e-ISJN: A4372-3114
p-ISJN: A4372-3115

ISSN: 2321-7782 (Online)
ISSN: 2347-1778 (Print)
Impact Factor: 7.327

Volume 8, Issue 4, April 2020

International Journal of Advance Research in Computer Science and Management Studies

Research Article / Survey Paper / Case Study

Available online at: www.ijarcsms.com

Review on Multirate Signal Processing

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Abstract: Multirate techniques are now very popular in digital signal processing. Multirate sampling plays an very important role in the world of communication. They are particularly effective in subband coding, and various techniques for economical information saving have been developed. Their function is to alter the rate of the discrete-time signals, which is achieved by adding or deleting a portion of the signal samples. Multirate sampling is used in conversion of sampling rate. Firstly this paper gives an overview on multirate sampling. And this paper also gives an reviews on Digital filters. "Multirate" simply means "multiple sampling rates". A multirate DSP system simply uses more than one sampling rate within the system. In many systems, multirate DSP increases processing efficiency, which reduces DSP hardware requirements. During the last decade, however, they have increasingly found applications in new and emerging areas of signal processing, as well as in digital communications.

Keywords: Up-sampling, Doun-sampling, Subband coading, multirate filters.

I. INTRODUCTION

As long as people have tried to send or receive information through electronic media, such as telegraphs, telephones, television, radar, etc., there has been the realization that these signals may be affected by the system used to acquire, transmit, or process them. Sometimes these systems are imperfect and introduce noise, distortion, or other artifacts. Understanding the effects these systems have and finding ways to correct them is the foundation of signal processing. There are many types of signal processing. Among those Digital signal processing is more efficient and widely used. Multirate systems are building blocks commonly used in digital signal processing (DSP). Their function is to alter the rate of the discrete-time signals, which is achieved by adding or deleting a portion of the signal samples. "Multirate" simply means "multiple sampling rates". A multirate DSP system simply uses more than one sampling rate within the system. In many systems, multirate DSP increases processing efficiency, which reduces DSP hardware requirements[1].

Also, a few systems are inherently multirate, for example, a "sampling rate converter" system that converts an input sampling rate to a different output sampling rate. Multirate systems play a central role in many areas of signal processing, such as filter bank theory and multiresolution theory, they are essential in various standard signal-processing techniques such as signal analysis, denoising, compression and so on. During the last decade, however, they have increasingly found applications in new and emerging areas of signal processing, as well as in digital communication

Multirate Filtering techniques are used when conventional method becomes extremely costly and this technique is widely used in both sampling rate conversion system and in constructing filters with equal input and output rates[4].

Two discrete signals with different sampling rates can be used to convey the same information. The two basic operations in sampling rate alteration process are the down-sampler and up-sampler. Hence, the two signals operating at two different sampling rates are carrying the same information. By using the discretetime operations, signal {x[n]} can be converted to {y[n]}, or vice versa, with minimal signal distortions.

These two operators can perform the sampling rate alteration: a down-sampler used for decreasing the sampling rate, and an up-sampler used for increasing the sampling rate[1].

DOWN-SAMPLING OPERATION

The down-sampling operation with a downsampling factor M, where M is a positive integer, is

implemented by discharging M–1 consecutive samples and retaining every Mth sample. Applying the downsampling operation to the discrete signal {x[n]}, produces the down-sampled signal

y[m]= x[mM]

The down-sampling operation is sometimes called the signal compression, and the down-sampler is also known as a compressor. A block diagram representing the down-sampling operation is shown in figure below.

Figure 1: Block diagram representation of a down-sampler

UP-SAMPLING OPERATION

The up-sampling by an integer factor L is performed by inserting L-1 zeros between two

consecutive samples. Applying the up-sampling operation to the discrete signal {x[n]}, produces the upsampled signal {y[m]} is defined as,

$$y[m]= \begin{cases} x[\frac{m}{L}], & m = 0, \pm L, \pm 2L \ldots\ldots \\ 0, & otherwise \end{cases}$$

The up-sampling is sometimes called the sequence expansion, and the term expander is sometimes used for the device[3].

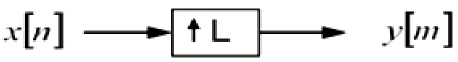

Figure 2: Converting the sampling rate with an up-sampler

II. LITERATURE SURVEY

The key importance of multirate filtering in modern digital signal processing systems is roughly three-fold. First, it is required when connecting together two digital systems with different sampling rates. Filtering is required to suppress aliasing when reducing the sampling rate, called as decimation, and to remove imaging when increasing the sampling rate, called as interpolation. The use of an appropriate filter enables one to convert a digital signal of a specified sampling rate into another signal with the desired sampling rate without significantly destroying the signal components of interest. In many cases, instead of using a single-stage system, it is more beneficial, in terms of lowed arithmetic complexity, to carry out the overall procedure with a multistage system. In this case, there is a need to design all stages containing a filter and a sampling rate conversion in the manner that the resulting overall system corresponds to a single-stage system. Either finite-impulse response (FIR) filters or infinite-impulse response (IIR) filters are used for generating the overall system. In some cases, both filter types are in use for building the overall conversion system.

Second, multirate filtering is required in constructing multirate as well wavelet filterbanks. Third, it one of the best approaches together with the proper use of complementary filter pairs for solving complex filtering problems when a single filter operating at a fixed sampling rate is of a significantly high order and suffers from output noise due to multiplication round-off errors and from the high sensitivity to variations in the filter coefficients.

The purpose of this paper is to give a short review on the above-mentioned first and third key advantages of using multirate filtering[2].

Multirate Filtering techniques are used when conventional method becomes extremely costly and this technique is widely used in both sampling rate conversion system and in constructing filters with equal input and output rates.

A multirate filter is a digital filter that changes the sampling rate of the input signal into another desired one. These filters are of essential importance in communications, image processing, digital audio, and multimedia. Unlike the single–rate system, the sample spacing in the multirate system can vary from point to point[3][4]. This often result in more efficient processing of signals because the sampling rates at various internal points can be kept as small as possible, but this also results in the introduction of a new type of error,i.e., aliasing[5].

The speech signal is taken as the input signal. AWGN is added with the input speech signal. The noisy speech signal spectrum is down sampled into multiple sampling rates using sampling rate conversion. Various transforms like Fast Fourier Transform (FFT), Fast Walsh Hadamard Transform (FWHT) and Discrete Wavelet Transform (DWT) are applied to the noisy speech signal and its sub bands. The FIR filters are designed and implemented using different window functions such as Rectangular, Hanning, Hamming, Blackman and Kaiser windows and the IIR filters are designed using Butterworth and Chebyshev filters. Then Quantization is applied to the filter coefficients. Finally the performance of the filter coefficients are measured based on the Signal to Quantization Ratio (SQNR). The Fig.1 shows the overall process of the proposed method [6].

This paper proposes a new adaptive noise cancellation structure based on multirate techniques. Noise Cancellation is chosen as the application because noise is one of the main hindering factors that affect the information signal in any system. Noise and signal are random in nature. As such, in order to reduce noise, the filter coefficients should change according to changes in signal behaviour. The adaptive capability will allow the processing of inputs whose properties are unknown. Multirate techniques can be used to overcome the problem of large computational complexity and slow convergence rate[7].

III. Conclusion

From this paper it is concluded that Multirate sampling is very important for converting sampling rate.It is also useful for eliminating problems of computational complexity,it also reduces cost against cvonvetional method.

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