Chirp Identification Using STFT Technique and Its Implementation on FPGA

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Abstract—Modern radars use frequency and phase modulated signals to spread their spectrum to improve the processing gain. The Identification of the modulation format of a detected signal, the intermediate step between signal detection and demodulation, is a major task of an intelligent receiver, with various civilian and military applications. Obviously, with no knowledge of the transmitted data and many unknown parameters at the receiver, such as the signal power, carrier frequency and phase offsets, timing information, etc., blind identification of the modulation is a difficult task. This becomes even more challenging in real-world scenarios. An algorithm is designed to identify the LFM Chirp signal using an STFT Process and calculating the chirp rate and the frequency from the spectrum.

Keywords—Radars, Chirp signals, ADC, LFM, STFT, Signal Identification

I. INTRODUCTION

Modern radars use frequency modulated signals and phase modulated signals to spread their spectrum to improve the processing gain. There are several approaches of using the phase modulation. The phase change can be π, $\pi/2$, $\pi/4$, and so forth. When the phase change is π, it is referred to as bi-phase shift keying (BPSK). Communication signals use various types of phase modulations, this paper is mainly for LFM Chirp because it is more popular in radar applications. It is important to detect the existence of a LFM Chirp signal for an electronic warfare (EW) receiver. The detection of a LFM Chirp signal can help identify the radar type, which is important information. If more information can be obtained from the signal, the identification approach can be easier. In this study the requirements are to detect the existence of a Chirp signal and find the frequency and its chirp rate. The chip time is the instantaneous rate of change of the frequency.

The Identification of the modulation format of a detected signal, the intermediate step between signal detection and SSDE modulation, is a major task of an intelligent receiver, with various civilian and military applications. Obviously, with no knowledge of the transmitted data and many unknown parameters at the receiver, such as the signal power, carrier frequency and phase offsets, timing information, etc., blind identification of the modulation is a difficult task. This becomes even more challenging in real-world scenarios. So a new method has been developed to identify the LFM Chirp signal. This is achieved using an STFT Process and by calculating the chirp rate and frequency from the spectrum.

II. DIGITAL RECEIVER

Radio receivers that perform the analog-to-digital conversion process close to the antenna and do most of the signal processing in the digital domain are known as digital receivers. Digital receivers, often called software radios, place a high performance burden on the ADC, but allow a good deal of flexibility in post detection signal processing. EW receiver parameters of interest include sensitivity, dynamic range, resolution, simultaneous signal capability, complexity, and cost.

The block diagram of a wideband digital EW receiver is shown below. The input signal from the antenna is first amplified by a wideband LNA. Most digital EW receivers use frequency conversion before digitizing the signal. That is, the signal is first down converted in frequency, and then digitized by an ADC. The digital signal is then processed by a spectrum analyzer that extracts the frequency information. Using this frequency information, the signal is sorted, and a parameter encoder then forms a pulse descriptor word (PDW). For LPI CW emitters, the PDW contains the center frequency fc, the signal coding details such as the modulation period and bandwidth (FMCW),

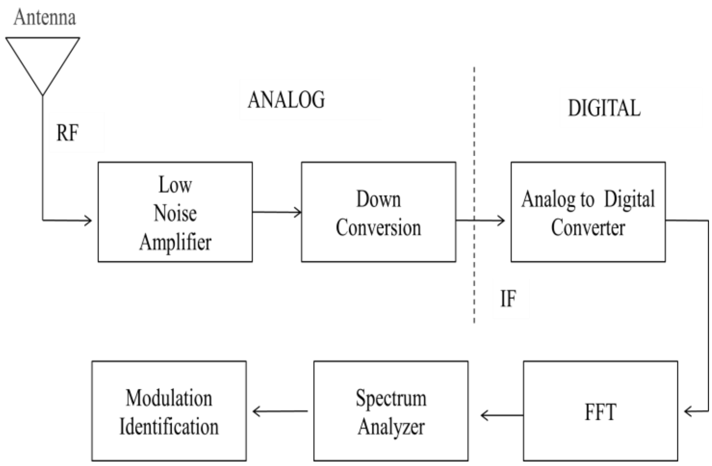


Fig.1: Digital receiver block diagram

The advantage of this approach is that by driving the mixer with a frequency-agile LO, the frequency of the desired signal or channel is converted to a fixed frequency. Once converted to a fixed IF, it can be processed by highly selective narrowband filtering (e.g., using surface-acoustic wave devices or high-temperature superconductors). Also, all subsequent frequency translations can be done using fixed-frequency LOs. Also performed is signal amplification using fixed gain LNAs (at RF), and variable gain amplifiers (at IF). The distribution of gain across the IF stage prevents instabilities in the amplifiers, and reduces the chance of saturation.

A direct conversion (homodyne) down conversion can also be used. This two-channel approach uses only a single local oscillator, and translates the signal of interest to zero frequency (zero-IF). Due to the elimination of the IF stages, all signal conditioning must be performed either at RF or baseband. The direct conversion approach offers a higher degree of integration at the front end with fewer components, allowing most of them to be monolithically fabricated on a single chip. The direct conversion receiver performance still does not match the IF receiver, due to filter saturation and distortion caused by the dc offsets and self mixing at the mixer inputs. To take advantage of both receiver topologies, a low-IF receiver is now an alternative (a few hundred kilohertz). The low-IF receiver has a high degree of filter integration, and is also insensitive to dc offsets and LO-to-RF crosstalk. In all cases, the signal is down converted to a baseband frequency that depends on the analog-to-digital converter technology that is available.

III. LINEAR FREQUENCY MODULATION

Frequency modulation is a type of modulationwhere the frequency of the carrier is varied in accordance with the modulating signal. The amplitude of the carrier remains constant. Since the amplitude is kept constant, FM modulation is a low-noise process and provides a high quality modulation technique.

A chirp is a signal in which the frequency increases ('up-chirp') or decreases ('down-chirp') with time. Chirp modulation, or linear frequency modulation employs sinusoidal waveforms whose instantaneous frequency increases or decreases linearly over time. These waveforms are commonly referred to as linear chirps or simply chirps.

The rate at which the frequency changes is called the *chirp rate*. In binary chirp modulation, binary data is transmitted by mapping the bits into chirps of opposite chirp rates. For instance, over one bit period "1" is assigned a chirp with positive rate a and "0" a chirp with negative rate $-a$. Chirps have been heavily used in radar applications and as a result advanced sources for transmission and matched filters for reception of linear chirps are available. In a *linear* chirp, the instantaneous frequency $f(t)$ varies linearly with time:

$f(t) = f_0 + k$

where f_0 is the starting frequency (at time $t = 0$), and k is the rate of frequency increase or chirp rate. The corresponding time-domain function for a sinusoidal linear chirp is:

$$x(t) = \sin\left[2\pi \int_0^t f(t')\,dt'\right] = \sin\left[2\pi \int_0^t (f_0 + kt')\,dt'\right] = \sin\left[2\pi(f_0 + \frac{k}{2}t\right.$$

Classification of chirp signals:

a) UP Chirp
b) DOWN Chirp
c) UP-DOWN Chirp

A. Up Chirp

An Up chirp signal is a frequency modulated signal whose frequency increases linearly with respect to time as shown below. The up chirp signal can be generated using the formula given below

*x = sin(2*pi*((f0.*(n/Fs)) + ((k./2).*((n/Fs).^2))))*
where k = (f1-f0)./t1
fo=Starting frequency
f1= ending frequency
t=time period
t1=end time
Fs=Sampling frequency.

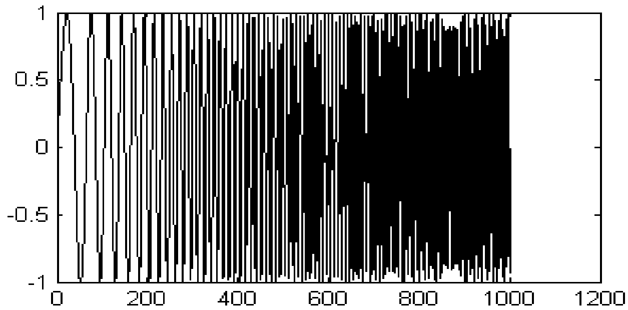


Fig.2: Up chirp signal

B. Down Chirp

A Down chirp signal is a frequency modulated signal whose frequency decreases linearly with respect to time as shown in the figure below

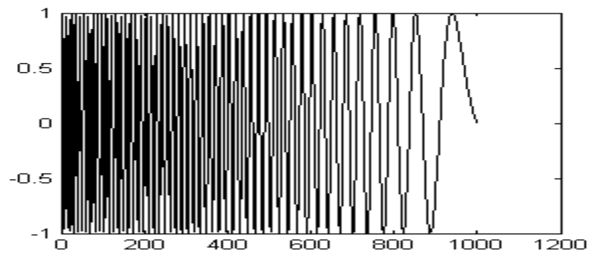


Fig.3: Down chirp signal

The Down chirp signal can be generated using the formula given below

*x = sin(2*pi*((f0.*(n/Fs)) + ((k./2).*((n/Fs).^2))))*
where
k = (f1-f0)./t1 " chirp Rate"
f0=Starting frequency
f1= ending frequency
t=time
t1=end time
Fs=Sampling frequency
In the Down chirp signal the starting frequency is high and decreases linearly with respect to time

C. Up-Down Chirp

An up down chirp signal is a combination of both up chirp signal and down chirp signal. It is a frequency modulated signal whose frequency increases linearly with respect to time up to half of the time and decreases linearly from the half time to the end time is called as an up-down chirp signal as shown in the figure below

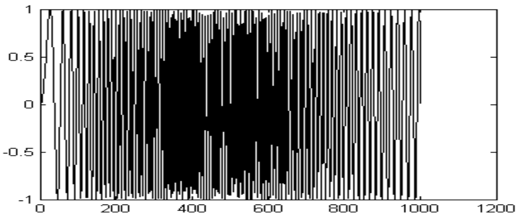


Fig.4: Up- Down chirp signal

The up down chirp signal can be generated using the formula given below

Sub pw=PW/2
t=0:1e5/Fs:1;
t2=(t+t1)/2;
k = (f1-f0)./t2;
*x = sin(2*pi*((f0.*t) + ((k./2).*((t).^2))));*
M=(f1-f0)./t2;
*z = sin(2*pi*((f0.*t1) + ((M./2).*((t1).^2))));*
y=[x y];

IV. SHORT TIME FOURIER TRANSFORM(STFT)

Short time Fourier Transform is a process, which involves repetitiveFFT operations for small intervals of time**.** The given time period is divided into 'n' number of slots where 'n' is the no of samples/ No of Point FFT, Therefore the time resolution for the calculation of FFT is' T/n' where T is the Total time period of the received signal.

STFT is used to know the phase and the frequency at different instants of time so that the signal modulation type is possible whereas in FFT the frequency and phase calculation is not possible as it contains several peaks in a particular bin, but the frequency or phase will be calculated at every bin number as a result of which several frequencies and phases are missed using an fat, This is the main reason why we have designed STFT process for time to frequency conversion of a signal

V. ALGORITHM FOR THE IDENTIFICATION OF THE LFM CHIRP SIGNALS

Detection and interception of LPI signals requires sophisticated receivers that use time domain to frequency domain conversion process of a signal, correlation techniques and algorithms to overcome the processing advantage of the LPI radar. These signal processing algorithms require a large amount of computing speed and memory. Managing processing speed is not a problem with the current digital capabilities, but carrying enormous amounts of data is still problematic. Increasing the sensitivity of the receiver allows for detecting side lobes of the emitter, but at the same time obligates the receiver to process a significantly large number of signals. In addition to these challenges, new signal detection and feature extraction systems are needed to effectively analyze these new waveforms in a complex signal environment. Time-frequency data analysis can be performed using complex instrumentation through computer analysis. The received signal is an analog signal, converted into a digital signal using an ADC. The digital signal is given to the FPGA as an input. The algorithm is developed with the digital signal as the input in the time domain. The Algorithm for the modulation identification is given below

Algorithm

- The digital time domain signal from the ADC is the input to the modulation identification process, where the total number of samples in the signal are calculated as
  Total no of samples (m) = Pulse width * Sampling frequency used by the ADC
- The time domain signal will be converted into frequency domain using a FFT process.
- Using an FFT a single frequency can be found, but if the signal is a frequency modulated signal then it has multiple frequencies which cannot be identified by a single FFT process thus a new technique called STFT is introduced.
- STFT is a process of calculating the FFT for multiple times on the same signal, i.e the time domain signal is divided into n number of frames, where FFT is performed for each frame.
- The total number of frames in the signal can be calculated as follows
  No of frames (n) = total number of samples (m) / maximum point in the FFT (N)
  Therefore each frame consists of N samples
- The amplitudes and its corresponding index from the FFT are given to the peak detector, the peak detector will find the max amplitude and its corresponding output index will be considered as the bin number of that frame
- The bin numbers of all frames are taken and compared.
- If bin numbers of all the frames are the same then the signal is said to be a phase modulated signal
- If the bin numbers are varying from frame to frame then the signal is said to be frequency modulated signal

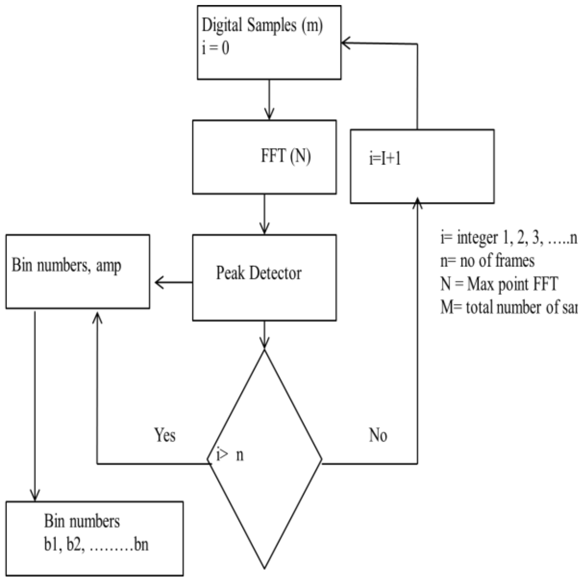


Fig.5: STFT Process

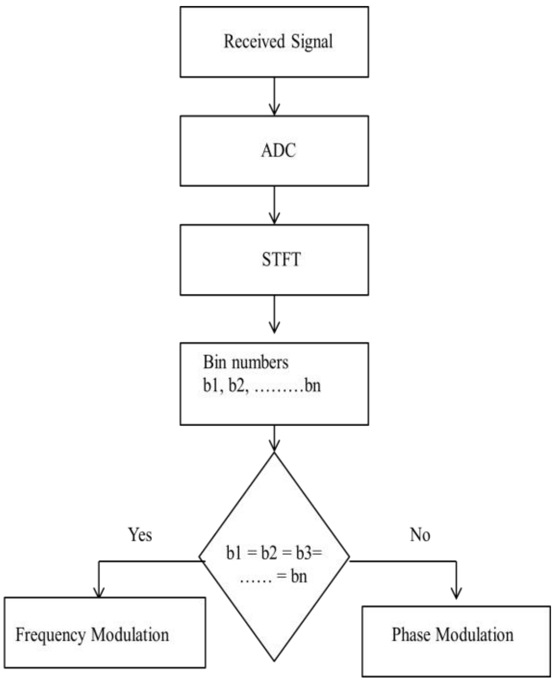


Fig.6: Flowchart for modulation identification

This algorithm is the first stage of modulation identification, where it can be decided whether the signal is a frequency modulated signal or phase modulated signal, further the signals are classified from the algorithm given below.

If the received signal is frequency modulated signal, then there are three different types of linear frequency modulation techniques used by the LPI RADAR they are:

- UP Chirp
- Down Chirp
- UP-DOWN Chirp

ALGORITHM FOR CLASSIFICATION OF CHIRP SIGNAL

Algorithm

- The Signal is converted from time Domain to frequency domain using the FFT Process
- The FFT Process is performed multiple times until all the samples are processed (This process is called STFT)
- The output index of the FFT at which the magnitude peak occurs is considered as frequency bin number
- Frequency is calculated using the frequency bin number using (F=bin number *(Fs/N))
- The frequencies of all the frames are compared
- If the frequencies are increasing linearly from frame to frame then the signal is said to be an up chirp signal
- If the frequencies are decreasing linearly from frame to frame then the signal is said to be an down chirp modulated signal
- If the frequencies are increasing linearly from frame to frame up to half of the total number of frames and then decrease linearly up to the last frame then the signal is said to be an up down chirp

Flow chart of the chirp classification

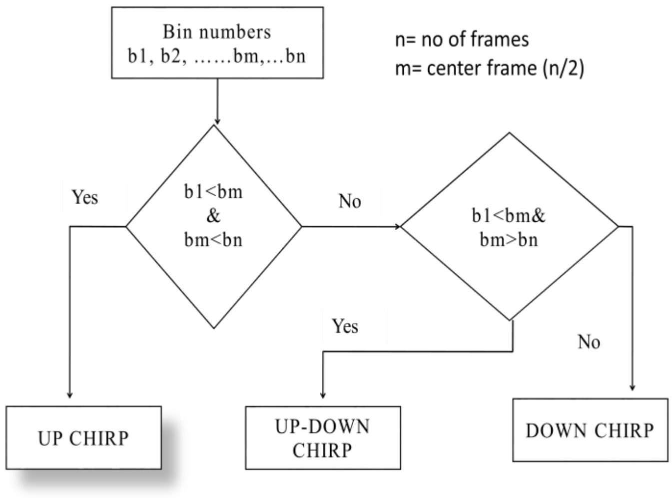
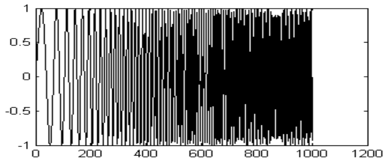


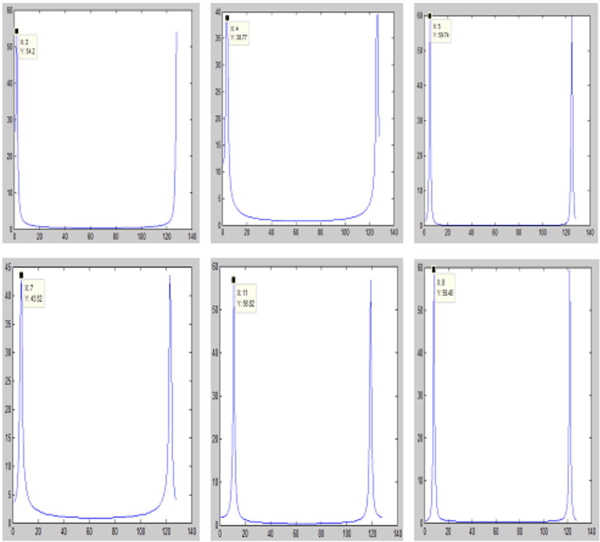
Fig.7: Flow chart for chirp identification

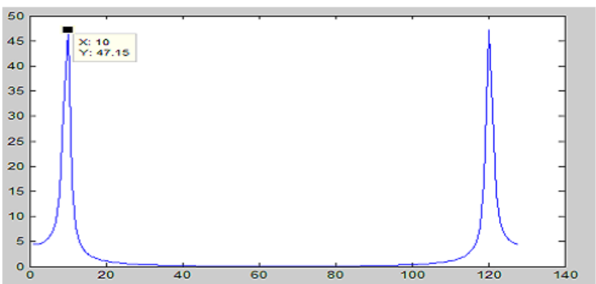
VI. RESULTS AND DISCUSSION

The algorithm is first modelled in Matlab to estimate the Hardware Resources. A sample LFM Chirp signal is generated in Matlab and applied as an input to the design. The Matlab results for the algorithm are shown below.

```
enter the type of modulation: "0" for chirp modulation
THE SIGNAL IS A CHIRP SIGNAL.
enter the type of modulation "00" for up chirp; "01" for down chirp ; 02 for up down chirp: 00
enter the pulse repetition interval in us(1-10ms): 1
enter the pulse width in us(1-10us) : 10
 enter the pulse amplitude in Volts (1.8V-2.2V): 2
enter the starting frequency in Mhz (140Mhz-160MHz):140
enter the ending frequencyin Mhz(160Mhz-180Mhz):180
*****The Received signal is an up chirp*****

frequency =

  1.0e+008 *

  Columns 1 through 12

    1.4580    1.4580    1.4580    1.4580    1.5100    1.5100    1.5100    1.5100    1.5100    1.5100    1.5100    1.5621

  Columns 13 through 24

    1.5621    1.5621    1.5621    1.5621    1.5621    1.5621    1.6142    1.6142    1.6142    1.6142    1.6142    1.6142

  Columns 25 through 36

    1.6663    1.6663    1.6663    1.6663    1.6663    1.6663    1.6663    1.7183    1.7183    1.7183    1.7183    1.7183

  Columns 37 through 48

    1.7183    1.7183    1.7704    1.7704    1.7704    1.7704    1.7704    1.7704    1.7704    1.8225    1.8225    1.8225

  Columns 49 through 52

    1.8225    1.8225    1.8225    1.8225

starting_frequency =

   145796875

Ending_frequency =

  1.8225e+008
```

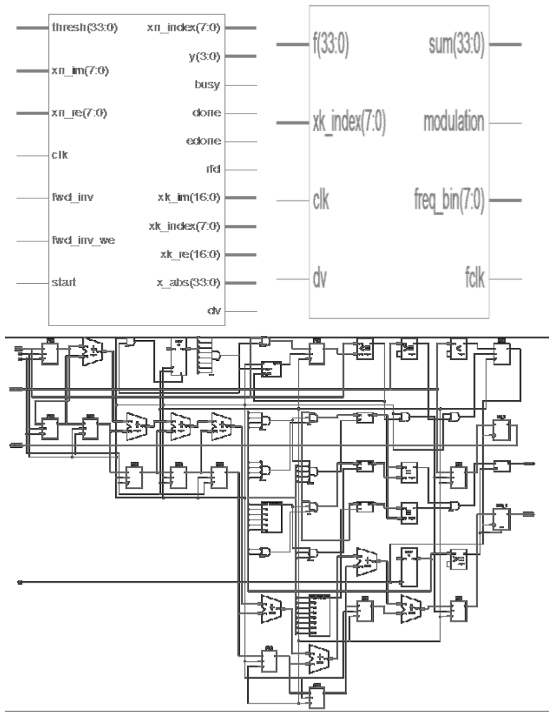






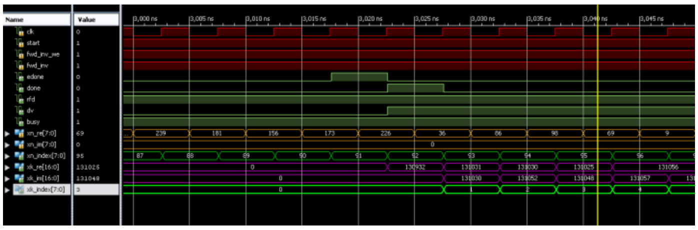
The Matlab generated Signal is given as input an Analog to digital converter and thus the obtained digital samples are given to the FPGA as an input. The algorithm consists of fifo , STFT operations to be performed and the obtained outputs are given to the LFM Chirp analysis block which classifies the type of LFM Chirp signal . The algorithm is designed using VHDL language , Synthesized in Xilinx 13.1.i ,Simulated in Xilinx ISIM 13.1 i. The synthesis and simulation results are show below

RTL Schematic:

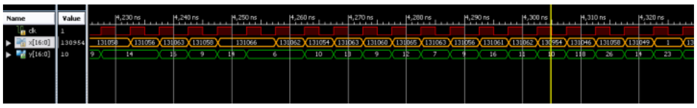


Simulation Results

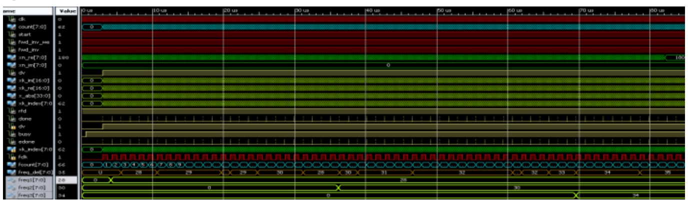
FFT CORE:



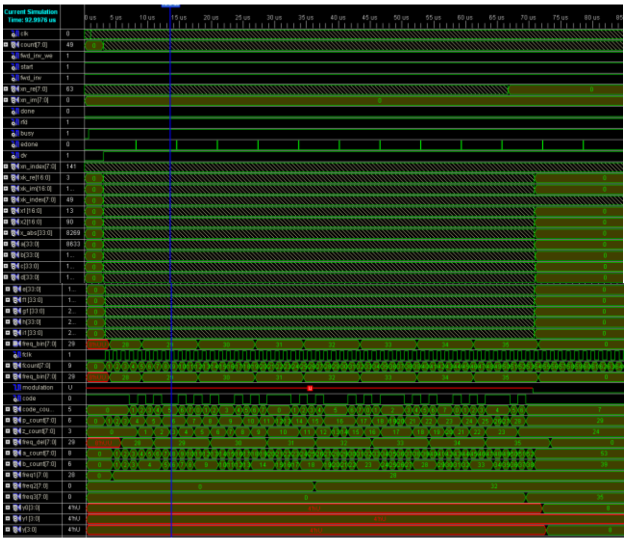
TWOS COMPLIMENT SIMULATION RESULTS:



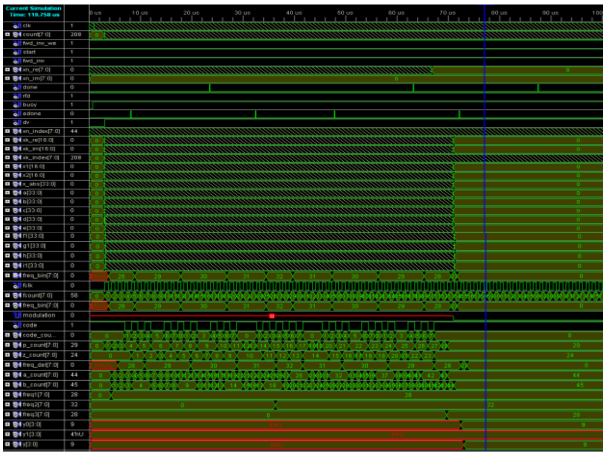
CHIRP



UPCHIRP



DOWNCHIRP



UPDOWN CHIRP

VII. CONCLUSIONS

This paper describes an algorithm for the identification of LFM Chirp signal which is based on the time to frequency analysis.

DSP processors are used for implementing many of the DSP applications. Although DSP processors are programmable through software, the DSP processor hardware architecture is not flexible. Therefore, DSP processors are limited by fixed hardware architecture such as bus performance bottlenecks, a fixed number of multiply accumulate (MAC) blocks, fixed memory, fixed hardware accelerator blocks, and fixed data widths. The DSP processor's fixed hardware architecture is not suitable for certain applications that might require customized DSP functions implementations.

FPGAs provide a reconfigurable solution for implementing DSP applications as well as higher DSP throughput and raw data processing power than DSP processors. Since FPGAs can be reconfigured in hardware, FPGAs offer complete hardware customization while implementing various DSP applications. Therefore, DSP systems implemented in FPGAs can have customized architecture, customized bus structure, customized memory, and customized hardware accelerator blocks.

So keeping in view of the advantages of using FPGAs as compared to other customized devices this work can further be Increased to the designing & implementation of a LFM Chirp Identification algorithm. The generosity of the design facilitates not only the rapid Signal Identification, but moreover, it explores the performance of different implementations in order to obtain the most suitable solution for a particular Electronic Warfare system. Some of the generic parameters of LFM Chirp Identification are, frequency (bin number), amplitude, chip rate, threshold, min/max ratio, sum of the ratios,

The Above design is implemented on Xilinx Virtex-5 SX240t an virtex-5 Fx200t, an 8 bit ADC is used to convert the analog signal into a digital signal the IC used is ADC08D1520, Using these hardware the design implemented on the FPGA and the algorithm is verified and has given Accurate and efficient results.

The above algorithm is an efficient algorithm, which is technologically feasible for the fast interception of signals without any parameter extraction (Pulse width, Pulse Repetition Interval, direction of arrival). It outperforms the previous approaches in terms of speed, sensitivity, accuracy in the identification of a LFM Chirp signal.

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