Multistage Implementation of Narrowband LPF by Decimator in Multirate DSP Application.

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Abstract: Decimator is an important sampling device used for multi-rate signal processing in wireless communication systems. Multirate systems have traditionally played an important role in compression for contemporary communication applications. In this paper it demonstrated that a multistage implementation of sampling rate conversion often provides for a more efficient realization, especially when filter specifications are very tight (e.g., a narrow pass band and narrow transition band) and there are a audio band of 4kHz bandwidth and compression by decimator to isolate the frequency component 80Hz. The LPF used by the decimator are acquire by two different approaches: first is the window method and other is frequency sampling method. Multistages implementations are used to further reduce the computational load. This approach drastically reduces the filter order and also reduces computational cost. Here it have reduce the overall computational complexity at single stage is 50 times and for second stages is near about 9 times reduce by the decimator factor is 50.

Keywords: DSP, Decimator, FIR, Multirate, MATLAB.

I. Introduction

The widespread use of digital representation of signals for transmission and storage has created challenges in the area of digital signal processing. Changing in the sampling rate of audio signal is common task in the signal processing field. In many practical application of digital signal processing, one is faced with the problem of changing the sampling rate of a signal, either expending it or reducing it by some quantity. For example, in telecommunication system that transmit and receive different type of signal (e.g., teletype, speech, video, etc.). There is a demand to process the different signals at different rates commensurate with the corresponding bandwidth of the signals. The process of changing a signal from a given rate to a different rate is called sampling rate conversion. In turn, systems that hire multiple sampling rates in the processing of digital signals are called multirate digital signal processing (DSP) systems. The noble identity states the equality of filtering when a filter is exchange with the down-sample or up sample operator which is shown in Fig.1 shows the two cases of the noble identity. Now assume it have a low pass filter whose transfer function is given by \( H(z) \). A polyphase representation of the transfer function is:

\[
H(z) = \sum E_i(z^M)
\]  

(1)

Every one of the \( E_i(z^M) \) terms in equation (1) is known as “one phase” of the actual transfer function \( H(z) \). It is simple to see that the phases can be implemented by using the noble identity. Thus, the low pass filtering can be done at a lower sampling rate. This conducts to decrease computational cost.

![Fig.1. Noble Identity](image)

Although in numerous practical applications the sampling rate is changed by a fractional number, say, \( M / L \). In such expressions, both decimation and interpolation are required to achieve the sampling rate conversion. In order to maintain the quality of the source audio, the narrowband frequency is done by the first decimator the source signal by the factor \( M \) followed by low pass filtering and interpolation by a factor \( L \). In the implementation of Narrow band Low pass Filter (LPF) by the compression of the audio band of 4kHz bandwidth by the multistage implementation of decimator of size \( M \), followed by a multistage implementation...
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II. Decimators

Basically low pass filters (LPF) are used for decimation and for interpolation. When decimating, low pass filters are used to decrease the bandwidth of a signal initial to decreasing the sampling rate. This is done to reduce aliasing due to the reduction in the sampling rate. When decimating, the bandwidth of a signal is decrease to an suitable value so that minimal aliasing occurs when decreasing the sampling rate.

**Fig. 2 (Down Sampler)**

Down sampler is primary sampling rate alteration appliance used to reduce the sampling rate by an integer factor. An down-sampler with a down-sampling factor M, where M is a positive integer, develops an output series y[n] with a sampling rate that is (1/M)-th of that of the input sequence x[n]. The down sampler is shown in Fig2 [2]. Decomposition of a signal into M components carrying a different 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), each element then carry a frequency band of width 2fₛ/M only, and can be represented using the sampling rate fₛ/M. This authorize for efficient parallel signal processing with processors operating at lower sampling rates. The technique is also applied to data compression in implementation of narrowband low pass filter (LPF), for example in communication processing, where the high frequency (audio-band) band components are represented with different narrowband. Multistage implementation of decimation by factor M shown in fig3.

**III. Multistage Implementation Of Sampling Rate Conversion**

In practical application of sampling-rate conversion it often encounter decimation factor and interpolation factor that are much larger then unity. It consider method for achieve sampling rate change for either M>>1 and/or L>>1 in multiple stages. First let us consider interpolation by a factor L>>1 and let us suppose that L can be factorizing into a product of positive integers as.

\[ L = \prod_{i=1}^{J} L_i \]

In a similar way, decimation by a factor M, Where M may be factored into a product of positive integer as

\[ M = \prod_{i=1}^{J} M_i \]

Can be implemented as a cascade of J stages of filtering and decimation as shown in fig.3. So the sampling rate at the output of the ith stage is

\[ F_i = F_{i-1}/M_i \quad i=1,2,\ldots,J \]

Where the input rate for the sequence \{x(n)\} is \( F_0 = F_x \). To ensour that no aliasing arise in the final decimation process. It can design each filter stage to avoid alising within the frequency bands of interest. To elaborate, let us define the desired passband and transition band in the overall decimator is

- Passband: \( 0 \leq F \leq F_{pc} \)
- Transition band: \( F_{pc} \leq F \leq F_{sc} \)

Where \( F_{pc} \leq F_x/2M \). Then,aliasing in the band \( 0 \leq F \leq F_{pc} \) is avoided by selecting the frequency band of each filter stage as follows:

\[ F = \frac{F_x}{2M} \]

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Passband: $0 \leq F \leq F_{pc}$
Transition band: $F_{pc} \leq F \leq F_{pc} - F_{sc}$ (3)
Stopband: $F_{pc} - F_{sc} \leq F \leq (F_{pc} - 1/2)$

For example, in the first filter stage it have $F_1 = F_x / M_1$, where the $F_x$ is the input frequency which is compressed by decimator at different different stages.

Passband: $0 \leq F \leq F_{pc}$
Transition band: $F_{pc} \leq F \leq F_{pc} - F_{pc}$ (4)
Stopband: $F_{pc} - F_{pc} \leq F \leq (F_{pc} - 1/2)$

After decimation by $M_1$, there is aliasing from the signal element that fall in the filter transition band, but the aliasing occurs at frequency above $F_{sc}$. Thus their is no aliasing in the frequency band $0 \leq F \leq F_{sc}$. By designing the filter in the following stages to content the specification given in equation (3), it ensure that no aliasing arise in the primary frequency band $0 \leq F \leq F_{sc}$.

It is a particularly simple formula for approximation the order of filter $N$, and length of the filter is calculated by the $D = (N+1)$, attributed to kaiser, is shown in equation (5) where $Δf$ is the normalized (by the sampling rate) width of the transition region, that is $Δf = (F_{sc} - F_{pc}) / F_s$ [3].

In this paper the sampling rate change from 8KHz to 160Hz, which required $M = F_x / 2 F_s = 50$ here it want to lowpass filter $F_{pc} = 75$ HZ and $F_{sc} = 80$ HZ, $F_p$, and $F_{sc}$ are the passband and the stopband frequency, respectively where $δ_1$ and $δ_2$ are the passband and the stopband ripple. Choose the value of $δ_1 = 10^{-2}$ and $δ_2 = 10^{-4}$, when it found the order of filter is $N = 5151$. In practice, it is difficult to implement a filter with such a high order. Another way is to do the low pass filtering (LPF) in multiple stages, and each stage need a much lower filter order. And also find the value of filter length is $N + 1 = 5152$. For reducing the order of the filter by the different stages in the multistage implementation by decimation factor $M$, which is shown in fig.3.

IV. Implementations and Simulation

In first stage filtering there are it downsampling by the factor $M = 50$ by the relation of $M = F_x / 2 F_s$, where $F_x$ is equal to the audio band sampled rate, and $F_s$ is equal to the generate narrow band sampled rate. It know the value of passband ripple $δ_1 = 10^{-2}$ and stopband ripple $δ_2 = 10^{-4}$ as per the LPF and also find the value of filter length $D$ with the help of kaiser window, that is equal to 5152. Finally find out the computational complexity, which is equal to the 824160 MPS (multiplication per second), by the equation (7). For first stage implementation, for an FIR $H(z)$ of
\[ R_{M,FIR} = N \times F_T \]  
For an FIR \( H(z) \) of length \( N \) followed by a down-sampler.

\[ R_{M,FIR,DEC} = N \times F_T / M \]  
When it note that in Fig. 3, for two stage implementations the decimation factors \( M_1 \times M_2 = M = 50 \).

If it demand \( G(z^{M}) \) to have the similar low pass specifications as \( H(z) \), and use \( I(z) \) to eliminate the images of \( G(z^{M}) \), then the two-stage filtering in Fig. 3 have the similar frequency response as \( H(z) \). Fig. 4 drawing the principle of this two stage IFIR approach. For more feature on the underling multirate signal processing theory, The advantage of using this IFIR implementation is that the transition bandwidth \( \Delta \omega \) in equation (5) can be mainly relaxed on both filters \( I(z) \) and \( G(z) \), therefore much lower filter orders are required to implement them. And find the computational complexity by the equation (7), for two stages. The next section describes design details of the IFIR approach.

\[ |G(\omega)| \]  
\[ F_p(75) \quad F_s(80) \quad \Pi \]

\[ |I(\omega)| \]  
\[ F_p(75) \quad F_s(240) \quad \Pi \]

\[ |H(\omega)| \]  
\[ F_p \quad F_s \quad (F_T - MF_s)/M \quad \Pi \]

Fig.(4) Frequency Response Two Stage Filtering
Simulation result of $G(z^M)$

![Simulation result of $G(z^M)$ filter response](image)

Simulation result of $I(z)$

![Simulation result of $I(z)$ filter response](image)

Simulation result of both filter is shown in fig-5-6.[5]

V. Design

The decimation factor $M=25 \times 2$ then $M_1=25$, $M_2=2$, a two stage decimator is implemented with $H(z) = G(z^{25})I(z)$. Filters $G(z)$ and $I(z)$ are designed in Matlab with the parameters given in section 3,4. Then first stage it have spacified $G=320$ Hz and

- Passband: $0 \leq F \leq 75$
- Transition band: $75 < F \leq 240$
- $\Delta f = \frac{165}{8000}$
- Ripple: $\delta_1 = \frac{\delta_1}{2}$, $\delta_2 = \delta_2$

Note that it have reduce the passband ripple $\delta_1$ by a factor of 2 so the total passband ripple in the cascade of two filter does not exceed $\delta_1$. On the other hand, the stop band ripple is maintained at $\delta_2$ in both stage. $\Delta f$ is find by $(F_s - F_p)/F_T$, where $F_s$ is syopband $F_p$ is passband frequency. By kaiser formula yields an estimate of length of filter $D_1=167$, and order of filter is 166, computational complexity of this stage is 53120 (MPS), by equation (7). For the second stage, it have $I=G/2=320/2=160$ and spacification

- Passband: $0 \leq F \leq 75$
- Transition band: $75 < F \leq 80$
- $\Delta f = \frac{5}{320}$
- Ripple: $\delta_1 = \delta_1/2$, $\delta_2 = \delta_2$

Hence the estimate of the length $D_2$ of the second filter is $=220$ and order of filter is 219 by kaiser formula. The computational complexity of 1 is 35040 (MPS).

Therefore, the total length of the two FIR filter is approximately $D_1 + D_2=387$. Thus represent the reduction in the length of filter by a factor of more then 13. Thus the reduction in the filter length result from...
increasing the factor $\Delta f$, which is appears in the denominator in equation (5). By decimating in multiple stages, it are able to increase the width of the transition region through a reduction in sampling rate.[7]

**TABLE 1**

<table>
<thead>
<tr>
<th>Filter</th>
<th>Kaiser Window</th>
<th>Computational Complexity (MPS)</th>
</tr>
</thead>
<tbody>
<tr>
<td>$G(z)$</td>
<td>219</td>
<td>35040</td>
</tr>
<tr>
<td>$I(z)$</td>
<td>166</td>
<td>53120</td>
</tr>
</tbody>
</table>

**TABLE 2** Filter order of window and computational complexity (MPS) and result how much complexity reduce.

<table>
<thead>
<tr>
<th>No of Stages</th>
<th>Filter Length</th>
<th>(MPS)</th>
<th>Result</th>
</tr>
</thead>
<tbody>
<tr>
<td>Without deominator</td>
<td>5152</td>
<td>41208000</td>
<td>-----</td>
</tr>
<tr>
<td>First stage (decimator)</td>
<td>5152</td>
<td>824160</td>
<td>50 time reduce (MPS)</td>
</tr>
<tr>
<td>Second stage (decimator)</td>
<td>387</td>
<td>88160</td>
<td>9.3 time reduce again (MPS)</td>
</tr>
</tbody>
</table>

**VI. Experimental Result**

To evaluate the performance of the multistage implementation of narrow band LPF and measure the filter order by the MATLAB program of kaiser window, and also find the length and the computational complexity for multistages, which is shown in Table 1,2. Table 1 show that computational complexity for both the filter $G(z)=35040$, $I(z)=53120$, computational complexity is find in term of number of multiplications per second, computational efficiency is improved significantly. Reason for using multistage are

1. Multistage system requires reduced computation
2. Storage space required is less
3. Finite word length effects are less

Table 2 show the computation for different stage first this show the without deominator, second show the first stage with decimator and third show the second stage with decimator, where the M=50 downsampled for conversion a 4kHz audio band into 80Hz narrowband frequency signal with the help of multistage technique.

Result in Table 2 show that how much number of computation is reduce in term of multiplication per second. When it take first stage in decimator then the computation is reduce 50 time shown in Table, when consider second stages then the no MPS (multiplications per seconds) is 9.3 times reduce.

**VII. Conclusion**

In this paper, generate the narrowband by the audio band with the help of multistage implementation. Here it also find the how much computation is reduce in MPS, which is shown in “Table 2” for different stages. The order of filter is also reduce, when computation load is less so the delay is also less, so speed is increased due to different stages, so that multistage decimator are best for performing downsampling and provide less computation solution and also cost effective solution for DSP based wireless communication applications. Here it have reduce the overall computational complexity at single stage is 50 times and 824160 multiplication per second (MPS) for second stages is near about 9 times and 88160 multiplication per second (MPS) reduce by the decimator factor is 50.
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References


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