# EMBEDDED DOPPLER ULTRASOUND SIGNAL PROCESSING USING FIELD PROGRAMMABLE GATE ARRAYS

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*Abstract*- **Doppler ultrasound is an important technique for non-invasively detecting and measuring the velocity of moving structures, and particularly blood, within the body. Most Doppler ultrasound systems are designed using fixed-function hardware to meet computational needs for flow and direction detection of blood within area being investigated. In this paper, we present a system where all Doppler-flow imaging is supported on a programmable platform. Using Matlab software, Xilinx Development Kit and System generator, we design a floating point precision system and a fixed point one to be implemented on FPGA . Fixed point signal processing has advantages of better hardware implementation and less cost over the use of floating point implementation. We will also show the analogy between designs of each block of Doppler flow system in floating point and fixed one and how it affects system output accuracy.**

## I. INTRODUCTION

Pulsed Doppler ultrasound is a technique for measuring the velocity of blood in a small sample volume. The location of the Doppler sample volume is illustrated by a cursor overlaid on the B-mode image. To provide a localized velocity measurement, the instrument transmits a pulse that is 6 wavelengths to 40wavelengths long - depending on the desired length of the sample volume. The received signal is gated so that the time elapsed between the transmission of the pulse and the opening of the gate determines the depth of the velocity measurement, i.e., the position of the sample volume. The Doppler signal is processed by a Fourier spectrum analyzer, which performs a Fourier transform on the Doppler signal at intervals of approximately 10 ms. The amplitudes of the resulting spectra are encoded as brightness and these are plotted as a function of time (horizontal axis) and frequency shift (vertical axis) to provide a two-dimensional spectral display. With this technique, a range of blood velocities in the sample volume will produce a corresponding range of frequency shifts on the spectral display. Recent ultrasound machines have some built-in flexibility in core ultrasound processing through the use of digital signal processors (DSP) and reconfigurable hardware, such as field programmable gate arrays (FPGAs). FPGA provide the ability to accelerate the arithmetic operations via parallel processing, making much better suited for medical imaging applications than solutions that rely only on DSP. Regular system upgrades, bug fixes need a flexible architecture which is provided by FPGA that they are reprogrammable, flexible, reusable since any upgrades involves merely a change of FPGA bit stream. This gives FPGA a major advantage, over for instance application specific integrated circuit(Asics).Our goal is to support all the core doppler ultrasound processing in software on a programmable system. We will begin by designing doppler built up blocks using matlab package, transform the system operations to fixed point accuracy in order to design FPGA, we use Xilinx development tools to design system blocks . we use system generator tool to allow us to use Xilinx blockset from within matlab work area so we have a whole one integrated design and after simulation verification we use this tool to create HDL that can be downloaded to a FPGA.

We will show difference in system response due to using floating point arithmetic versus fixed point one with different number of bit representations. Fixed point allows better hardware implementation, simpler circuits , small chip size, lower power consumption and over all lower unit cost which very important for competing PC-based Doppler ultrasound systems.

## II. Doppler ultrasound system

The high level block diagram of a Doppler ultrasound system is shown in fig. 1. The transducer is excited with bursts of pulses, these burst of ultrasound travels into the body, where it is doppler shifted by moving structures along the sound path. Returning echoes from both stationary and moving targets are received by the same transducer. This process is then repeated for the next burst of ultrasound. Returned signals from the probe are amplified by receiver amplifier where signals from stationary targets and small signals from moving blood are amplified equally. Returned signal which is a real one then pass through Hilbert transform filter to be transformed to a positive-frequency complex signal by simply generating a phase quadrature component to serve as imaginary part of complex signal and original signal represents the real part of complex signal. Hilbert filter apply a phase quadrature demodulation to preserve the real(in phase" I") and imaginary(phase shifted "Q") doppler difference components so as to provide doppler directional component. The under-sampling and demodulation block undersamples the output from the hilbert transform filter. The sampling duration together with the transmitted pulse length, sets the range over which velocity information is gathered. The output from the under-sampling and demodulation contains not only doppler shift frequencies but also high amplitude low frequencies resulting from moving structures and wall motion, so the output is filtered through a clutter filter(wall motion filter)to remove these signal which is usually variable over a range of 100 - 800 hertz(Hz) and with amplitude 40 – 60 decibels (db) higher than scattered signals from blood. The filtered in phase(I) and phase shifted(Q) doppler signal is then introduced to a complex Fast Fourier Transform(FFT) to draw a spectrum of doppler velocities.

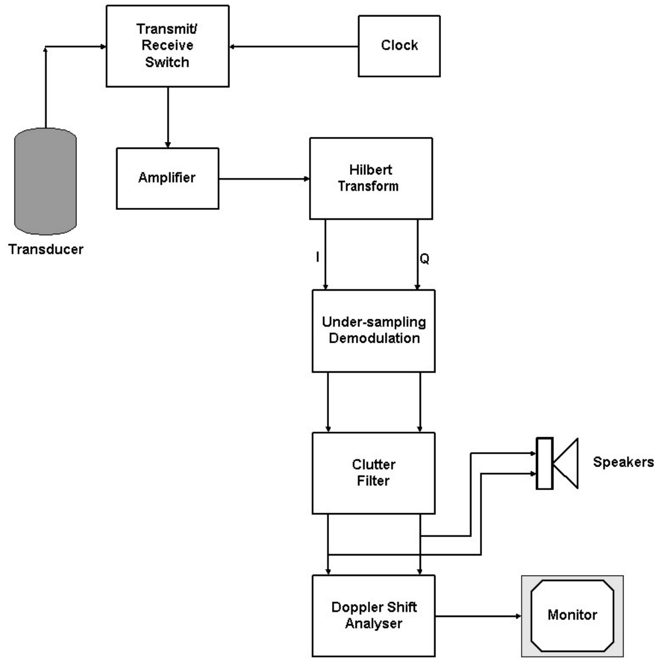


Fig.1.Doppler Ultrasound Block Diagram

## III. Doppler Flow Processing

### *A. Algorithms*

We will assume transmitted signal to probe with center frequency 3MHz, so the received amplified RF signal will have frequency of 3MHz plus for example a 2KHz doppler shift signal. The first main computing block is ***Hilbert Transform Filter***, hilbert transform helps us to relate the *I* and *Q* components of doppler signal needed for direction detection and a special class of causal signals called analytic signal which are especially important in simulation. We can describe hilbert transform as shown in fig. 2.

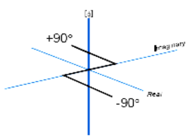


Fig. 2. Hilbert Transform

*All negative frequencies of a signal get a +90° phase shift and all positive frequencies get -90° phase shift.* For this reason hilbert transform is called a quadrature filter, It only affects the phase of the signal. It has no effect on the amplitude at all. Describing Fourier analysis of hilbert filter in (1) its net effect is to double the spectral magnitudes and then chop-off all negative components.

$$G_{+(f)} \begin{cases} 2G(f) & \text{for } f > 0 \\ G(f) & \text{for } f = 0 \\ 0 & \text{for } f < 0 \end{cases}$$

Where $G_{+(f)}$ is the analytical signal complex signal created by taking a signal and then adding in quadrature its hilbert transform. This analytic signal from hilbert filter has a spectrum that exists only in the positive frequency domain, which is our goal.
The real and imaginary part of analytic signal then pass through second block **under-sampling and demodulation** block**,** where received signals is sampled and demodulated from carrier frequency which is 3MHz in our assumption. we need to set sampling frequency for input signal. Since it take sound waves about 13 microseconds (µsecs) to travel 1 cm. deep in tissue and back again to probe (sound velocity in tissue is 1540 m/sec). For a sample volume located 10 c.m. deep, a pulse repetition period of 130µsec is needed to acquire data from this depth. The best appropriate sampling frequency for input signal is pulse repetition frequency (P.R.F) which is approximately 8KHz, taking into account the maximum doppler shift frequency that can be detected is half P.R.F which is 4KHz band of doppler shift that can be measured. The sampled data contains not only doppler signal but also signals from stationary targets, wall motion. We are not breaking the Nyquist criterion because Nyquist actually said the sampling rate must be at least double the signal's bandwidth, not the signal's highest frequency component.
The purpose of third block ***Clutter Filter*** is to filter input signal from high amplitude (40 -60 db higher than scattered signals from blood) low frequency(variable from 100 – 800 Hz)due to wall movement. This is done by applying a high pass filter (HPF), with cut-off frequency 100Hz and sampling frequency P.R.F.
We design this HPF using Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filter to show difference between them. FIR filters offer several advantages over IIR filters: Completely constant group delay throughout the frequency spectrum and complete stability at all frequencies regardless of the size of the filter but come with some disadvantages as well that the frequency response is not as easily defined as it is with IIR filters and the number of states(filter tabs) required to meet a frequency specification may be far larger than that required for IIR filters.
Once the data from each sample volume are complete, filtered we have two doppler data components ; the in-phase (I) and phase shifted (Q) ready to be applied to the last block ***Fast Fourier Transform*** (FFT) to draw a spectrum showing frequencies detected, their amplitude and direction of blood flow within that sample volume.

*B. Algorithm Implementation*

We use Matlab simulink tool to create a GUI of each block design. Hilbert Transform Filter (HTF) is designed as shown in fig. 3. First we define the returned signal from the transducer by generating a sine wave of 3MHz + 2KHz frequency and unity amplitude sampled at 10MHz. we create HFT by defining a complex and nonlinear-phase equiripple FIR filter; with frequency and amplitude response as stated in [1] ,and we show output filtered signal using a FFT scope. We also design a Xilinx n-tap multiply accumulator (MAC) fixed point FIR filter with same frequency response. We introduce gateway block which converts signal from floating point accuracy to fixed point (defining number of bits and fractional length) to filtered with designed fixed point FIR. Adding *System Generator* block to provide control of system and simulation parameters, and is used to invoke the code generator. Every Simulink model containing any element from the Xilinx Blockset must contain at least one System Generator block. Once a System Generator block is added to a model, it is possible to specify how code generation and simulation should be handled.

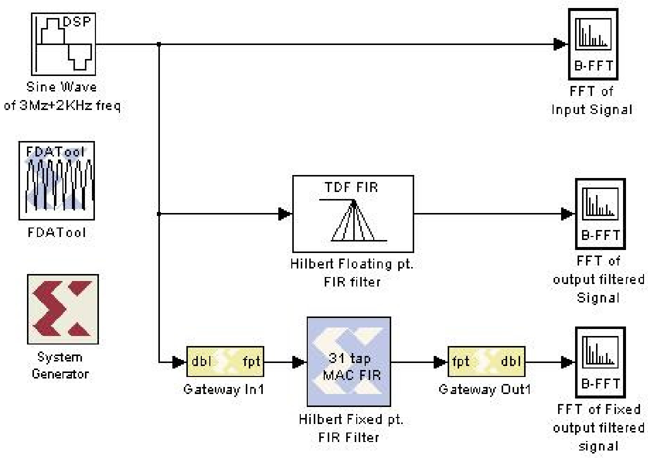


Fig. 3. Hilbert Transform Filter simulation Block

Designing clutter filter, we will apply it to real component of output hilbert filtered signal and imaginary part will undergo the same processing. The input signal will be sinusoidal with frequency 2KHz representing doppler shift band added to it undesired high amplitude low frequencies as shown in fig. 4. We will sample data with P.R.F which is 8KHz. We use Least Square FIR method, this filter design has ripple in pass band and stop band but mean least square error is minimized. We set cut-off frequency to 100Hz and choose low number of taps ("we will show the effect of number of tabs in its response). For IIR filter we design we choose Elliptic method with minimum number of tabs. We also use system generator to design fixed point MAC filter and we draw FFT of filtered data of each filter type for comparison.

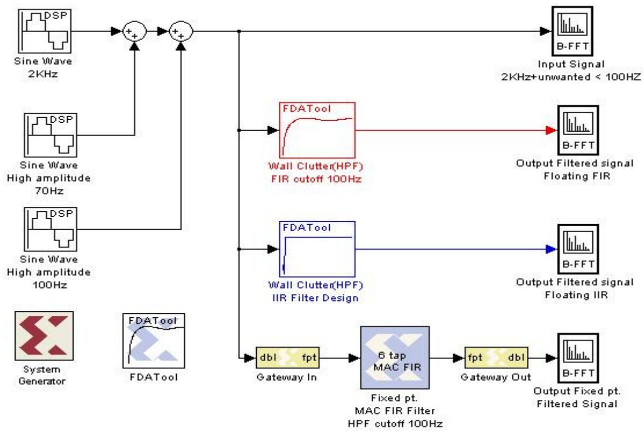


Fig.4. Clutter Filter Design

## IV. Results and Discussion

We design Hilbert transformation filter by defining a low pass filter with frequency response as in [1] and 5 taps. The frequency response of floating point filter versus the fixed one is shown in fig.5. using high bit word coefficient with appropriate fractional length makes frequency response of filter approximate the response of floating point; using 16 bit coefficient word length with 16 bit fractional length gives nearly the exact response of floating one.

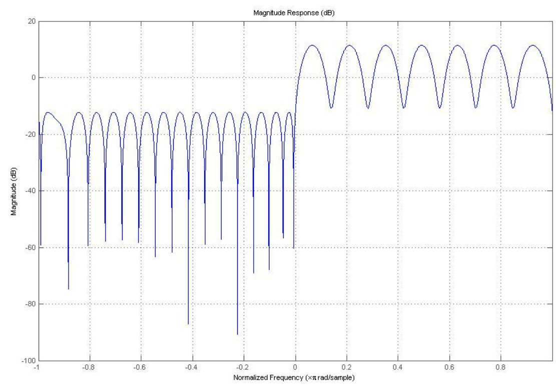


Fig.5. Hilbert filter frequency Response

Received signal of 3MHz with 2KHz doppler band frequencies is filtered with hilbert filter and output is shown in fig. 6. All negative frequencies are chopped off.

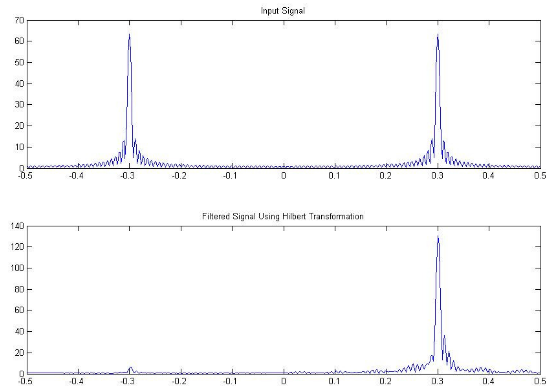


Fig.6. Output filtered signal from Hilbert filter

Working on real part of filtered signal from hilbert filter which has 2KHz band frequencies plus high amplitude low frequency (100Hz) of wall motion. We apply a high pass filter (HPF) of type least square, sampling frequency 8Khz and cutt-off frequency of 100Hz. In fig. 7. we show effect of changing number of taps in filter response, the higher the tap number the better the response. It is shown that floating point FIR with 31 taps has best response than ones with 5, 9 and 15 taps.

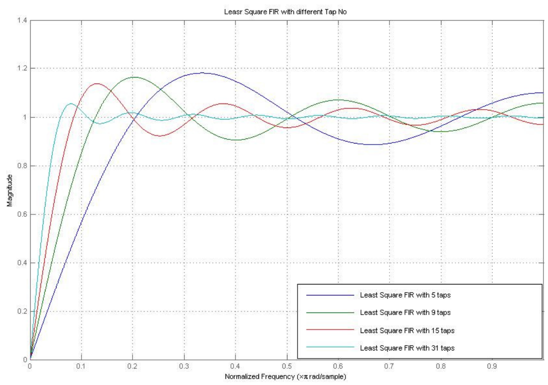


Fig.7. Clutter floating pt. FIR with different taps.

We design clutter filter as a fixed point one with different representations of coefficient word length 6 bits, 8 bits and 16 bit. Comparing the response with reference clutter filter response which is floating point is shown in fig.8.
It is clearly shown the fixed point filter with 16 bit word length nearly coincide with floating point response and other bit representation till 6 bits has no great deviation from them.

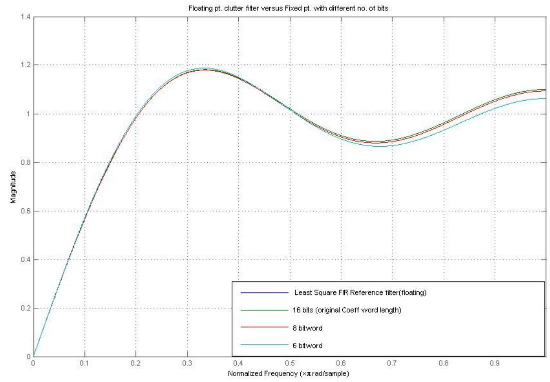


Fig.8.Clutter filter floating pt. VS fixed pt.

Appling a signal with wall motion artifacts and filtered it with clutter filter is shown in fig.9. where these artifacts are completely filtered out successfully with both floating and fixed point design.

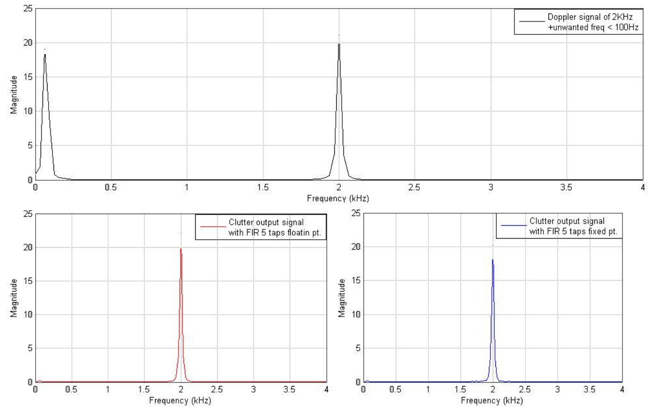


Fig.9. Output filtered signal from clutter filter.

## V. Conclusion

A fully programmable Doppler ultrasound system has been developed and demonstrated where doppler flow processing is supported in software using Xilinx development tool as compared to hardwired based designs used in traditional ultrasound system. A doppler system designed on a FPGA is more flexible and compact, by using nowadays high end processors, the system can be scaled from a low-cost to high performance ultrasound system. We believe this is a step towards a flexible and easy updated doppler system that can benefit from rapid development and improvements in programmable processors and memory technology.

## References

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