DSP Filter Design for Flexible Alternating Current Transmission Systems

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Abstract. The paper analyses the design principles of both filtering techniques, FIR and IIR and will also overview the possibilities of the multirate and adaptive filters. The paper describes its features and makes a comparative analyse of them. It will also include some interesting algorithms with the implementations of some of these filters, generically and also in the machine language of one of the most important DSP producers of the time; Analog Devices.

Key words

FACTS, DSP, FIR, IIR

1. Introduction

Flexible Alternating Current Transmission Systems (FACTS) can be considered as an alternative to the construction of new transmission lines to face the increasing demand of energy without damaging the supply quality. Through the use of static power converters in the electrical energy networks it increases its transmission capacity and improves the supply quality.

To operate and control these systems multiprocessor systems are normally used, in which the Digital Signal Processors (DSP) play the most important part. DSPs have similar features to microcontrollers but they are more powerful and its CPU is optimized to process mathematical operations in real time, that is way they are very suitable for signal control.. In FACTS, DSPs are use to implement basic, global and centralized control algorithms.

This signal processing requires frequently of highly efficient filters which design and development we will analyze in this work. Instead of calculating the values of R, L and C like n Analogue filtering, in digital filtering we will have to calculate and use the filtering coefficients. We are going to study the most used filter structures; FIR (Finite Impulse Response) and IIR (Infinite Impulse Response).

The increase in the electrical energy demand has opened a way to new means of transmission with increased capacity and without damaging the power quality but improving it. One of these is the FACTS, and one of the elements to ensure the operativity and efficiency of these systems have turned to be the Digital Signal Processors, powerful, flexible real time processors.

The success of the DSPs in processing signal is that they made possible the implementation of the most efficient filters ever designed.

2. Digital Filters

Filter Design is based on the frequency Domain processing and on the time discrete systems. The mathematical grounds of these systems are the following:

- Z transformation:
  a) to analyse the time discrete signals in the time domain

- Fourier Series:
  a) to analyse the periodical and time continuous systems in the frequency domain
  b) to analyse the periodical and time discrete signals in the frequency domain

- Fourier Transformation:
  a) to analyse the non-periodical and time continuous signals in the frequency domain
  b) to analyse the non-periodical and time discrete in the frequency domain

There are two approaches to Fourier Transformation, which are very suitable for the signal processing and the core of its theory; the DFT (Discrete Fourier Transformation). This Transformation is very practical because we will always be working with sampled discrete signals, will we mostly be periodical. The problem is that we will only have a limited number of samples.

The second approach is based on the DFT. And is called the FFT (Fast Fourier Transform). This Transformation is a fast way of calculating the DFT, we can really call it a solving algorithm of the DFT which manages to solve it in less instructions. The number of additions and multiplication is less and thus the execution time also.

The signal theory and the FT, DFT and FFT are the means we use to design digital filters of which the most known are the following:

3. FIR. Finite Impulse Response

Filtering is a process of frequency selection. Because of this, the frequency response of the filter is the most important parameter. If we know the kind of response, we can calculate the filter coefficients according to it.

The general filter structures are defined through the values of the following parameters:

- Stopband: the frequencies belonging to this band are attenuated. The stopband ripple and the ripple ratio determine the attenuation.
- Passband: band, which frequencies will be allowed to pass by.
- Ripple: oscillations that happen in the stopand.

These parameters determine the filter structure:

- Low Pass filter: filters which allow the lower frequencies to pass by
- High Pass filter: filter which allow the higher frequencies to pass by.
- Band Pass filter: The passband area is located betwen two border frequencies that limit the stopband area.
- Stop Band filter: Here we also have two border frequencies but in this case they limit the stopband area. On the other sides of these frequencies we find the passband area, the lowest and highest frequencies are allowed to pass.
- Notch filter: It is a Band Pass filter but the with a extremely narrow bandwide.

FIR filters have not a corresponding analogue partner, and the simplest way of describing them is to use an equation that consists only on zeros.

$y(n)= \Sigma c(k) * x(n-k)$

Thus, these filters are non-recursive, and if we filter and impulse signal, its output signal will be zero, that is why they are called Finite Impulse Response. Its frequency answer is the following:

$H(f)=\Sigma c(k)*e^{-2\pi k\Delta f}$

This means that the frequency response is the Fourier transformation of the filter coefficients. We can calculate these coefficients using the inverse Fourier Transformation.

We can design a FIR filter with different techniques:

a) The window method:

If we want to calculate the coefficients, we will have the following problems; the inverse Fourier Transformation take samples from the continuous frequency response, and if we want to describe a good filter, we need to to take small sample slots, and this means that we will have a rather big amount of samples, which is a disadvantage. To solve these problems do the following; we determine the frequency response with a lot of samples and calculate the inverse Fourier Transformation, producing a lot of coefficients, which number we will reduce. With this reduced number of coefficients we will do again the Fourier Transformation and confirm if the result meets our former demands. Through the reduction the frequency response of the filter will be distorted. To rebuild the signal we will use the so called window functions. Thee are different windows; Bartlett, Blackman, Hamming, Hanning,... The disadvantage of the window function is the attenuation that the lobes of the signal suffer, ranging 20 to 97 dB's.

b) The equiripple Methode

It is achieved through the Remez exchange algorithm, changing between the frequency response and the filter coefficients until it finds the smallest number of coefficients. It is very effective but the execution of this algorithm takes long. A compromise between velocity and effectiveness needs to be taken, depending on the application.

ALGORITHM FOR A FIR

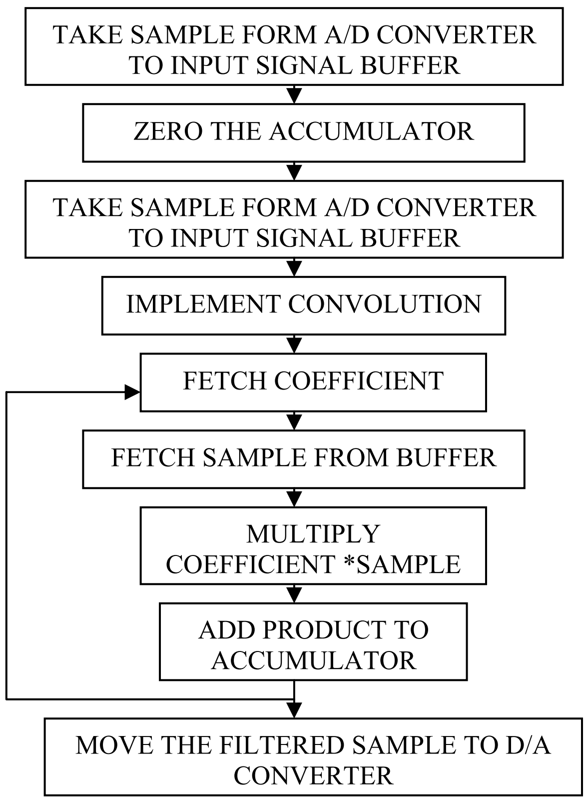

Fig1. FIR algorithm

If we want to implement the algorithm shown above, in an ADSP 21XX, from analogue devices we will have to write the following code:

First, it zeros the accumulator and positions the pointer in the sample and coefficient buffer respectively.

```
MR=0;
MX0=DM(I0,M1);
MY0=PM(I4,M5);
CNTR=N-1;
```

Then it does the convolution until all the coefficients have been used, by multiplying the sample and the coefficient and adding it in the accumulator

```
DO convolution UNTIL CE;
MX0=DM(I0,M1);
MY0=PM(I4,M5);
MR=MR+MX0;*MY0;
IF MV SAT MR;
RTS;
.ENDMOD;
```

This code is generally a subroutine of a main program, which has to include the initialization of the processor and normally have a different purpose than filtering.

4. IIR Infinite Impulse Response

As we said, FIR filters have no analog partners and are only defined by zeros. On the other hane IIR filters have analogue partners such as Chebyshev, Butterworth, Elliptical and Bessel. Because of these, they can be analysed and synthetized through traditional filter design techniques.

The response answer to the Impulse tends to infinite in this filters. This reaction occurs because they use feedback, they are not only defined by zeros but also by poles.

$$y(n) = \sum_{k=0}^{M} b_k x(n-k) - \sum_{k=1}^{N} a_k y(n-k)$$

The fact that IIR filters need less calculations to get to the final result is compensated because they don´t have a linear phase. Even though they are more effective than the FIR.

IIR filters are often represented by two biquads (Pole sections) and in case they have a higher order than two, we achieve the structure by cascading the biquad sections. In filters of higher orders, the problem can be, like in any other modelized system, that they are not stable enough. By cascading we avoid the instability.

The usual design techniques calculate the corresponding analog filter first and calculate the transference function in Laplace domain to Z domain. This conversion is a necessary step

The features of the analog filters are the following:

a) Butterworth.

This filter hat just a pole and no ripple in band pass and stopband areas. It gives the biggest response area.

b) Chebyshev

There are two kinds the first kind, is defined by poles and presents ripple in the Band pass area. The second kind had no ripple in the passband but presents some in the stopband.

c) Cauer (elliptical)

It is defined with poles and zeros and presents ripple in both band pass and stopband areas. It doesn´t have a good phase response.

d) Bessel

It is defined by poles and have no ripple. It´s effect is extremely good when the phase is linear.

Each type has it´s own transference function based on Laplace transformation. From this point on there are different ways of finishing the design:

a) Variable impulse transformation method:

We calculate the corresponding Z transformation and its corresponding sampled response to the impulse. The Z transformation gives the coefficients of the filters. The sample velocity has to be controlled to avoid the aliasing effect.

b) Bilinear transformation method:

We convert the transference function form H(s) to H(z) and effectivity of the conversion will be controlled by the differential equation that defines the analog system. There is no risk of aliasing.

c) Unified transformation method:

This method converts the Laplace domain transference function into the z domain, but can only be used on filter systems defined by poles and zeros.

If we compare the main features of the FIR and IIR filters we come to the following table:

TABLE I

FIR FILTER	IIR FILTER
Not so effective	Very effective
No corresponding analog filter	Analog partners
Very stable	Could get instable
Linear phase response	Non-linear phase response
No glitching or Ringing	Glitching and ringing.

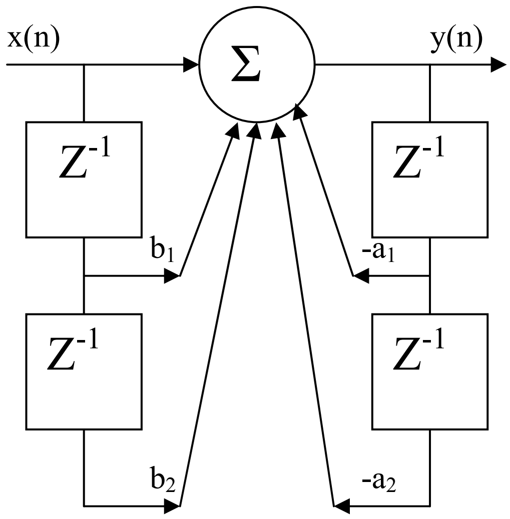

Fig2. Second-order IIR filter (biquad)

If we want to implement a IIR Band Pass filter with amplification of the Bass Band area, we can use the following subroutine (we don´t include the whole code lines). In this case we have used C language. Most of the Processors used now have C compilers and its programming using this language makes its use easier.

```
{
 float a,b,c,d;
float a0,a1,a2,b1,b2;
//Description of the variables for the filter coefficients
float Wn and Wp;
//Descripcion of the variables for the center and prewarp
//frequencies
float gain, freq_rate, Q
//Description of the cariables with the frequency rate,
gain and quality
//Asignation of values for al the variables
x[i]= sf*in[i]-b1*x[i-1]-b2*x[i-2];
out[i]=a0*x[i]+a1*x[i-1]+a2*x[i-2];
//Being sf the scale factor
//With these two last steps we implement the main
//equation for an IIR filter taking in account poles and
//zeros
}
```

5. Multirate and Adaptive filters

A.MULTIRATE FILTER

There are a lot of applications that require a change in the sampling rate. In these cases we use Multirate filters. This kind of filters use the following techniques to make this sampling rate change possible:

a) Decimation:

Through this method we reduce the sampling rate through a M factor. If we imagine the original signal with its frequency; fo and know that we will have to sample it with another frequency fs, we will realize that fs is much higher than necessary to transmit the signal. This means fo is oversampled. In this case we can reduce the sampling rate without causing whether lost of information nor aliasing.

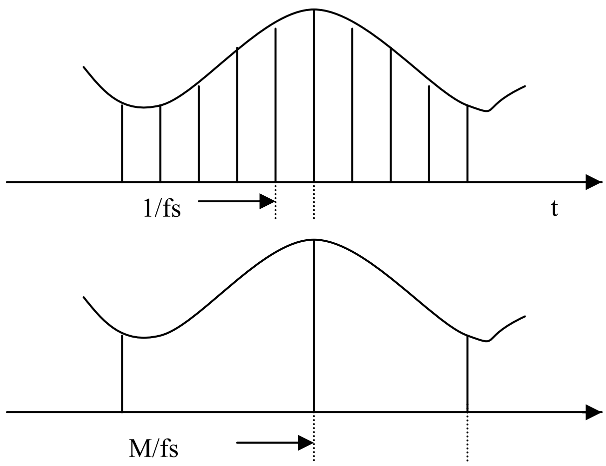

Fig3. Oversampled signat and decimated signal

b) Interpolation:

With this method we elevate the sampling rate by a factor of L. As in the example shown before we would have the original signal at a frequency of fo sampled with a frequency fs. If we multiply this fs by a L factor we have to add more zeros to the signal and then add a interpolation filter to produce the additional values.

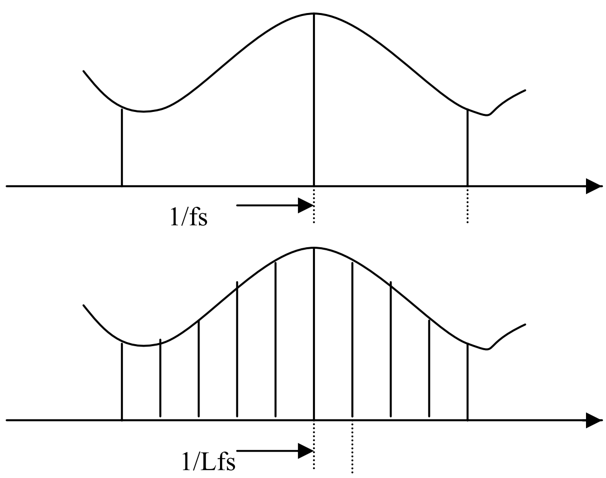

Fig4. Original signal and interpolated signal

The digital implementation of the interpolation happens this way; first the original signal will get through a multiplier which will amplifies the samplig frequency and add the necessary zeros. Then the data goes through the interpolation filter which flattens the input signal and interpolates it with the original data.

This two method can be used together to achieve certain applications in which the reduction as well as the amplification are needed.

B. ADAPTIVFILTER

The features of digital filters can easily be changed, to this aim, it is enough to change the filter coefficients.

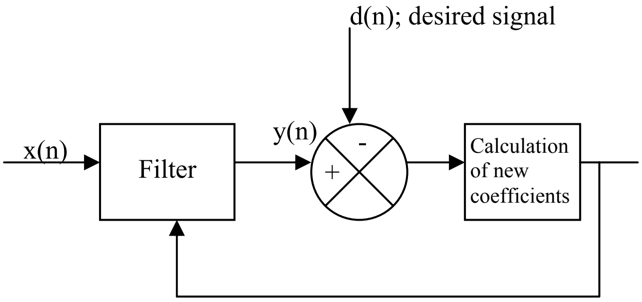

Fig5. Adaptivfilter

In this kind of filters we have an input signal that we filter in order to get an output signal. Additionally we have the desired signal which actually differs from the output signal in what we will call the error signal

The inovation of this filter is that it uses this error signal which will have to get through a block with an adaptive algorithm which will calculate the new optimized coefficients. By the use of these coefficients we minimize the error signal

4. Conclusion

Digital filters implemented on DSP lead to a successful signal process and enables these processors to have a leading role in control systems which operate Flexible Alternating Current Transmission Systems, thus enabling a improved capacity and power quality supply of the electrical energy transmission lines, which involves the general network system improvement.

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