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# Designing of Two Channel QMF In Matlab

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***Abstract:*** **- Digital Signal Processing (DSP) is one of the fastest growing techniques in the electronics industry. It is used in a wide range of application fields such as, telecommunications, data communications, image enhancement and processing, video signals, digital TV broadcasting, and voice synthesis and recognition. The focus of this thesis is on one of the basic DSP functions, namely filtering signals to remove unwanted frequency bands. Multirate Digital Filters (MDFs) are the main theme here. Theory and implementation of MDF, as a special class of digital filters, will be discussed. Multirate digital filters represent a class of digital filters having a number of attractive features like, low requirements for the coefficient word lengths, significant saving in computation and storage requirements results in a significant reduction in its dynamic power consumption.**

***Keywords:*** **MDF, QMF**

## I. INTRODUCTION

In many practical application of digital signal processing, there is a problem of changing the sampling rate of a signal, either increasing it or decreasing it by some amount. For example, in telecommunication system that transmits and receives the different types of signals (e.g. fax, speech, video, etc), there is a requirement to process the various signals at the different rates with corresponding bandwidth of the signals. The process of converting a signal from a given rate to a different rate is called as “sampling rate conversion” and the systems that employ multiple sampling rates in the processing of digital signals are called as as “multirate digital signal processing system”

In signal processing, down sampling (or "sub sampling") is the process of reducing the sampling rate of a signal. This is usually done to reduce the data rate or the size of the data. The down sampling factor (commonly denoted by M) is usually an integer or a rational fraction greater than unity. This factor multiplies the sampling time or, equivalently, divides the sampling rate. Since down sampling reduces the sampling rate, it is usually a good idea to make sure the Nyquist–Shannon sampling theorem criterion is maintained relative to the new lower sample rate, to avoid aliasing in the resulting digital signal.

There are many reasons why FIR filters are very attractive for digital filter design. Some of them are Simple robust way of obtaining digital filters, inherently stable when implemented non recursively, Free of limit cycles when implemented non recursivelyEasy to attain linear phase, Simple extensions to multirate and adaptive filters.

## II. BASICS OF SAMPLING RATE CONVERSION

The sampled sequence x(n) obtained by sampling the analog signal $x_a(t)$ at the sampling rate of $F_s$. The sampling rate conversion is the process of obtaining a new sampling sequence y(m) of $x_a(t)$ directly from x(n).The newly obtained samples of y(m) are equivalent to the sampled values of $x_a(t)$ generated at the new sampling rate of $F_s$. Decimation and interpolation by integer values, both operations are considered fundamentals to multirate signal digital signal processing.

***Decimation :*** A continuous-time signal $x_c(t)$with sampling periods T and TD, where D is an integer. The corresponding discrete-time signals $x[n] = x_c(nT)$ and$x_D[n] = x_c(nTD)$ are related by

$$x_D[n]= x[nD]$$

The system defined as a down sampler or sampling rate compressor (SRC) or simply compressor, because it reduces the sampling rate of a discrete-time signal (down sampling) by an integer factor D. This system is essentially a discrete-time sampler which retains only one out of each group of D consecutive input samples into its output.
It can be implemented in matlab functions

y=downsample (X,D)

Where ‘X’ is signal

D is decimation factor

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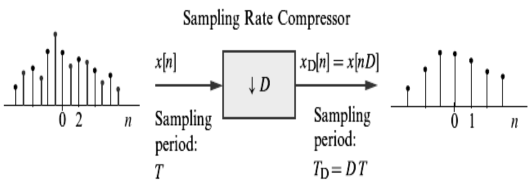



*Fig. Representation of a sampling rate compressor*

***Interpolation:***The processes of sampling rate reduction (often called decimation) and sampling rate increase (or interpolation). Sampling rate increase and sampling rate reduction are basically interpolation processes and can be efficiently implemented using finite impulse response (FIR) digital filters. The processes of sampling rate reduction (often called decimation) and sampling rate increase (often called interpolation)

Sampling rate increase and sampling rate reduction are basically interpolation processes and can be efficiently implemented using finite impulse response (FIR) digital filters. The IFIR (Interpolated FIR) approach results in a Two-stage decimator/interpolator. For the multistage approach, the number of stages can be either automatically optimized or manually controlled. But multirate/multistage design introduces the most delay as compare with IFIR Design. Increasing the sampling rate of a discrete-time signal x[n] by an integer factor I (upsampling) requires the insertion of (I −1) samples between consecutive samples of x[n]. Thus, upsampling is an information preserving operation.

$$x_1[n] \triangleq x_c(nT_I) = x_c\left({}^{nT}/_{I}\right)$$

WhereTI = T/I, from the samples of the discrete-time signal

$$x[n] = xc(nT).$$

It can be implemented in matlab functions

Y=upsample (X,I)

Where 'X' is signal

I is interpolation factor

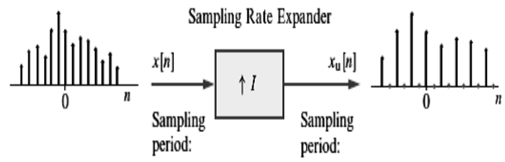


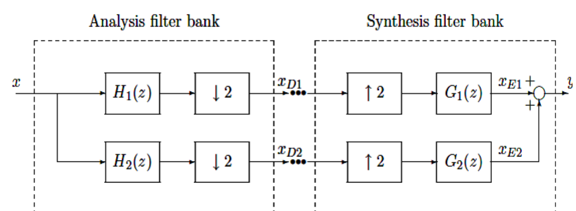

*Fig.: Representation of a sampling rate expander*

Which expands the input by a factor of I and then shifts the obtained sequence by insertingk zeros, k = 0, 1. . .I −1, in the beginning.

***Two- Channel QMF***

Quadrature mirror filter have been used extensively for splitting speech signals into sub band to achieve compression and efficient representation. The substantial progress in multirate digital filters and QMF filter banks has been made because of wide applications in many processing fields such as subband coding of speech, image signal processing, antenna system and Transmultiplexer. In QMF bank the input signal x[n] splits into two subband signals having equal bandwidth.

The subband signals are then processed and finally combined by a synthesis filter bank resulting in an output signal y[n]. The subband signal are band limited to frequency ranges much smaller than that of the original signal, so before processing they are down sampled. After processing these signals are up sampled before being combined by the synthesis band into a higher rate signal. The combined structure is called a Quadrature Mirror




***Filter.***

***Fig: General structure of a two channel QMF bank***

The z-Transforms of the input signal $X(z)$ are

$$X(z)= X(z)H_0(z) \qquad (1)$$

$$X_1(z)=X(z)H_1(z) \qquad (2)$$

The output signals $Y_0[n]$ and $Y_1[n]$ are added to obtain the single output $y[n]$.the z-transform of $y[n]$ is given as

$$Y(z) = Y_0(z) + Y_1(z) \qquad (3)$$

$$Y(z) = \frac{1}{2}[H_0(z)G_0(z) + H_1(z)G_1(z)]X(z) + \frac{1}{2}[H_0(-z)G_0(z) + H_1(-z)G_1(z)]X(-z) \quad (4)$$

$$Y(z) = T(z)X(z) + A(z)X(-z)(5)$$

Where, $T(z)$ is the distortion transfer function and $A(z)$ is the aliasing distortion. The first term is a desired signal and the second term is because of effect of aliasing which is to be eliminated with $A(z) = 0$

Therefore

$$H_0(-z)G_0(z) + H_1(-z)G_1(z) = 0 \qquad (6)$$

This condition is simply satisfied by selecting [5], the equations for perfect reconstruction conditions are.

$$H_0(Z) = H(Z)(7)$$

$$H^2(Z) - H^2(-Z) = 2Z^{-K}(8)$$

The aliasing and phase distortion has been eliminated completely. The distortion transfer function of the two-channel analysis/synthesis filter bank satisfying the Perfect Reconstruction property is a pure delayed function.

$$T(Z) = Z^{-n0}(9)$$

The FIR two–channel filter bank with linear–phase analysis and synthesis filters will be perfect reconstruction type if,

$$\left|H_0(e^{jw})\right|^2 + \left|H_1(e^{jw})\right|^2 = 1(10)$$

The equation (7) cannot satisfy exactly due to finite length of filter so it always exhibits some amplitude distortion unless it is a constant for all value of ω.

$$\emptyset = max\left\{\left|H_0(e^{jw})\right|^2 + \left|H_1(e^{jw})\right|^2 - 1\right\}(11)$$

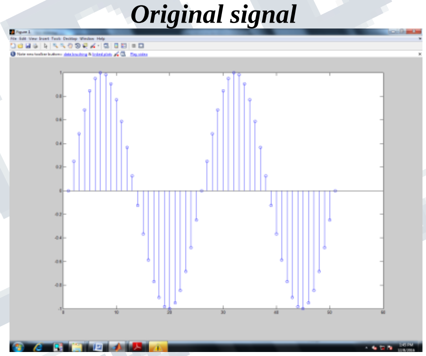
The two methods, the window method and Remez exchange algorithm are used generally to design prototype FIR filter, $(z)$. The objective function can be minimized using iteration procedure. The peak reconstruction error [8] is given as

$$PRE = max20 \log \log_{10}\left\{\left|H_0(e^{jw})\right|^2 + \left|H_1(e^{jw})\right|^2\right\}(12)$$

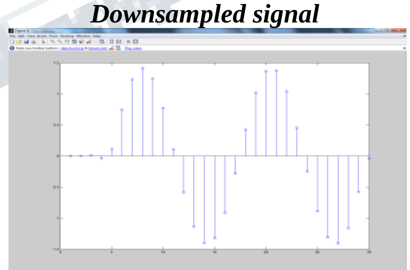
Window method is closed form method. The design entails a relatively insignificant amount of computation. Disadvantage of window method is that it needs a higher-order filter to satisfy the required specifications. A higher-order filter means more computations per sample, which implies that these filters are slower and less efficient in real-time applications.

## III. SIMULATED RESULTS

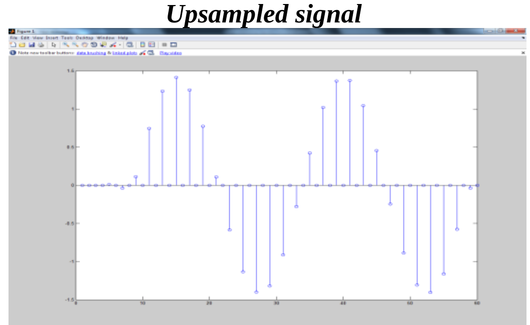
***Original signal***



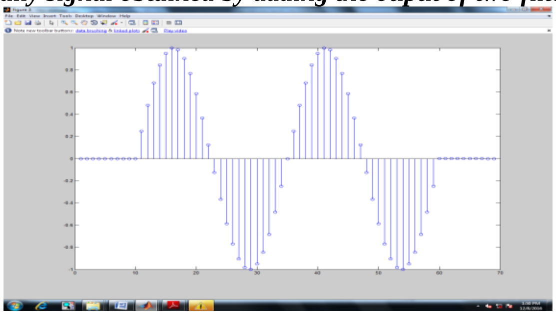
***Downsampled signal***



***Upsampled signal***



***Finally signal obtained by adding the ouput of two filters***



## IV. CONCLUSION

Design and implementation of digital filter bank have been done in this paper to reduce noise andreconstruct the input signals. This new process represents a significant improvement over analog filters that also reduce noises. Among the different tasks of digital filter bank, only one task is shown in this paper which is noise removal. In future work, I will try to implement others work ofDigital filter bank and invent new work over digital filter bank.

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