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Implementing Resampling Circuitry for the Better Performance of Telecommunication Networks

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Abstract: **In this work, the main focus was on designing and implementing a resampling circuitry for telecommunication Networks. There are many situations where unwanted signal namely noise create problems for modulation and demodulation of data in telecommunication networks. This paper describes the implementation of device to avoid up this issue to a great extent. There are two parts in this paper. The first part develops a model of resampling device using MATLAB Simulink. The signals are resampled at this stage. Next, the Verilog description of the design is implemented using Xilinx ISE and the accuracy of the device is tested and verified. The resampled signals are compared with the original signal in the results.**

Keywords: **Resampling, Modulation, Telecommunication, Networks, Simulink, Verilog.**

1. Introduction

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

Existing researches mainly concentrates on the applying oversampling in order to avoid the problems occurred during 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 indented to be digitized is a radio frequency signal resulting in high sampling rate and hence high-power dissipation. To overcome this issue there are three blocks used in the device- RF processor which is analog, a digital IF stage and a base band processor.

In the analog end first stage that is, 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, is used to sample the previously produced signal with twice the intermediate frequency. A rate which is much higher than Nyquist rate is used due to pass sampling rate. The resultant signal is the band limited IF signal sampled at a high frequency rate.

This signal is then mixed with another signal in a local oscillator to create a low frequency signal.

In the intermediate frequency stage, CORDIC algorithm is used. CORDIC algorithm uses the basic shift and adds operations to obtain the values to trigonometric functions. It uses coordinate rotation. The radix-2 CORDIC algorithm has iterative equation for vector notation as 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 oversampling has different effects. They are discussed in detail and the solution is recommended. To use a digital down converter which, in effect will reduce the power consumption

Pertsev L and Timoshenko A, introduced the relationship ADC and DQPSK has [2]. In today's world, the important thing is to provide devices with minimum power. But in reality, the two factors to be considered are the power consumed by FPGA and ADC. Thus, ADC limits and sample rate resolution are discussed.

B. Lakshmi and A. Agarwal, suggested using a sample rate converter for RF frequency receiver using FPGA [3]. The signal that comes as the receiving data is RF frequency and it has a frequency range which is large. The implementation of this sample rate converter is discussed.

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Dodgson, N. A. suggested a quadratic interpolation for resampling process [4]. There are different varieties of interpolation. Quadratic interpolation is a separate type discussed by author. The basic step of resampling is interpolation.

3. Resampling of Signal

Digital signal processing of signal causes different problems in the telecommunication field and resampling is used to solve such problems.

The reason behind using resampling and not oversampling is, the disadvantages of oversampling. The main disadvantage of oversampling is the frequency of the oversampling clock device which is determined by the existing oversampling algorithm.

Resampling is the process in where both interpolation and decimation are combined and the sampling factor of an already digitized signal is changed by a rational factor. Here the sampling rate is changed but at the same time the information is preserved.

4. Proposed System

A. Interpolation

Interpolation is the process where new data points are constructed within the range of a discrete set of the known data points. Interpolation is also called as the up-sampling process in simple terms.

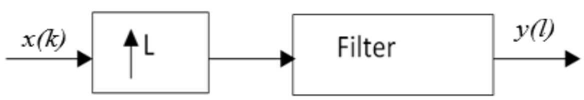

Fig. 1. Interpolator

It is possible to obtain interpolation results from different methods of interpolation.

1. Interpolation using linear method
2. Interpolation using bi –linear method
3. Interpolation using bi-cubic method
4. Interpolation using Lagrange's polynomials

In the proposed system interpolation by Lagrange's polynomials is used.

Hence the interpolator used is called Farrow interpolator.

Lagrange's polynomials explain about a curve that passes through N points and of which N points, only one is a unity.

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

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

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

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

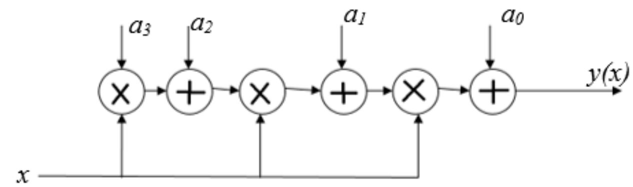

Fig. 2. Implementation of Lagrange's polynomial

Figure 2 shows the implementation of Lagrange's polynomial to obtain the interpolated signal. In the figure *x* represents the original signal which is to be resampled, that is the input signal. The same signal is processed by multiplications and additions to obtain the end result *y(x)*. This process is continuously repeated until equation 5 is obtained.

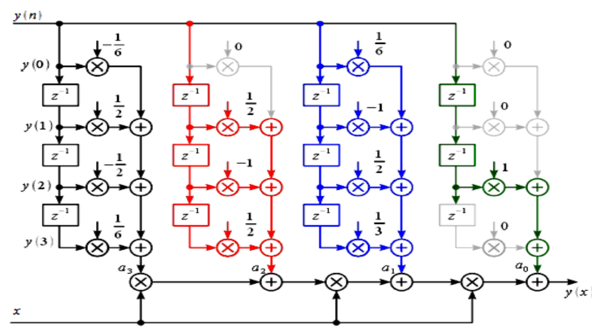

Fig. 3. Calculation of Lagrange's coefficients

Figure 3 is the calculation method of Lagrange's coefficients. Original signal is represented as *x* and *y(n)* represents the discrete values of the original signal.

5. Implementation

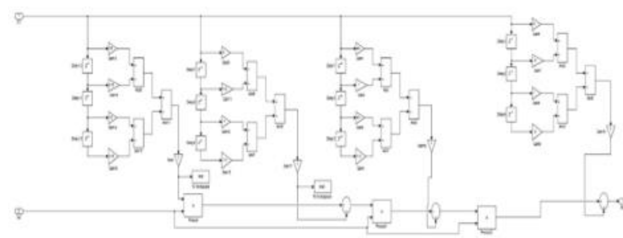

Fig. 4. Resampling device in MATLAB simulink

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

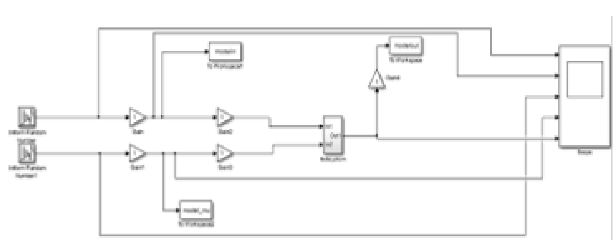

Fig. 5. Simulation of resampling

The main system of the implementation is shown by figure 5. The main system consists of several other blocks other than the Farrow interpolator subsystem. Main system HAS 'oscilloscope, gain blocks, buffers' etc. as components. The outputs are displayed on the oscilloscope.

6. Results

The output waveforms can be verified using oscilloscope as they can be displayed. Uniform random numbers are given to obtain the results of resampling.

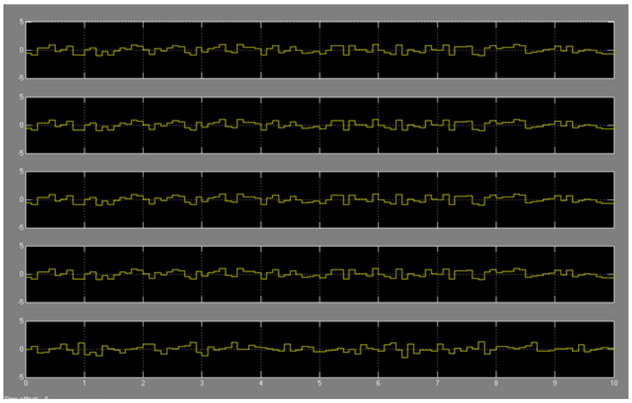

Fig. 6. Resampling of random signal

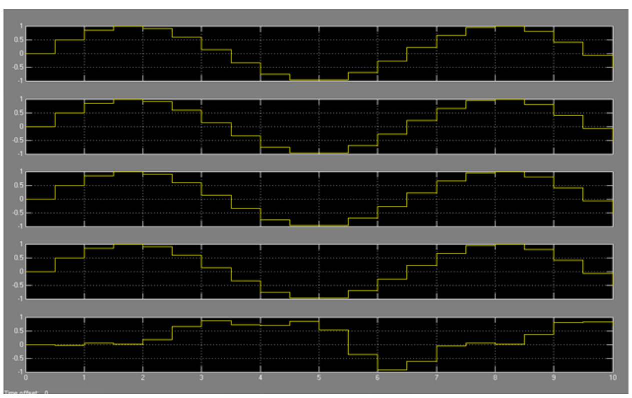

Fig. 7. Resampling of sinusoidal signal

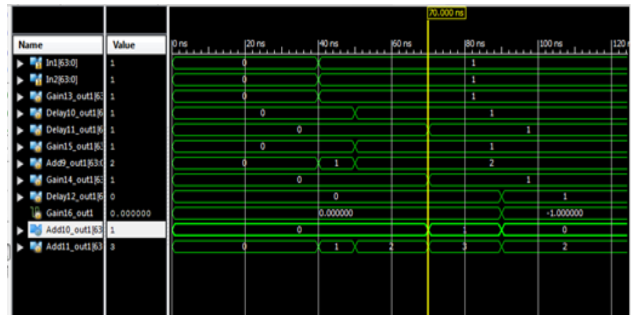

Fig. 8. Calculation of Lagrange's coefficients

The coefficients are calculated by the subsystem.

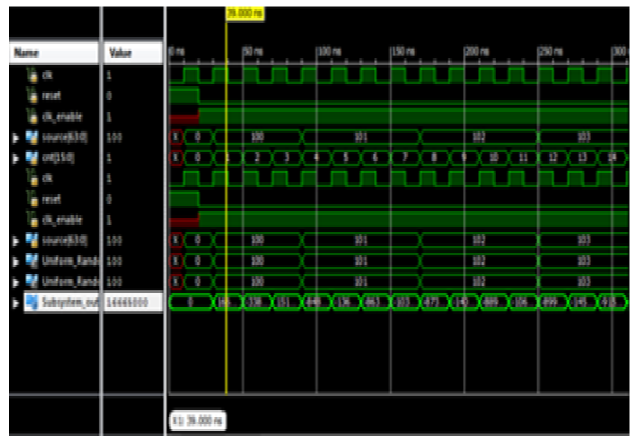

Fig. 9. Sinusoidal signal

7. Conclusions and Future Scope

The device developed using MATLAB Simulink and the corresponding Verilog implementation to resample the signals yields better output when compared to the results of previous works conducted. When tested the developed device showed the transmission of the signal in telecommunication field with less amount noise compared to the existing systems This shows that this field is having a lot of scope in future to explore and may even possibly be able to develop a nearly errorless transmission systems in the near future.

References

[1] A. Agarwal, L. Boppana and R. K. Kodali, "A factorization method for FPGA implementation of sample rate converter for a multi-standard radio communication," IEEE 2013 Tencon - Spring, 2013, pp. 530-534.