Performance Analysis of Fractional Sample Rate Converter Using Audio Applications

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Abstract: *Fractional rate converters which are generally used for many applications with different frequencies and are an essential part of communication systems. In this paper fractional rate converter with use of both FIR and Nyquist FIR have been compared and analyzed. Its implementation can be easily found in the developing communication systems, but here results are taken for audio applications. The proposed design and analysis have been developed with the help of MATLAB with order 50 for FIR and 71 for Nyquist, sampling frequency 48000Hz. The filters are then interpolated by an interpolation factor 2 and decimated by a decimation factor of 3. The cost implementation of both has been taken into consideration and a result is drawn which concludes that fractional rate converter for Nyquist FIR filter much more cost effective as compared to the fractional rate converter for FIR filter.*
Keywords: *Filters,MATLAB, Interpolation,Signal sampling, multirate filtering*

I. Introduction

DIGITAL representation of analog signals has a lot of advantages but changing the sampling rate of a required signal is the major problem, either increasing it or decreasing it by some required amount. For example, in communication system to transmits and receives the signals of different types (e.g. video, fax, speech etc.), in this case require to process the various signals at the different rates with respective bandwidth of the signals.

The process of converting a signal from a given rate toanother required rate is called as "sampling rate conversion" and the in the processing of digital signals systems that apply multiple sampling rates are called as "Multirate digital signal processing system" [1]. The need for digital products are growing day by day with programmability. Various industries like audio, video, and cellular industry rely heavily on digital technology [8].

In digital communication, we require Multirate digital signal processing where more than one sampling rate are needed. In digital audio applications of communication, the different sampling rates used are broadcasting use 32 KHz, compact disc use 44.1 KHz and audio tape use 48 KHz. In digital video applications, the sampling rates for composite video signal are 14.3181818 MHz for NTSC and 17.734475 MHz for PAL.

But the digital component of video signals have sampling rates are 13.5 MHz for luminance and 6.75 MHz for color difference signal. Different sampling rate are obtained by up sampler and down sampler .The digital signal processing is used for various field are communication system, speech and audio processing systems, antenna systems and radar system.Digital signal processing is also have advantages in multirate signal processing because of less computational requirements, less storage for filter coefficient , less finite arithmetic efficient, in multirate application, low finite order required and less sensitivity to filter coefficient length. Pseudo image for interpolation and aliasing effect for decimation should be avoided during the design of multirate systems [2].

II. Sampling

Consider continuous signal be x=(10, 9,8, 7, 6, 5, 4, 3, 2, 1,) and signal time is t (in sec), then x (t) be a continuous signal which is time varying that sampled with T sampling period at regular intervals of time. Then continuous sampling signal x (t) is given by:

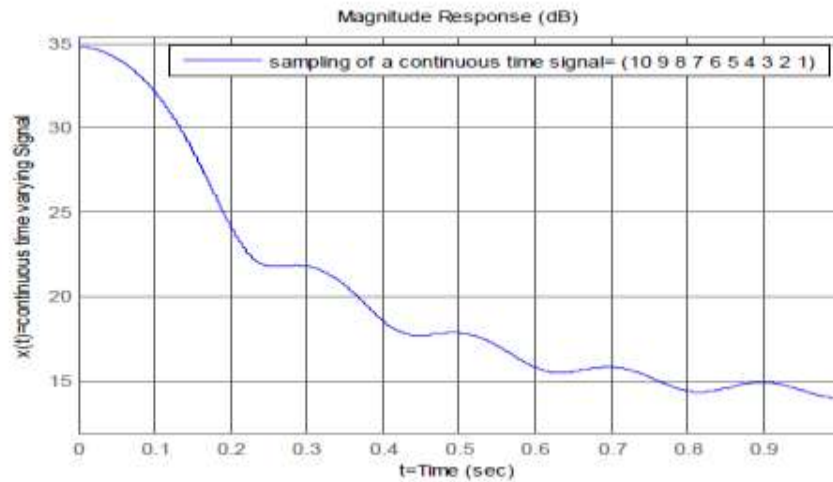$$X(nT) = x(t) \big|_{t=nT} \quad -\infty < n < \infty \quad (1)$$
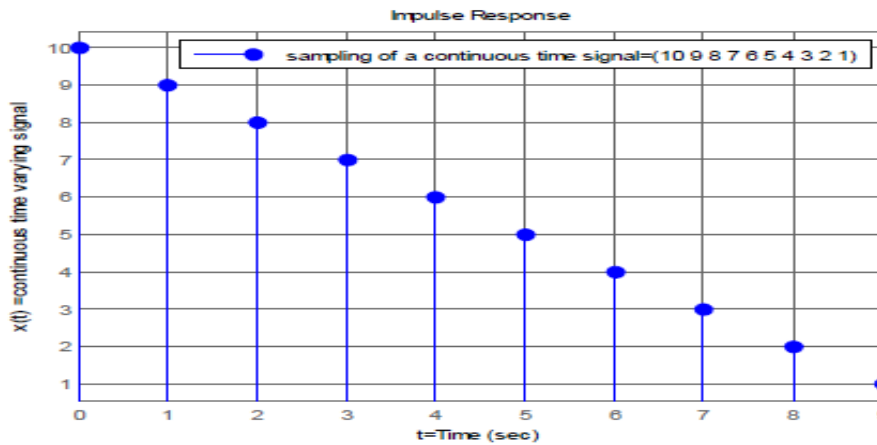
Figure 1: Continuous time signal with sampling

Figure 2: Discrete time signal with sampling

The continuous time signal x(t) is multiplied by the sampling function s(t) which is series of impulse, the x(n) resultant signal which is discrete time signal is given by:

$$x(n) = x(t)S(t) \big|_{t=nT} \quad -\infty < n < \infty \qquad (2)$$

The given signal output is shown in Fig. 2

x(0) = 10; x (1) = 9; x(2) = 8; x(3) = 7; x(4) = 6; x(5) = 5; x(6) = 4; x(7) = 3; x(8) = 2; x(9) = 1

II (i) Sampling rate conversion

Sample-rate conversion is the process of changing the sampling rate of a discrete signal to get a new sampling rate representation of given continuous signal. Sampling rate conversion have area of application are audio/visual systems andimage scaling where different sampling-rates may be used for economic, engineering, or historical reasons[4].

The sampling rate conversion (SRC) is based on the criteria of objective such as integration cycle, complexity, and performance characterization. The SRC proposed system has the ability to fully recovering characteristics and rounding noise behaviour [3].The applications of up/down sampling techniques and digital filter are form in every direction in modem electronic products. For every electronic product, lower circuit complexity is always an important design target since it reduces the cost. In every field many applications where the sampling rate must be changed [9].

Sampling rate conversion is the process of changing the x(n) discrete sequence of the continuous time signal x(t) with a period T,to y(k) another sequence obtained from sampling of continuous signal x(t) with a period T'. The new sequence y(k) can be generated by first reconstruction the x(t) original signal from x(n) the sequence discrete signal and then the reconstruction signal x(t) is sampled with a period T'.
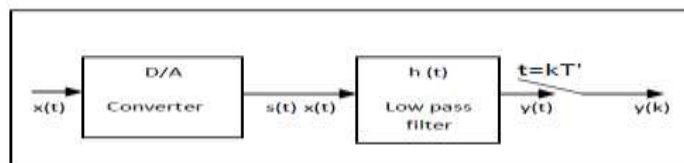
Figure3: conversion of discrete sequence x (n) to another discrete sequence y(k)

II (ii)Decimation

Decimation is the sampling rate conversion which reduces the original sampling rate of a sequence to a lower sampling rate. It perform the opposite nature of interpolation. Decimatelow pass filters the input to guard against aliasing and downsamples the result[5]. Decimator also known as Down-sampler. Let the sampling rate reduction factor be an integer M for the x (n) signal. Then the period is:

$T'/ T = M$

The F' new sampling rate become:

$F' = 1/T' = 1/MT = F/M$

Let the full band signal which is given by x(n), with non-zero values in the frequency range $-F/2 \le f \le F/2$, where $w = 2\pi fT$

By selecting only the M_{th} sample of the filtered output the y(k) sequence is obtained which results in sampling Rate reduction. Let h(n) be the impulse response of the filter, then the filtered w(n) output is given by:

$w(n) = \sum_{k=-\infty}^{\infty} h(k)x(n-k) \quad (3)$

$w(e^{jw})$ is *thespectrum* x (n). The y(m) decimation signal is:
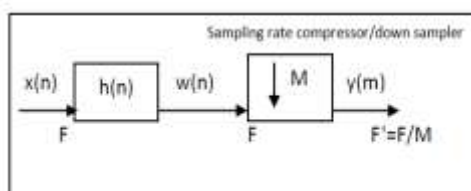
$y(m) = w(Mm)$

Figure 4: x(n) sequence conversion with down sample M

II(iii) Interpolation

Interpolation is a sampling rate converter which increases the original sampling rate of a given sequence to a higher sampling rate sequence. It perform the opposite nature of decimation. Interpolator inserts 0's into the original signal and then applying a low-pass interpolating filter to the new generated sequence by inserting 0"s.. [5]. Interpolator also known as Up-sampler. Let integer L be an interpolation factor be of the signal x (n), then the period is:

$T'/ T = 1/L$

The new sampling rate is given as:

$F' = 1/T' = L/T = LF$

Interpolation of a signal x(n) be a factor L refers to the process of interpolation L-1 samples between each pair of samples of x (n).

The y (m) output signal is given as:
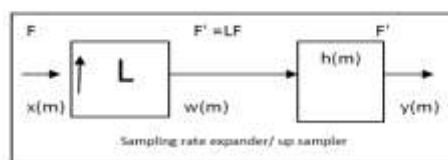
$y(m) = \sum_{k=-\infty}^{\infty} h(m-k)w(k) \quad (4)$

Figure 5: x(n) sequence conversion with up sample L

III. Multirate Signal Processing

Since the early 1980s Multirate systems have gained popularity and generally used for transform analysis, audio and video processing and communications systems. In various applications multirate signal processing are used for increased computational efficiency or to improve the performance. Multirate signal processing refers to systems which allow sequences that are from different sampling rates to be processed together. A multirate digital signal processing system that uses signals with different sampling frequencies. Multi-rate processing and sample rate conversion, or interpolation and decimation are clever digital signal processing (DSP) techniques that broadband and wireless design engineers can use during the system design process. Multi-rate processing finds use in signal processing systems where various sub-systems with differing sample or clock rates need to be design together. In other way multi-rate processing is used to reduce computational overhead of a system [6].

Multirate Signal Processing is the sub-area of DSP concerned with techniques that can be used to simply change the sampling rates within a system. There are many applications in which the signal of a given sampling rate require to convert into an equivalent signal with a different sampling rate. The process of decimation and interpolation both are used to perform multirate signal processing. They allow the sampling frequency to be decreased or increased without effects of errors such as quantization and aliasing [7].Interpolation is the computation of the lost, or the unknown, samples of a signal using a weighted average of a number of known samples at the district points. Interpolators are used in various forms in most decision making systems andsignal processing [10].

For example, an algorithm requires k operations to be completed per cycle. By reducing the sample rate of a signal or system by a factor of M, the arithmetic bandwidth requirements are reduced from fs operations to fs/M operations per second. The two basic operations in a multirate system are decreasing (decimation) and increasing (interpolation) the sampling-rate of a signal. Multirate systems are sometimes used for sampling-rate conversion, which involves both decimation and interpolation. In multirate digital signal processing the sampling rate of a signal is changed in order to increase the efficiency of various signal processing operations [6].

IV.   Filter Response And Characteristics

A multirate use both interpolation and decimation as per the requirement for audio applications. In this paper the sampling frequency Fs is 48000Hz. The interpolation factor is 2 and decimation factor 3. For further analysis the order 50 for FIR and 71 for Nyquist FIR. Magnitude and Phases analysis have been given in Figure6 and figure7 for fractional rate converter use FIR filter.The phase response of the filter shows variation from zero to negative and then again rise back from -39.22197 as in the Figure7. As for the Impulse Response of the filter, it shows that a symmetry occur on both side of time 0.5ms, which can be easily seen in Figure8.
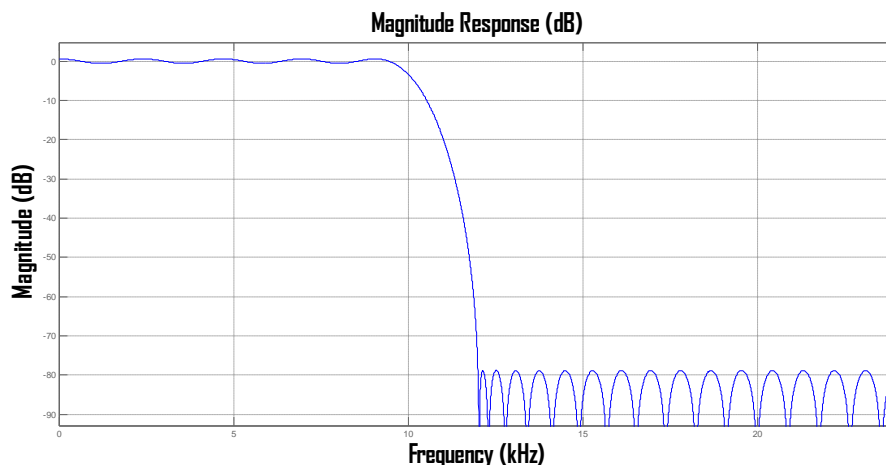

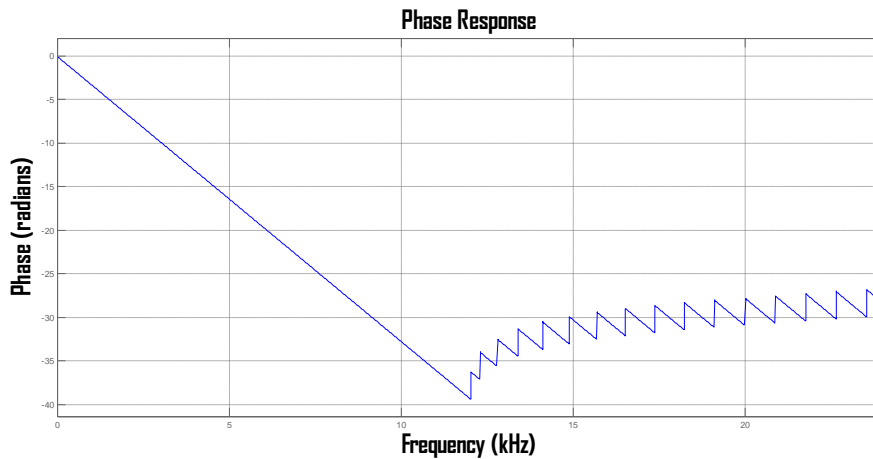**Figure 6: Magnitude Analysis of fractional-rate conversion using FIR Filter with Fs 48000HZ**

Figure 7: Phase Analysis of fractional-rate conversion using FIR Filter with Fs 48000HZ
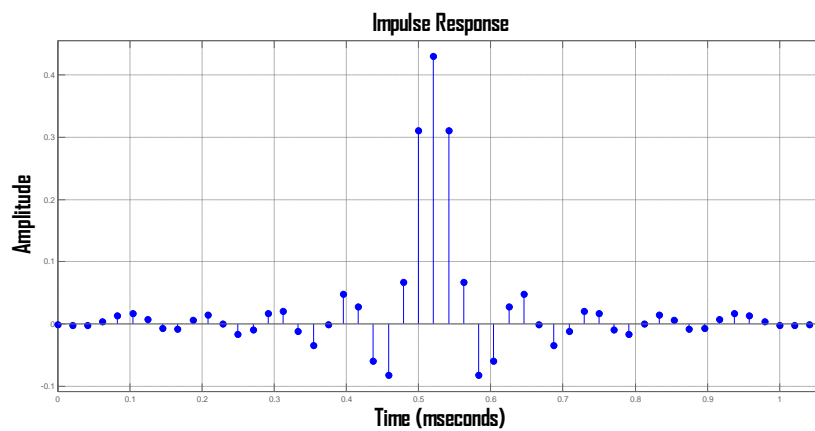

Figure 8: Impulse Analysis of fractional-rate conversion using FIR Filter with Fs 48000HZ

Another fractional rate converter use Nyquist FIR filter for audio application. It has the same sampling frequency of 48000Hz, as the previous filter. Interpolation factor and decimation factor also remains same, i.e. 2 and 3 respectively.

Magnitude and Phases analysis have been given in Figure9 and Figure10 for Nyquist FIR. As for the phase response of the filter shows variation from zero to negative (-45.61445) as can be seen in the Figure10. The Impulse Response of the filter shows that symmetry on both side of time equal to 0.75ms, which can be easily seen in Figure11.
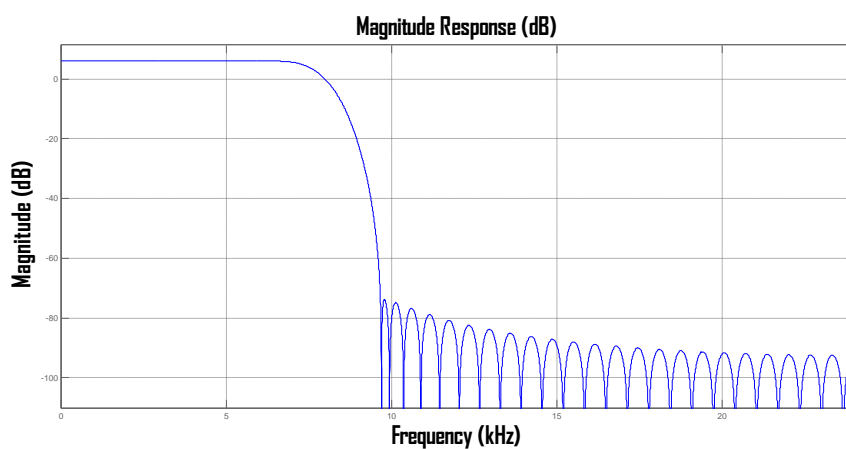

Figure 9: Magnitude Analysis of fractional-rate conversion using Nyquist FIR Filter with Fs 48000HZ
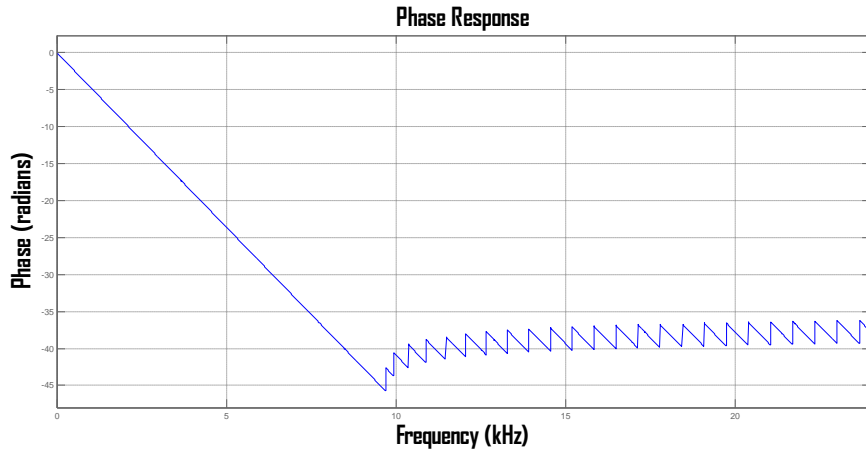
Figure10: Phase Analysis of fractional-rate conversion using Nyquist FIR Filter with Fs 48000HZ
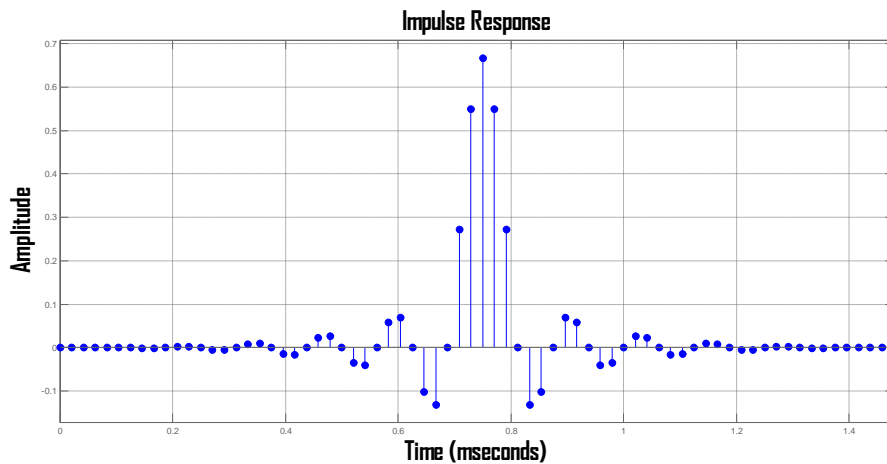

Figure 11: Impulse Analysis of fractional-rate conversion using Nyquist FIR Filter with Fs 48000HZ

V.   Filter Performance Comparison

The Fractional rate conversion have been developed for many applications, in this we use audio application with Fs= 48000HZ. Fractional rate conversion is carried out in both the filters with same decimation factor of 3 and same or interpolation factor 2. The sampling frequency r is kept same for FIR and Nyquist FIR filters so that the various performance parameters can be compared on an equivalent basis and hence comparison results can be obtained. In this paper magnitude response comparison has been made of the fractional rate converter for FIR and Nyquist FIR with L=2 and M=3, in case of audio applications, shown in Figure12.

Phase response comparison has also been made in Figure13. In this the blue plot(L=2 and M=3) for fractional rate conversion with FIR filter while the green plot (L=2 and M=3) for fractional rate conversion with Nyquist FIR filter.

As for the impulse response, both results have been brought up on a single plot for comparison. In the Figure14, blue plot represents Fractional rate converter use FIR filter with L=2 and M=3 which has maximum impulse response 0.4300479 at time 0.520833ms, while the green plot which represents fractional rate converter use Nyquist filter with L=2 and M=3 has maximum impulse response 0.66677 at time 0.75ms.
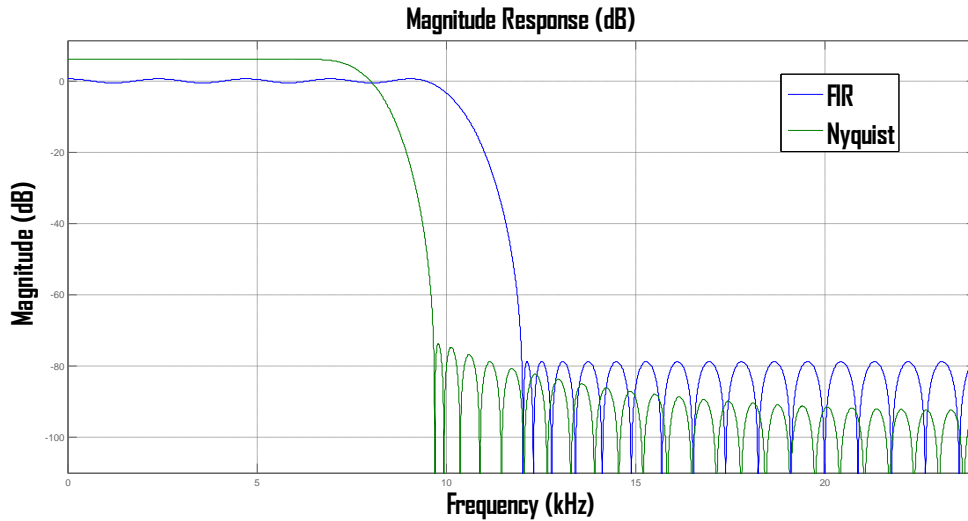
Figure 12: Magnitude Comparison of Fractional Rate Conversion of FIR and Nyquist FIR Filters
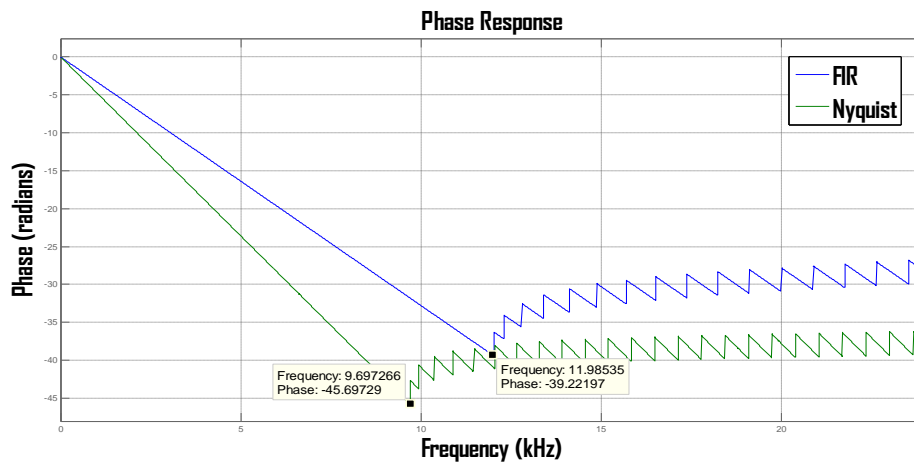

Figure 13: Phase Comparison of Fractional Rate Conversion of FIR and Nyquist FIR Filters
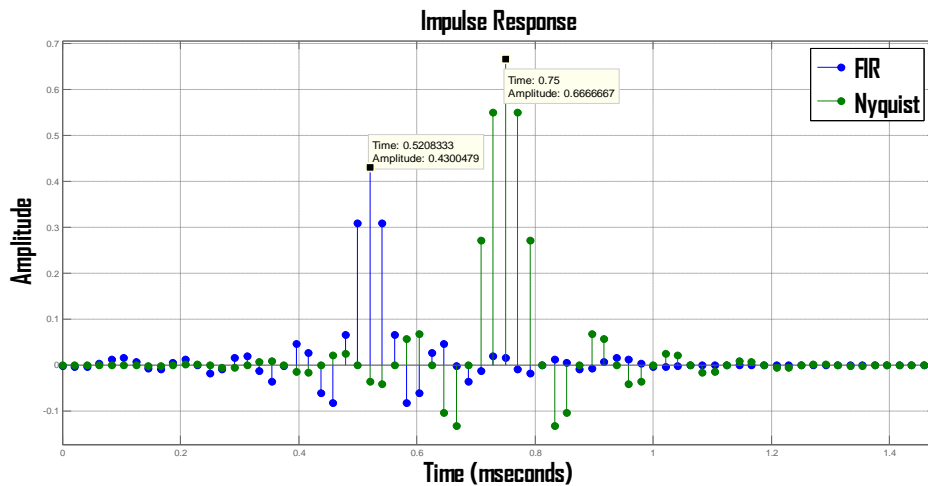

Figure 14: Impulse Response comparison of Fractional Rate Conversion of FIR and Nyquist FIR Filters

The cost of both the filters along with resource consumption are compared and results have been drawn in Table1

Implementation Cost	Fractional rate conversion using FIR filter	Fractional-rate conversion using Nyquist FIR filter
Poly-phase length	26	36
Filter length	51	72
No. of Multipliers	51	49
No. of Adders	49	47
No. of States	25	35
Multiplication per Input sample	17	16.3333
Additions per Input sample	16.3333	15.6667

Table 1: Resource Comparison of both the fractional rate converter using Fs 48000HZ

Hence the requirement of multipliers and adders are less in case of Nyquist FIR Filter, for same interpolation factor 2 and same decimation factor 3 with Fs =48000HZ (audio applications). . Both the filters have linear phase and stability.

VI. Conclusion

In this paper, an analysis and comparison of 2 fractional rate converter use FIR filter and fractional rate converter use Nyquist FIR filters with L=2 and M=3 has been presented with sampling frequency 48000 HZ and magnitude response, phase response and impulse response have been observed. These have been developed for various audio applications used in real time. The number of adders and multipliers, Filter length are also on a comparison, hence finding the better one for optimizing area and speed simultaneously. The whole design can be implemented by using sampling frequency as 48000Hz. From the developed design it is clear that the Nyquist FIR filter is more cost effective solution of audio application systems.

Acknowledgements

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