MULTIRATE SIGNAL PROCESSING APPROACHES

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ABSTRACT:

Multirate Signal Processing studies Digital Signal Processing systems which include sample rate conversion. This technique is used for systems with different input and output sample rates, but may also be used to implement systems with equal input and output rates. We are going to study the architecture of multirate signal processing circuits that is Upsampler, Downsampler, Interpolator and Decimator in this paper.

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INTRODUCTION:

In Digital Signal Processing System sometimes it becomes necessary to convert the data to a new rate to make it easier to process or to achieve compatibility with another system. Therefore Multirate Signal Processing is used which is defined as the discrete time system that process data at more than one sampling rate to perform the desire digital operations. For example, if we wish to play Compact Disk (CD) music which has a rate of 44.1 KHz in a studio which handles data at a rate of 48 KHz rate, then the CD date must first be increased to 48 KHz using Multirate Approach. In this work, in general we are going to present a systematic approach for the low-power design of a general linear time-invariant (LTI) FIR/IIR system based on the multirate approach. The direct implementation of the system transfer function ***H (z)*** (see Fig. l(a)) has the constraint that the speed of the processing elements must be as fast as the input data rate. It cannot compensate the speed penalty under low supply voltage. On the other hand, the multirate system in Fig. l(b) will require only low-speed processing elements at one-third of the original clock rate to maintain the same throughput. Therefore, the processing elements can be operated at a lower supply voltage to reduce the power dissipation and the data throughput rate is not degraded by the lowered voltage. As a result, the multirate implementation can provides a direct and efficient way to compensate the speed penalty in low-power designs at the algorithmic/ architectural level**.**

Based on this design concept, we present a design methodology for the design of DSP systems.

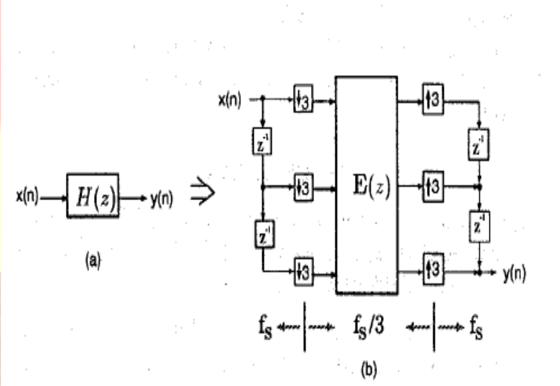


Figure 1.(a) An LTI FIR/IIR system. (b) Its equivalent multirate implementation, where ***fs*** is the data sampling Rate.

OPERATIONS OF MULTIRATE PROCESSING:

1] Upsampler
2] Downsampler
3] Decimator
4] Interpolator

1] Upsampling:-

Upsampling creates more samples in the same amount of time, typically by inserting zero-valued samples between the preexisting samples. However, if one considers that a discrete signal is already zero between the sample points, the approach begins to make more sense. An up-sampler with an up-sampling factor L, where L is a positive integer and every Lth sample is taken from x[n] with all others zero which develops an output sequence xe[n] with a sampling rate that is L times larger than that of the input sequence x[n]. To restore the original spectrum, the upsampler should be followed by a low-pass filter with gain L and cutoff frequency π / L. In this application, such an anti-aliasing filter is referred to as an interpolation filter and the combined process of upsampling and filtering is called interpolation.

$$x[n] \longrightarrow \boxed{\uparrow L} \longrightarrow x_e[n]$$

Block-diagram representation

Syntax:-

y = upsample (x,n)
Increases the sampling rate of x by inserting n-1 zeros between samples. x can be a vector or a matrix. If x is a matrix, each column is considered a separate sequence. The upsampled y has x*n samples.

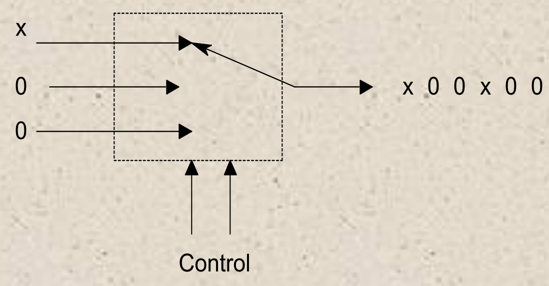
Example:-

Increase the sampling rate of a sequence by 3:

x = [1 2 3 4]; y = upsample(x,3);

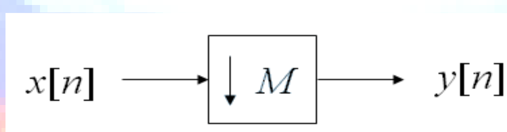
x = 1 2 3 4

y = 1 0 0 2 0 0 3 0 0 4 0 0



2] Downsampling:-

Downsampling is the process of discarding certain samples so that there are fewer samples in the same amount of time. Once discarded, those samples can never be replaced and error is introduced into the system. However, a downsampled system can also be processed with a slower filter, which is typically less expensive. An down-sampler with a down-sampling factor M, where M is a positive integer, develops an output sequence $y[n]$ with a sampling rate that is $(1/M)$-th of that of the input sequence $x[n]$. If the original sequence contains frequency components above π / M, the downsampler should be preceded by a low-pass filter with cutoff frequency π / M. In this application, such an anti-aliasing filter is referred to as a decimation filter and the combined process of filtering and downsampling is called decimation.



Block-diagram representation

Syntax:-

y = downsample(x,n)

Decreases the sampling rate of x by keeping every n-th sample starting with the first sample. x can be a vector or a matrix. If x is a matrix, each column is considered a separate sequence.

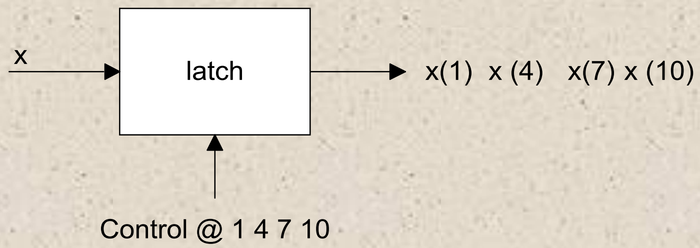
Example :-

Decrease the sampling rate of a sequence by 3:

x = [1 2 3 4 5 6 7 8 9 10];

y = downsample (x, 3)

y = 1 4 7 10



3] Decimator:-

In digital signal processing, **decimation** is a technique for reducing the number of samples in a discrete-time signal. The element which implements this technique is referred to as a **decimator**.

Decimation is a two-step process:

1. Low-pass anti-aliasing filter
2. Downsampling.

Syntax:-

y=decimate(x,r)

Decimation reduces the original sampling rate for a sequence to a lower rate, the opposite of interpolation. The decimation process filters the input data with a lowpass filter and then resamples the resulting smoothed signal at a lower rate. y = decimate(x,r) reduces the sample rate of x by a factor r. The decimated vector y is r times shorter in length than the input vector x.

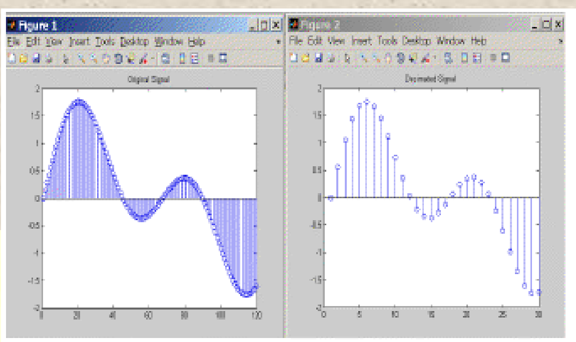
Example:-

Decimate a signal by a factor of four:

```
t = 0:.00025:1;                 % Time vector
x = sin(2*pi*30*t) + sin(2*pi*60*t);
y = decimate(x,4); % view the original and decimated signals
stem(x(1:120)), axis([0 120 -2 2])            %Original signal
title('Original Signal')
figure
```

stem(y(1:30)) % Decimated signal

title('Decimated Signal')



4] Interpolation:-

In the mathematical field of numerical analysis, interpolation is a method of constructing new data points within the range of a discrete set of known data points.

Syntax :-

y = interp(x,r)

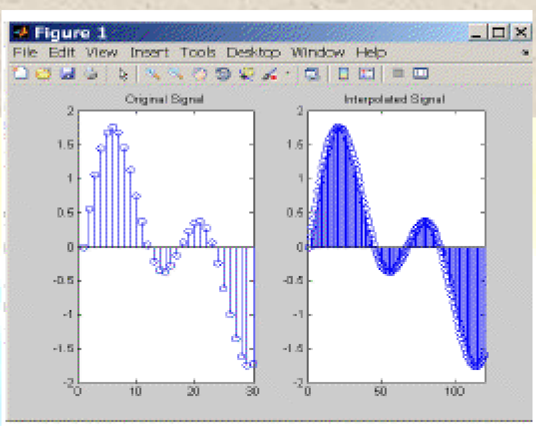
Interpolation increases the original sampling rate for a sequence to a higher rate. interp performs lowpass interpolation by inserting zeros into the original sequence and then applying a special lowpass filter. The filter returned by intfilt is identical to the filter used by interp. y = interp(x,r) increases the sampling rate of x by a factor of r. The interpolated vector y is r times longer than the original input x.

Example:

Interpolate a signal by a factor of four:

```
t = 0:0.001:1;     % Time vector
x = sin(2*pi*30*t) + sin(2*pi*60*t);
y = interp(x,4);
subplot(121);
stem(x(1:30));
axis([0 30 -2 2]);
```

```
title('Original Signal');
subplot(122);
stem(y(1:120));
title('Interpolated Signal');axis([0 120 -2 2]);
```



CONCLUSION:

In this paper, we presented the four main basic operations of multirate signal processing that are upsampler, downsampler, interpolator and decimator. This is a low power design using multirate approach which reduces the power consumption to a great extent but it increases the hardware complexity.

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