Multirate Signal Processing Concepts Using Simulink

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Abstract- Different frequency domain multirate systems are blocks commonly used in digital signal processing (DSP). Their function is to adjust the rate of the discrete-time signals, which is achieved by adding or deleting a portion of the signal samples. During the last decade, however, they have increasingly found applications in new and emerging areas of signal processing such as biological measurements , electrocardiograms, control system signals, telecommunication transmission signals (analog and digital communication) ,image processing and many others. In this paper focus frequency domain what is happening at each stage of a system involving up samplers, down samplers, and low pass filters all computations will be performed using MATLAB and the signal processing toolbox. This is study paper.

Keyword: *Multirate, Up samplers, Down samplers.*

I. INTRODUCTION

Digital signal processing (DSP) is the mathematical process of an information signal to modify or improve it in some way. It is characterized by the representation of discrete time, discrete frequency, or other discrete domain signals by a sequence of numbers or symbols and the processing of these signals. The goal of DSP is usually to measure, filter and/or compress continuous real-world analog signals. The first step is usually to convert the signal from an analog to a digital form, by sampling (quantization) and then it using an analog-to-digital converter (ADC), which turns the analog signal into a discrete. However, often, the required output signal is another analog output signal, which requires a digital-to-analog converter (DAC). Even if this process is more complex than analog processing and has a discrete value range, the application of computational power to digital signal processing allows for many advantages over analog processing in many applications, such as error detection and correction in transmission as well as data compression.[1] Digital signal processing and analog signal processing are subfields of signal processing. DSP applications include: audio and speech signal processing, sonar and radar signal processing, sensor array processing, spectral estimation, statistical signal processing, digital image processing, signal processing for communications, control of systems, biomedical signal processing, seismic data processing, etc. DSP algorithms have long been run on standard computers, as well as on specialized processors called digital signal processor and on purpose-built hardware such as application-specific integrated circuit (ASICs). Today there are additional technologies used for digital signal processing including more powerful general purpose microprocessors, field-programmable gate arrays (FPGAs), digital signal controllers (mostly for industrial apps such as motor control), and stream processors, among others.[2] Digital signal processing can involve linear or nonlinear operations. Nonlinear signal processing is closely related to nonlinear system identification [3] and can be implemented in the time, frequency, and s patio-temporal domains [4]

The theory of multirate digital signal processing (DSP) has traditionally been applied to the contexts of filter banks and wavelets. These play a very important role in signal decomposition, analysis, modeling and reconstruction. Many areas of signal processing would be hard to predict without the use of digital filter banks. This is especially true for audio, video and image compression, digital audio processing, adaptive and statistical signal processing. However, multirate DSP has recently found increasing application in digital communications as well. Multirate building blocks are the crucial ingredient in many modern communication systems for example, the discrete multitone (DMT), digital subscriber line (DSL) and the orthogonal frequency division multiplexing (OFDM) systems as well as general filter bank precoders, just to name a few. Multirate signal processing has become an important area of digital signal processing since there are few standards that govern the rate at which data is collected and sampled. Multirate signal processing is a rich field for research, encompassing everything from deterministic operations like sampling rate conversion to statistical treatments using multiple observations. Some applications include sampling rate conversion for oversampling subsystems for CD or DAT players and sub band coding of speech in digital communications systems. Some statistical research involves wavelets for modulating signals to be transmitted over channels with unknown characteristics .In multirate digital signal processing the sampling rate of a signal is changed in order to increase the efficiency of various signal processing operations. Decimation, or down-sampling, reduces the sampling rate, whereas expansion, or up-sampling, fol- lowed by interpolation increases the sampling rate. Some applications of multirate signal processing are Up-sampling, i.e., increasing the sampling frequency, before D/A conversion in order to relax the requirements of the analog low pass anti aliasing. This technique is used in audio CD, where the sampling frequency 44.1 kHz is increased fourfold to 176.4 kHz before D/A conversion. Various systems in digital audio signal processing often operate at different sampling rates. The connection of such systems requires a conversion of sampling rate. Decomposition of a signal into M component containing various frequency bands. If the original signal is sampled at the sampling frequency fs (with a frequency) [4-16]. Some applications of multirate signal processing are:

A. Up-sampling

That is increasing the sampling frequency, before D/A conversion in order to relax the requirements of the analog low pass anti aliasing filter. This technique is used in audio CD, where the sampling frequency 44.1 kHz is increased fourfold to 176.4 kHz before D/A conversion. Various systems in digital audio signal processing often operate at different sampling rates. The connection of such systems requires a conversion of sampling rate.

B. Down-sampling

Decomposition of a signal into M components containing various frequency bands. If the original signal is sampled at the sampling frequency fs (with a frequency band of width fs/2, or half the sampling frequency), every component then contains a frequency band of width 2fs=M only, and can be represented using the sampling rate ½ fs/M only, and can be represented using the sampling rate fs/m .This allows for efficient parallel signal processing with processors operating at lower sampling rates. The technique is also applied to data compression in sub band coding, for example in speech processing, where the various frequency band components are represented with different word lengths.

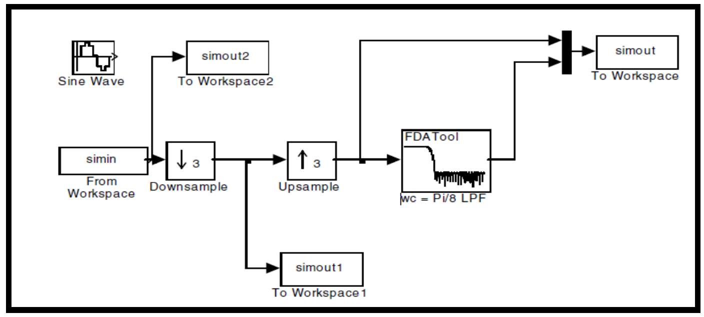

Fig.1: Block diagram of multirate dsp

A sinusoidal input can be connected (the plot currently off to the side) or a signal vector from the MATLAB workspace can be used as the input. As shown above the input from the workspace is a triangular spectrum signal. The up sampler and down sampler blocks works just like the command line functions. The low pass filter block has been designed using a GUI filter design tool (FDA tool), but ultimately uses coefficients similar to those obtained from MATLAB's command line filter design tools. The To Workspace blocks allow the signals to be exported to the MATLAB workspace for further manipulation. Such as running MATLAB code, and loading saved workspaces [17-19]

II. RESULT

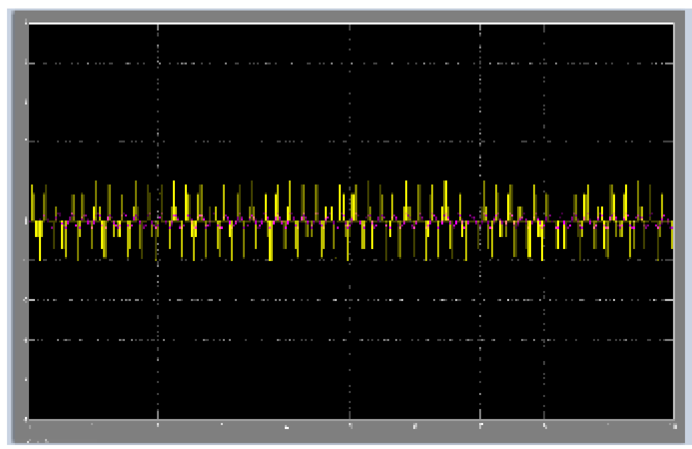

Fig.2: The output of simulation

From this simulation output the importance of performing the expansion first and the decimation second is to preserve the desired spectral characteristics of input signal which is in frequency domain of triangular shape. Further with the cascaded configuration illustrated in the above block diagram. The low pass filter operated at the same rate. Must incorporate the filtering operations for both expansion and decimation. Hence it should ideally posses the frequency response characteristic

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