Designing and testing of an Interpolation and Decimation circuitry for Communication modules

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Abstract— this work focuses on designing and implementing a resampling circuitry for a telecommunication Networks. In telecommunication networks, there occur many situations where unwanted signals (noise) creates problem for modulation and demodulation. In this paper the work done is to design a system to avoid up this issue to an extent. The first part of this paper deals with the development of the model of resampling device in MATLAB Simulink. The resampling of the signals takes place at this stage. Next, the Verilog description of the design is implemented using Xilinx ISE and verifies the correctness of the design. The resampled signals are compared with the original signal in the results.

Index Terms— demodulation, modulation, MATLAB, networks, resampling, telecommunicatons, Verilog, Xilinx.

1 Introduction

THE Digital Signal Processing is a domain that exhibits faster growth rate compared to other domains. The growth in this domain is due to the fact that the areas of microelectronics and digital computing have had a tremendous growth in the recent past. DSP means the different techniques that are conducted on the digital signals to improve the same. It includes numerical manipulation of signals and data in digital form. There are basically two types of signals- analog and digital signals. Quantized signals are called digital signals while continuous signals are called analog signals. In order to obtain digital signal analog signals undergo a process called sampling.

There are a wide range of applications for DSP which includes T.V broadcasting, visual broadcasting, image broadcasting and numerous other day to day applications. DSP is used by Military agencies and various other related fields for their day to day applications. Digital signal processing also is being widely used in the field of telecommunication technologies, possess a great challenge. In telecommunication, the transferring signals will be mostly non trivial or most of the signals contains information that are too important that digital signal processing becomes impossible. Resampling of a signal poses great challenges while considering modulation and demodulation.

2 Problem statements

2.1 Literature Review

Existing researches are mainly concentrated on the application of oversampling to avoid the problems in modulation and demodulation.

In telecommunication system, the transmission of signal is the most important step. During the transmission many processes will be done. At this phase the transmitted signal, which is generally a band pass signal will get sampled at rate which is higher than the Nyquist rate [1].The signal to be digitized is of radio frequency range. Hence the sampling rate will be very high and thus the power dissipation. As a result in the device 3 blocks are used, an analog RF processor, digital IF stage and a baseband processor.

In the analog RF processor the radio frequency is converted to a signal with lesser frequency which is called intermediate frequency. In the second stage that is digital IF stage this signal is sampled with a frequency that is twice the intermediate frequency. Due to the factor of band pass sampling the signal is digitized at a rate which is much higher than that of Nyquist rate. The produced signal is bandlimited IF signal sampled at a high frequency rate.

This signal is mixed with other signal in local oscillator to produce low frequency signal.understanding these guidelines before submitting their manuscript.

2.2 CORDIC Algorithm

CORDIC algorithm is used in the intermediate frequency stage. CORDIC algorithm uses simple shift and add operation to obtain value of trigonometric functions using coordinate rotation. The iterative equation for radix-2 CORDIC algorithm for vector notation is given below.

$$x_{i+1} = x_i - \sigma_i y_i 2^{-i} \quad (1)$$
$$y_{i+1} = y_i + \sigma_i x_i 2^{-i} \quad (2)$$
$$z_{i+1} = z_i - \tan^{-1}(\sigma_i 2^{-i}) \quad (3)$$

The effects of oversampling are discussed in detail and the solution is to use a digital down converter which in effect will reduce the power consumption.

Pertsev L and Timoshenko A, introduced the relationship between ADC and DQPSK [2]. It is important to produce minimum power devices in today's world. But in fact power consumed by FPGA and ADC are the two factors needed to be considered. Thus ADC limits and sample rate resolution are discussed.

B. Lakshmi and A Agarwal, introduced a sample rate converter for radio receiver in FPGA [3]. The signal received to be received is radio signal and has high frequency range. Hence the sampling will be done at a rate than the Nyquist rate due to band limitation. The paper discusses the implementation of a sample rate converter.

Dodgson, N.A introduced a quadratic interpolation for resampling [4]. There are different varieties of interpolation.

Author describes a separate variety of interpolation called quadratic interpolation. Interpolation is the basic step of resampling.

3 Resampling of Signals

Since digital signal processing of signal causes problems in telecommunication field resampling is used to solve such problems.
The reason to use resampling over oversampling is the disadvantages of oversampling. The main disadvantage of oversampling is the frequency of the oversampling clock device. This is determined by the existing oversampling algorithm.
Resampling is the process in which interpolation and decimation are combined to change the sampling factor of an already digitized signal by a rational factor. By doing this process even though the sample rate changes the information is preserved.

4 Proposed System

4.1 Interpolation

IJSER Interpolation is the process of constructing new data points within the range of a discrete set of the known data points. Interpolation is also referred as the up sampling process.

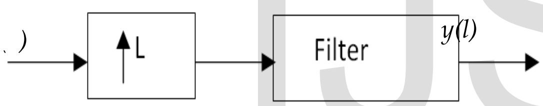

Figure 1. Interpolator

It is possible to distinguish the following from methods of interpolation
1) Linear interpolation
2) Bi-linear interpolation
3) Bi-cubic interpolation
4) Interpolation by Lagrange's polynomials
Proposed system uses interpolation by Lagrange's polynomials.
Thus the interpolator used is called Farrow interpolator.

Lagrange's polynomials describe a curve passing through N points of which only one is a *unity.*

$$L_{N-1}(x)=\sum_{i=1}^{N} y(i) \prod_{j\neq i}^{N} \frac{(x-xj)}{(xi-xj)} \quad (4).$$

For N=4, a cubical polynomial is obtained while applying Lagrange's equation.

$$L_{N-1}(x) = a_0+a_1.x+a_2.x^2+a_3.x^3= x.(x.(x.a_3+a_2)+a_1)+a_0 \quad (5)$$

The coefficients a_0,a_1,a_2,a_3 are known as the Lagrange's coefficients. These coefficients are calculated using discrete samples of the original signal.

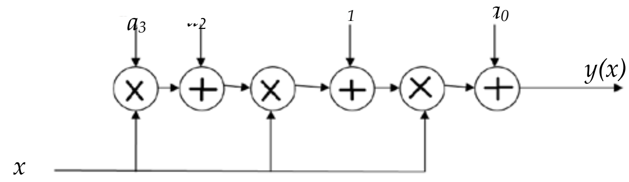

Figure 2. Diagram for implementing of Lagrange's polynomial

Figure 2 shows the implementation of Lagrange's polynomial. In the figure *x* is the original signal which is to be resampled. The same signal undergoes multiplications and addition to obtain the final result *y(x)*. This process is repeated until equation 5 is obtained.

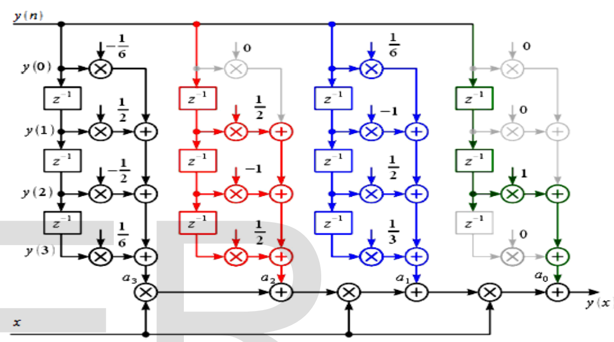

Figure 3. Calculation of Lagrange's coefficients

Figure 3 shows the calculation of Lagrange's coefficients. As earlier original signal is *x* and *y(n)* is the discrete values of the original signal.

4 Implementation

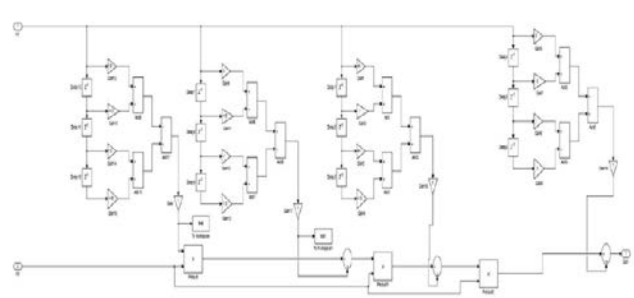

Figure 4. Resampling device in MATLAB Simulink.

Figure 4 is the MATLAB implementation of the resampling device. This structure acts as the subsystem for the main system. Subsystem is called as the Farrow interpolator. The subsystem is the major part in the implementation. It calculates the value of Lagrange's coefficients with the help of original signal and discrete value of original signal.

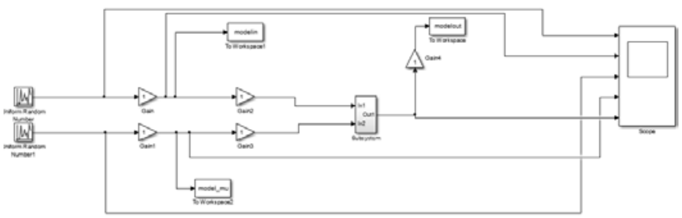

Figure 5. A block diagram for simulation of resampling

Figure 5 shows the main system of the implementation. The main system consists of several other blocks other than the subsystem. Main system consists of oscilloscope, gain blocks, buffers etc. Oscilloscope is the digital scope used for the display of signal.

5 Results

The output waveforms can be verified using oscilloscope. The resampling of a random signal is obtained by giving uniform random numbers.

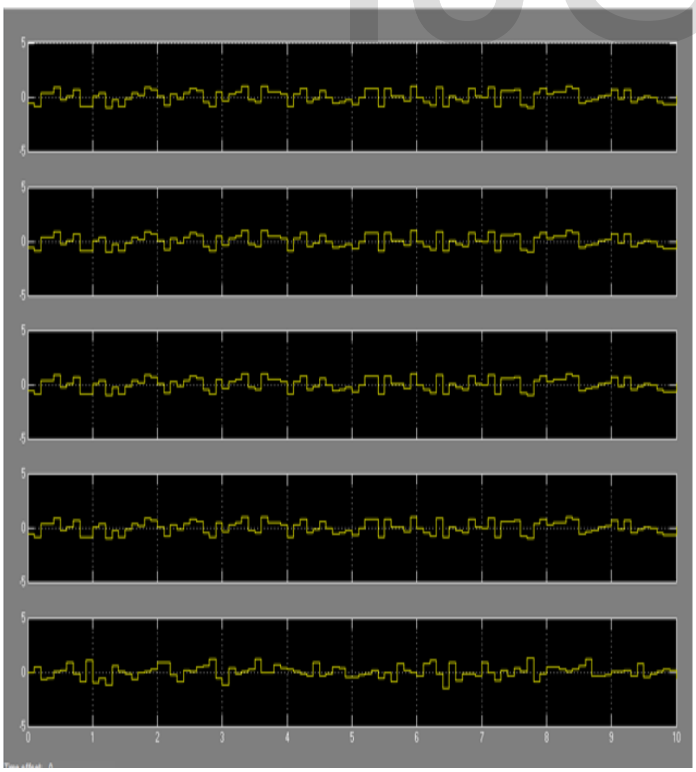

Figure 6. Resampling of random signal

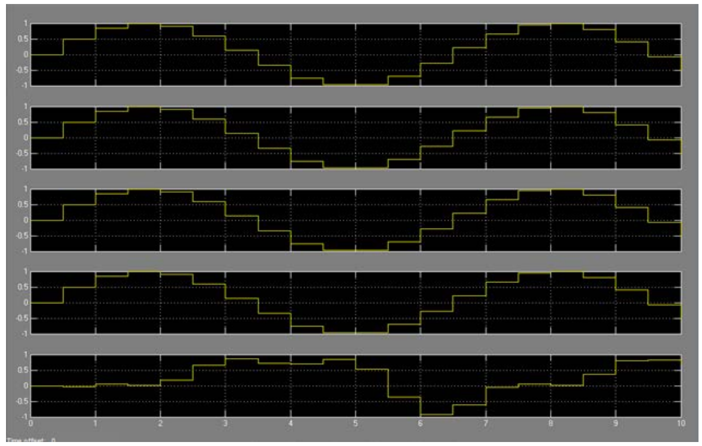

Figure 7. Resampling of sinusoidal signal

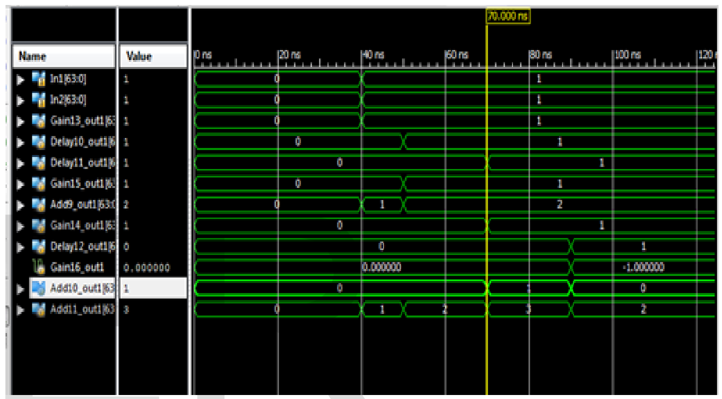

Figure 8. Calculation of Lagrange's coefficients

The process of calculating the coefficients is done by the subsystem.

Figure 9. Sinusoidal signal

6 Conclusion

The resampling device is developed using MATLAB Simulink and the corresponding verilog implementation is also done. The developed device allows the transmission of the signal in telecommunication field in error free manner.

Acknowledgment

This work was supported by the management, teaching and non- Teaching staffs of Sahyadri college of Engineering and Management.

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