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Harmonic Estimation in Power Signals Using a Filter Bank and an Adaptive Filter

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Abstract : A method is proposed that uses a filter bank and an adaptive filter for the harmonic estimation in power signals. Here the amplitudes and frequencies of each harmonic is estimated. A fundamental filter bank (FB) decomposes the input power signal into sub signals and multistage FB is used for higher order harmonic decomposition. An adaptive filter is used to estimate the parameters more accurately. Parameters describing the harmonic components are estimated from the output of the adaptive filter which is self-tuning.

Keywords: *Filter Bank (FB), power quality, adaptive filter, harmonic components, parameter estimation.*

1. Introduction

There has been much interest recently in the operation of an electrical power system. Power Quality (PQ) has to be maintained in electricity transmission and distribution. Electrical power is generated from remote areas and using several transformers, overhead lines or sometimes underground cables, power is transferred to load points. But electromagnetic interferences like lightning, undesirable effects propagate throughout the network change the characteristics of voltage and current waveforms in the power system. The increasing use of nonlinear loads, including electronic converters, renewable electric power sources, and equipment that has tighter tolerances for drive voltage disturbances, has placed growing demands on the quality of the power supply [1]. Power should be distributed to necessary locations with the fundamental frequency, 50 Hz, having a constant voltage magnitude. But due to the presence of harmonics, the voltage or current waveform get distorted. Then signal becomes a non-sinusoidal signal which it should not be. If power quality is not maintained, it will affect our home appliances and causes overheating of transformers, building wiring etc. [2].

Harmonics are voltages or currents at frequencies that are multiples of the fundamental frequency and any signal component having a frequency which is not an integer multiple of the fundamental frequency is designated as an interharmonic component or referred to more simply as an interharmonic. If 50 Hz is the fundamental frequency, then second harmonic is 100 Hz (50 x 2=100), third harmonic is 150 Hz (50 x 3=150) and so on. The even multiples of the fundamental frequency are called as even-order harmonics while the odd multiples are called as the odd-order harmonics.

Here a technique based on filter banks (FBs) and an adaptive filtering is used to characterize both stationary and nonstationary power signal variations. The proposed method consists of a multistage FB, an adaptive filter and an amplitude and frequency estimation algorithm. The input power signal is decomposed by the fundamental FB which is used successively in each stage to decompose higher order harmonics. Then these harmonics are processed using a self – tuning adaptive filter to estimate the parameters, amplitude and frequency, more accurately.

2. Description of the Method

2.1 FB Design

FBs are designed for decomposing a signal into sub signals [3]. A fundamental FB consists of a low pass filter, a high pass filter and a band pass filter and all the filters are designed as a quadrature mirror filter (QMF) bank to satisfy perfect reconstruction conditions as shown in Fig. 1 [4].

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In Fig. 1, x (n) is the input power signal, y_L (n) is the low-pass decimated signal, y_H (n) is the high-pass decimated signal and $y_B(n)$ is the band pass signal. $H_{LP}(z)$ is the low pass filter, $H_{HP}(z)$ is the high pass filter and $H_{BP}(z)$ is the band pass filter. $H_{LP}(z)$ and $H_{HP}(z)$ are designed with less overlap at the frequency $\pi/2$. So the overall magnitude response of

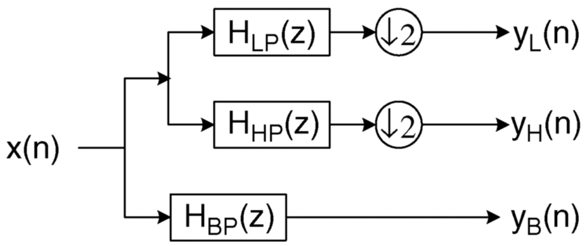

Fig. 1 Fundamental FB

the upper path is effectively a notch filter. Thus even order harmonics are rejected by the upper path and are processed through the band pass filter.

If the input power signal whose harmonics to be detected is applied to the fundamental FB shown in Fig. 1, then the first harmonic is y_L (n), second harmonic is $y_B(n)$ and third harmonic is $y_H(n)$. The frequency response of the filter under the aforementioned condition is shown in Fig. 2 [4].

The fundamental FB shown in Fig. 1 can be used in multistage for estimating higher order harmonics as shown in Fig. 3 [4]. Here the fundamental FB is used in three stages for detecting the harmonics h_1 to h_{15}. If x(n) is the input power signal given to the FB in stage 1, low pass filter produces low order harmonics h_1 to h_7, band pass filter gives h_8 and high pass filter produces high order harmonics h_9 to h_{15}. h_1 to h_7 is again given to the first FB at stage 2 where low order harmonics h1 to h3 is produced by the low pass filter, band pass filter gives h4 and h5 to h7 is produced by the high pass filter. Like this way, three stage FB decomposes the harmonics and h1 to h15 is detected and its frequency and amplitude is estimated. The relationship between the maximum harmonic order to be decomposed i and the stage number M is $i = 2^{M+1}-1$.

2.2 Frequency and Amplitude Estimation

The frequency f_i and amplitude a_i (i = 1 to 15) is estimated using the recursive algorithm [5] and [6]. For the frequency estimation of harmonic h_i, three consecutive samples of h_i are considered and frequency can be estimated as follows: [4]

$$z_i(n) = h_i(n-1) \tag{1}$$

$$q_i(n) = \frac{1}{2}[h_i(n) + h_i(n-2)] \tag{2}$$

$$x_i(n) = \frac{\sum_{k=0}^{n} \Psi^{2(n-k)} z_i(n) q_i(n)}{\Psi^{2(n-k)} z_i^2(n)} \tag{3}$$

$$f_i = \frac{\cos^{-1} x_i(n)}{2\pi T_i} \tag{4}$$

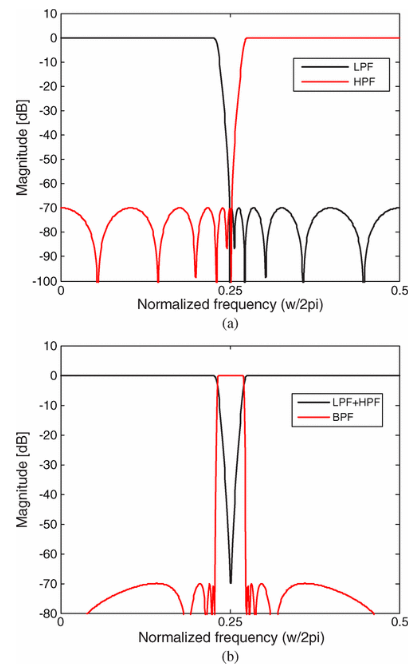

Fig. 2 Magnitude responses of (a) the $H_{LP}(z)$ and $H_{HP}(z)$ and (b) the upper path and $H_{BP}(z)$ in Fig. 1.

where Ψ^2 ($\Psi^2 < 1$) is the forgetting factor. In (4) T_i is the sampling period in the i^{th} sub band. If the i^{th} harmonic is represented by $h_i = a_i \sin(i\omega_1 n T_i)$,where a_i is the amplitude and ω_1 is the angular frequency of the fundamental, then amplitude can be estimated as follows: [4]

$$A_i(n) = h_i^2(n) - 2h_i^2(n-1)\cos(2i\, i\omega_1 n T_i) + h_i^2(n-2) \tag{5}$$

$$Bi(n) = 1 - \cos(2i\omega_1 nTi) \tag{6}$$

$$a_i(n) = \sqrt{\frac{A_i(n)}{B_i(n)}} \tag{7}$$

The amplitude and frequency of each harmonic is estimated from (4) and (12) respectively.

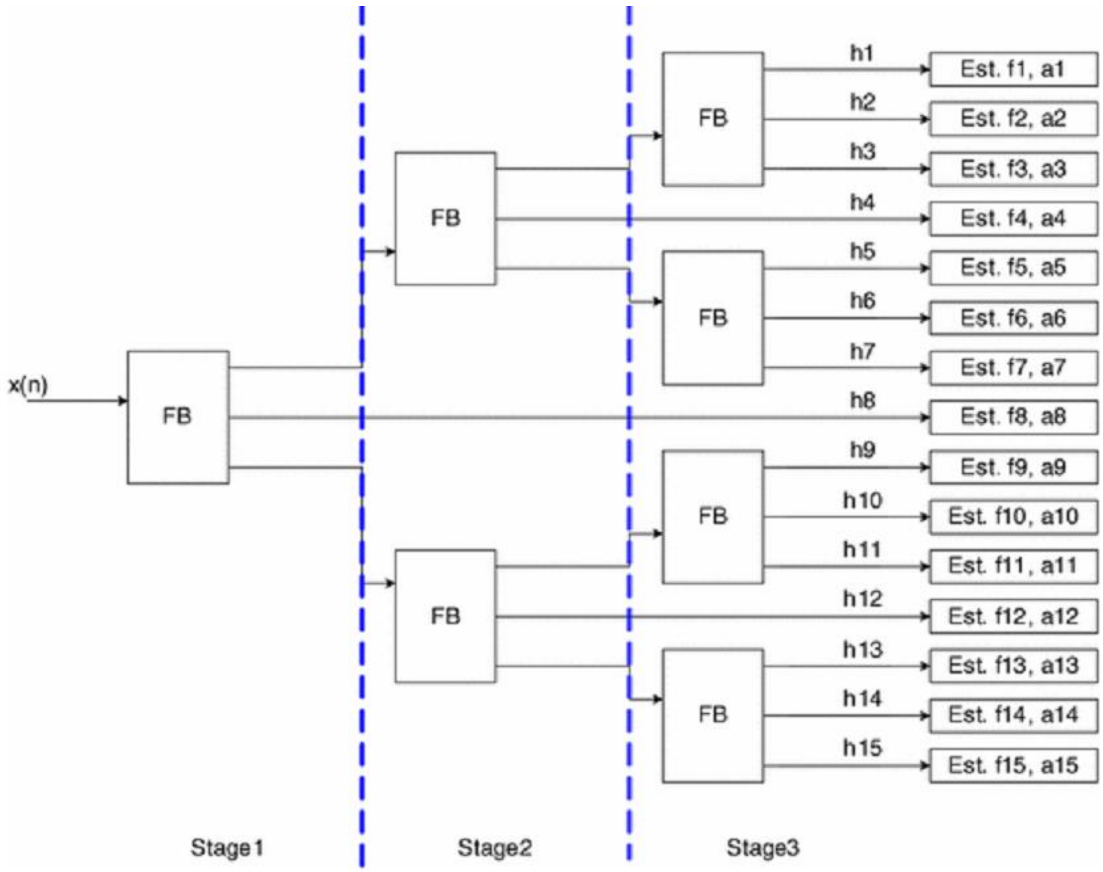

Fig. 3 Multistage FB for higher order harmonic decomposition

2.3. Self - Tuning Adaptive Filtering for Improved accuracy Harmonic Parameter Estimation

Adaptive filtering is used to estimate the parameters of time-varying power signals more accurately. In Fig. 4 [3], h_i (n) is the decomposed signal from the FB, $z-\Delta$ is the delay element and w (z) is the adaptive filter. Harmonic components are estimated from the output of the adaptive filter y(n), which is self-tuning [7].

Least Mean Square Algorithm is used in adaptive filtering. The sinusoidal model used is

$$x(n) = \sum_{i=1}^{15} a_i \cos(i\omega_1 n) \tag{8}$$

The above equation can be rewritten in discretized form as

$$x_k = real(a_i e^j \omega_k n) \tag{9}$$

If $x_k^{\wedge}$ is the estimated value of voltage at kth instant, equation (9) becomes

$$x_k^{\wedge} = real\ (w_k x_{k-1}^{\wedge}) \tag{10}$$

$$w_k = e^j w_k^{\wedge} n \tag{11}$$

where w_k is the weight of the voltage signal and $\omega^{\wedge}$ is the angular frequency.

The error signal is

$$e_k = x_k - x_k^{\wedge} \tag{12}$$

This algorithm minimizes the error signal by calculating the weight vector each time using

$$w_k = w_{k-1} + u_k e_k x_k^{*\wedge} \tag{13}$$

where * denotes the conjugate of the variable and step size u_k is varied for achieving faster convergence of the LMS algorithm.

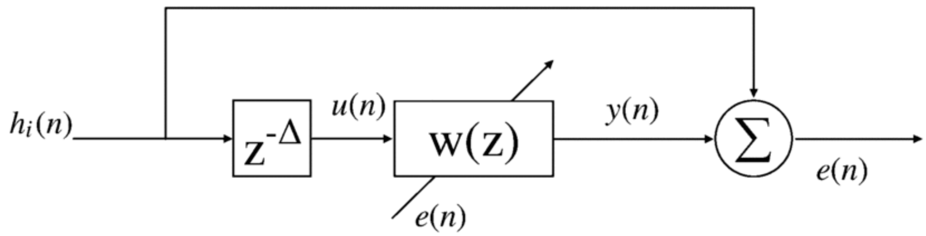

Fig. 4 Adaptive filtering for time-varying frequency estimation

3. Simulations for Performance Evaluation

A signal consisting of harmonic components is considered. The sinusoidal model used is

$$x(n) = \sum_{i=1}^{15} a_i \cos(i\omega_1 n) \tag{14}$$

where a_i is the magnitude and $i\omega_1$ is the harmonic/interharmonic frequency. The sampling frequency and fundamental used are 1.92 kHz and 50 Hz respectively. Simulations are also done for 50±3 Hz. The results obtained are

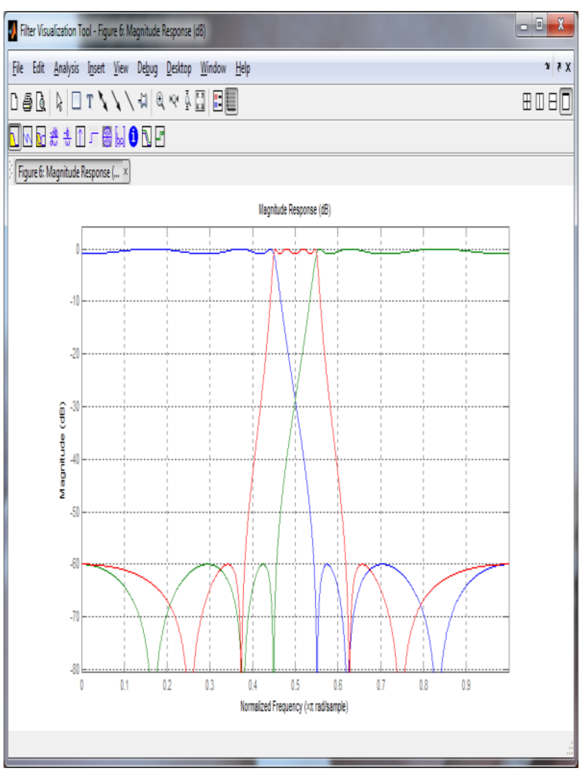

Table 3: Harmonic amplitude estimation with adaptive filtering

Table 1: Harmonic amplitude estimation without adaptive filtering

Fig.5 Magnitude response of low pass, high pass and band pass filter

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AMPLITUDES

Order	actual	47 Hz	50 Hz	53 Hz
1.0000	100.0000	98.0984	99.3434	95.1743
2.0000	1.5000	1.3181	1.4154	1.4127
3.0000	3.0000	2.8572	3.4767	3.2811
4.0000	0.3000	0.2414	0.2894	0.2893
5.0000	4.0000	3.7857	3.3541	4.9080
6.0000	3.0000	3.2351	3.2942	3.4973
7.0000	4.0000	4.2780	4.3232	3.8267
8.0000	1.0000	1.0708	1.2036	1.2181
9.0000	0.8000	0.7435	0.7561	0.7238
10.0000	0.3000	0.2768	0.2744	0.2773
11.0000	0.5000	0.5355	0.4560	0.6273
12.0000	2.0000	2.1060	1.8175	1.9241
13.0000	0.8000	0.8488	0.7185	0.7350
14.0000	2.5000	2.5424	2.4213	2.3450
15.0000	0.3000	0.3196	0.2618	0.3253

Table 2: Harmonic frequency estimation without adaptive filtering

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>> FREQUENCY

Order	47 Hz	50 Hz	53 Hz
1.0000	45.4358	48.5423	51.4335
2.0000	92.8716	98.0877	104.8670
3.0000	139.3074	147.6269	157.3005
4.0000	185.7432	196.1692	205.7340
5.0000	233.1790	248.7115	262.1675
6.0000	280.6148	297.2538	315.6010
7.0000	327.0506	346.7961	369.0345
8.0000	375.4864	397.3384	421.4680
9.0000	420.9222	446.8807	475.9015
10.0000	466.3580	497.4230	528.3350
11.0000	515.7938	546.9653	580.7685
12.0000	561.2296	598.5076	634.2020
13.0000	609.6654	646.0499	687.6355
14.0000	656.1012	698.5922	740.0690
15.0000	703.5370	748.1345	793.5025

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>> AMPLITUDES

Order	actual	47 Hz	50 Hz	53 Hz
1.0000	100.0000	98.2884	99.4934	95.2543
2.0000	1.5000	1.4032	1.4654	1.5237
3.0000	3.0000	2.9472	3.2267	3.0231
4.0000	0.3000	0.2652	0.3047	0.3043
5.0000	4.0000	3.8957	3.5417	5.0278
6.0000	3.0000	3.1243	3.0424	3.1373
7.0000	4.0000	4.1324	4.2732	3.8976
8.0000	1.0000	1.0218	1.1236	1.0306
9.0000	0.8000	0.7658	0.7851	0.7432
10.0000	0.3000	0.2868	0.2844	0.2877
11.0000	0.5000	0.5255	0.4760	0.5973
12.0000	2.0000	2.0062	1.9257	1.9842
13.0000	0.8000	0.8277	0.7835	0.7758
14.0000	2.5000	2.4347	2.3734	2.4674
15.0000	0.3000	0.2996	0.2837	0.3127

Table 2. Harmonic frequency estimation with adaptive filtering

Command Window

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>> FREQUENCY

Order	47 Hz	50 Hz	53 Hz
1.0000	46.5583	49.6322	52.3245
2.0000	93.7398	99.7897	105.8716
3.0000	140.7632	148.7423	158.4374
4.0000	186.8179	198.6782	207.7456
5.0000	234.2962	249.2675	263.6478
6.0000	281.5239	298.5478	316.6216
7.0000	328.6217	347.6423	370.4652
8.0000	375.9864	398.8297	422.6782
9.0000	421.3582	447.7834	477.1278
10.0000	467.7843	498.5768	529.2348
11.0000	516.3256	547.8574	581.6984
12.0000	562.5936	599.5674	635.2363
13.0000	610.6324	647.4379	688.3626
14.0000	657.3154	699.3245	741.2643
15.0000	704.3264	749.3462	794.0328

4. Conclusion

A method is described for the parameter estimation of harmonics using a FB and a self-tuning adaptive filter. FB is used to decompose the signal into odd and even harmonics. A fundamental FB is used successively in each stage for higher order harmonic decomposition. An adaptive filter is used to estimate the parameters more accurately. The method can be employed for both stationary and nonstationary signals.

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