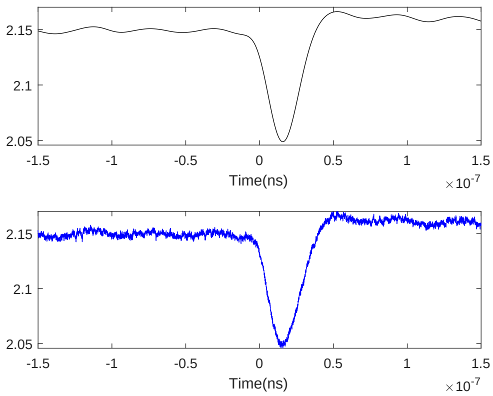
Figure 21: Filtered, Raw signal and histogram of filtered and Raw signal for 10fc to 30fc.



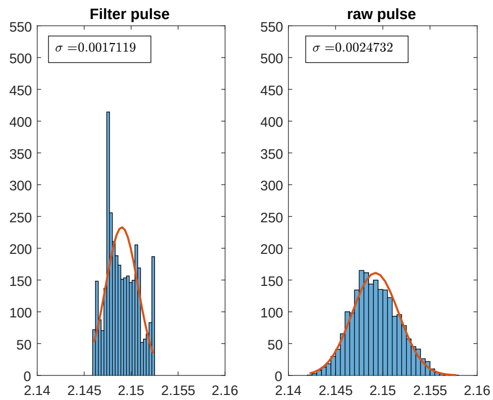
(a) For 40 fC input Charge



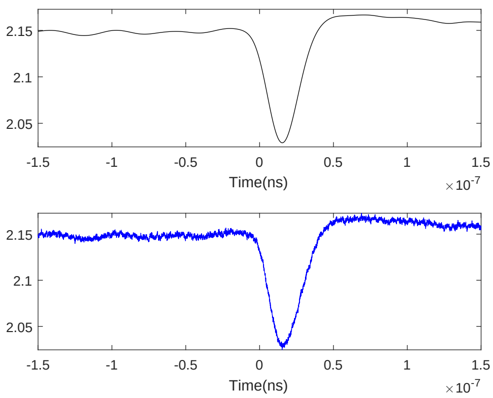
(b) For 40 fC input Charge



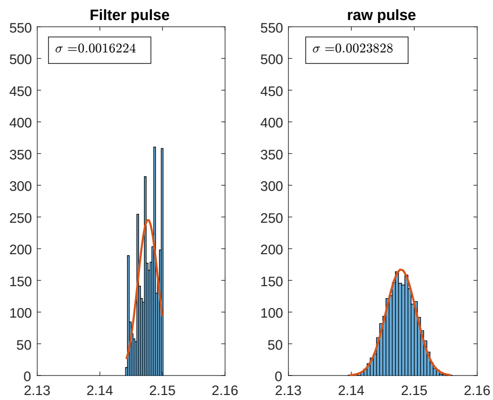
(c) For 50 fC input Charge



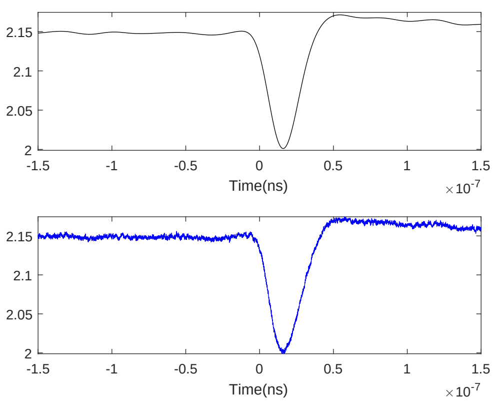
(d) For 50 fC input Charge



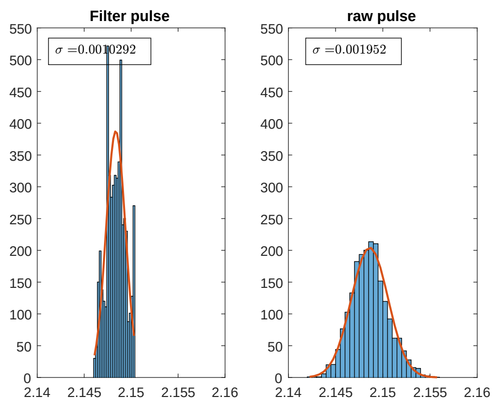
(e) For 60 fC input Charge



(f) For 60 fC input Charge

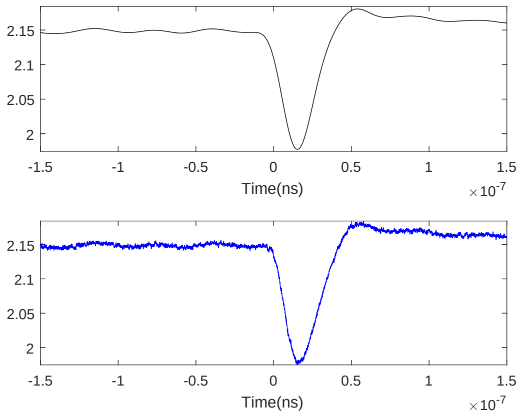


(g) For 70 fC input Charge

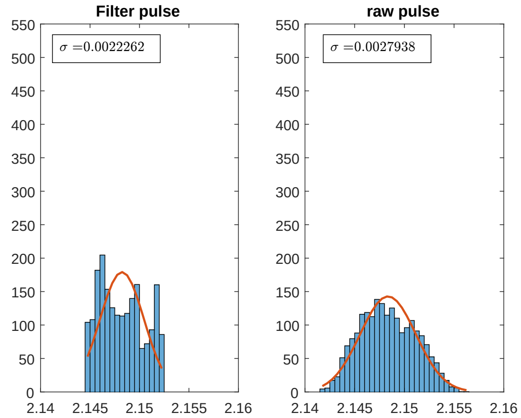


(h) For 70 fC input Charge

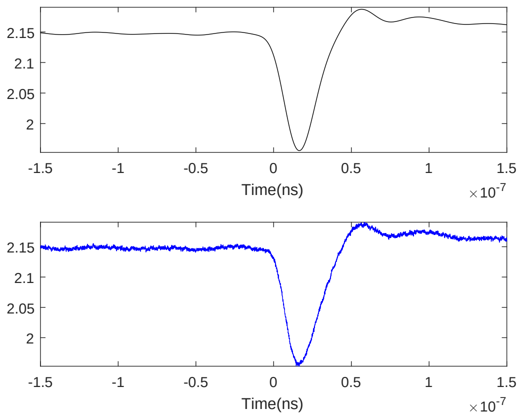
Figure 22: Filtered, Raw signal and histogram of filtered and Raw signal for 40fc to 70fc.



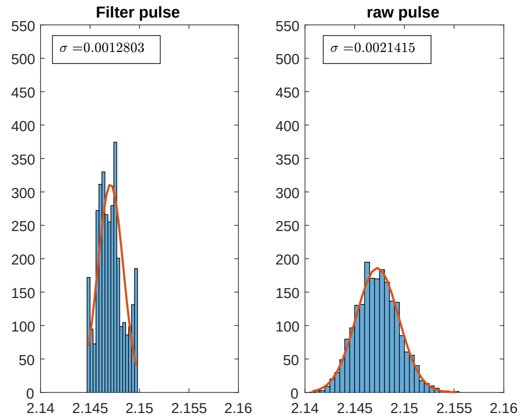
(a) For 80 fC input Charge



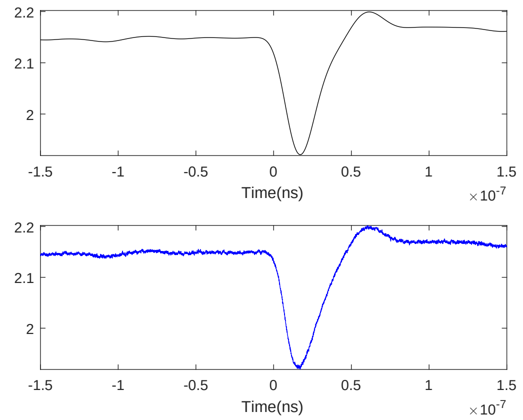
(b) For 80 fC input Charge



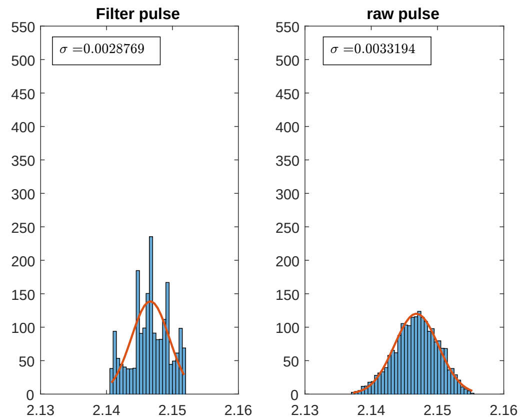
(c) For 90 fC input Charge



(d) For 90 fC input Charge



(e) For 100 fC input Charge



(f) For 100 fC input Charge

Figure 23: Filtered, Raw signal and histogram of filtered and Raw signal for 80fc to 100fc.

10 Results of IIR Filter Design

The performance of the proposed optimized IIR filter design algorithm has been studied using matlab.The standard deviation and peak to peak volatge of filtered and Noisy pulse signals is observed. The good improvement is found in the filterd signal as shown in the given figure.

Figure 24: Quantitative comparison of filtered and Raw signal

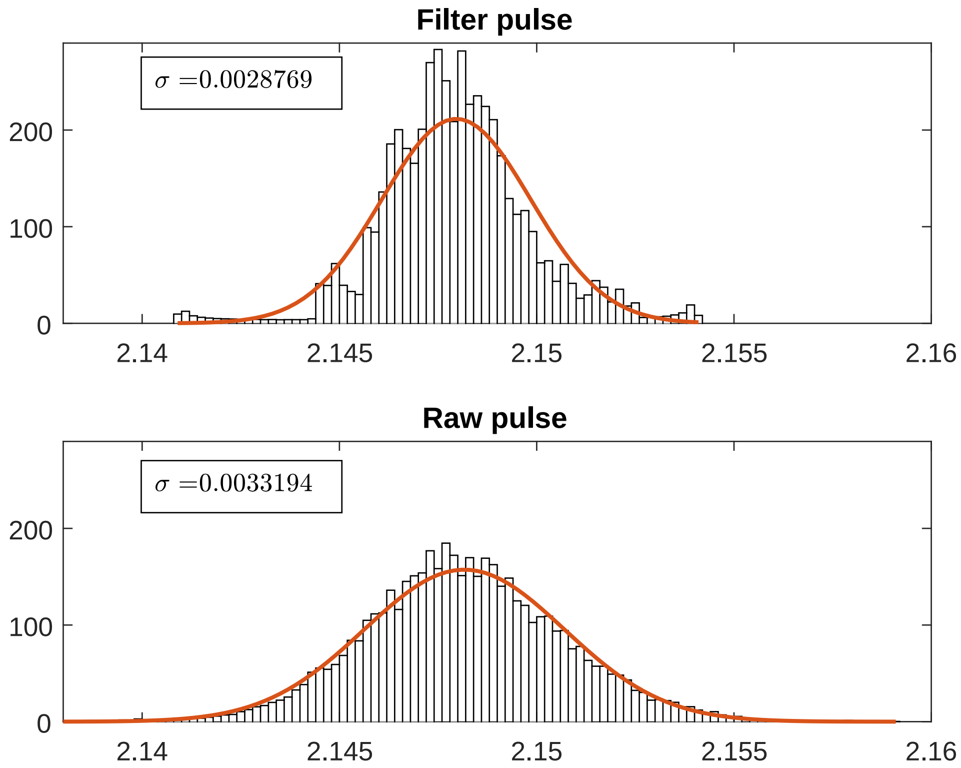


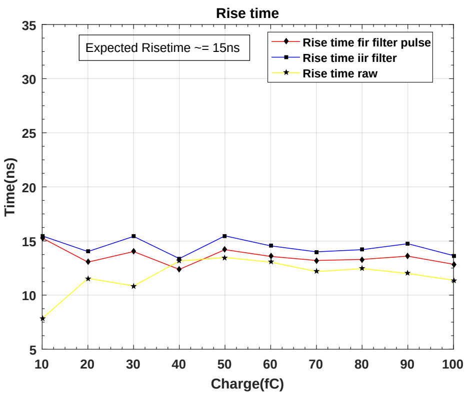
Figure 25: Frequency Distribution of the filtered and Raw signal

11 Feature extraction algorithm of filtered signal

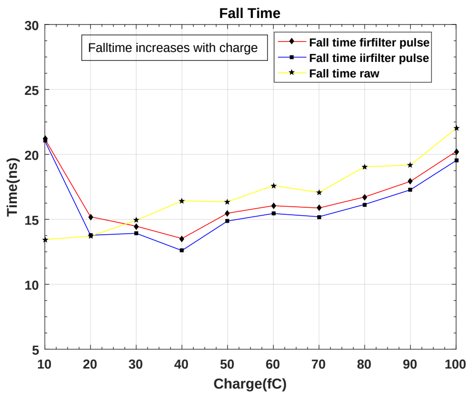
The feature extraction is the process of taking out some significant characteristics or pieces of information or property from the signals. These characteristics may be some quantitative data measured for some frame reference. We have used the low pass FIR and IIR filter to reduce the noise of the hardroc signal pulse. The bandwidth is an important characteristic to decide the cut off frequency of the low pass filter. The bandwidth is computed the with FFT (fast Fourier transform) algorithm in MatLab. Once the bandwidth is determined accordingly the cutoff frequency is decided. Then The Low pass FIR and IIR filter are designed. The Pulse parameter is extracted after developing the MatLab code according to the time measurement definition.

11.1 Improvement in Time Measurement after Filtering

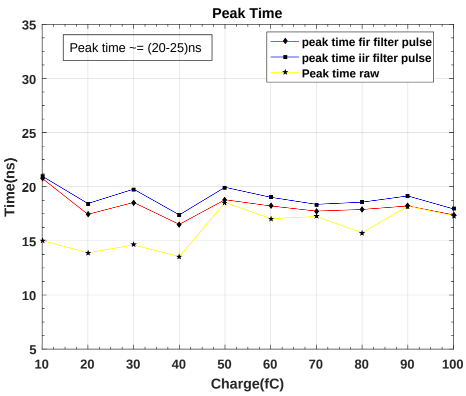
The Time measurement parameter like Rise time, Fall time,Pulse width and Peak Time are measured using an algorithm. The improvement in these parameter are observed as expected after processing through the designed FIR filter and IIR filter.An algorithm code is developed in Matlab for measuring these parameters according to their definition mention above in table1. FIR and IIR filter is used to reduce the noise of the signal using matlab signal processing tool.Time measurement results of the filtered pulse are found more stable, sustained and much tend towards the expected values as compare to raw pulse measurement.The improved Result shown in the given figure26. [?].



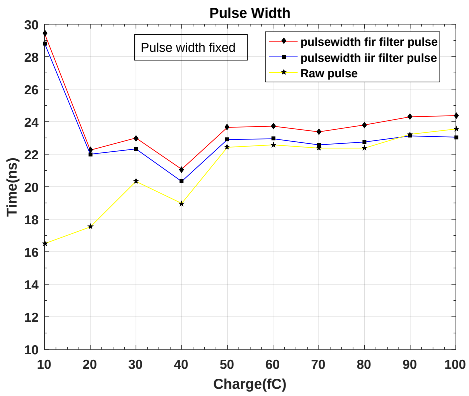
(a) Rise Time



(b) Fall Time



(c) Peak Time



(d) Pulse width

Figure 26: Time Measurement of filtered and Raw signal

12 Conclusion

The signal to noise ratio is improved significantly by designed and suitably choosen FIR and IIR filters. Suitably choosen filters means the deciding the cuttoff frequency and order of the filters. The cutt-off frequency is determined using fast fourier Transform (FFT) methode Fig 4. These operations include both baseline noise reduction and main signal pulse noise removal, de-noising. Noise reduction by my method leads to better the signal to noise ratio of real time pulse. Measurement of parameters like Rise time, peaking time and pulse width are improved with these noise reduction methods. The improvement in these parameters will ultimately leads to better muon track reconstruction by offline correction.

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13 Appendix

13.1 Charge Calibration using Simpson and trapezoidal method

The RPC pulse is mimicked with the help of RC circuit and function generator. A certain amount of voltgae is applied to the RC circuit for a particular Charge.This Charge value further need to be confirmed. I have used the Trapezoidal and simpson methode for the Charge Calibration. The area of pulse is calculated by these methodes and devided by the 50 to convert the volatge into the current.

$$Charge(Q) = \frac{1}{50}\int_a^b V(t)dt \tag{9}$$