ESTIMATION OF POWER SYSTEM FREQUENCY USING ADAPTIVE NOTCH FILTERS

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Abstract :

A new approach for frequency relaying in power system is presented in this paper. The approach consists of passing the power system voltage signal through a two stage adaptive notch filter, which produces an estimate of the fundamental frequency of the voltage waveform and its enhanced amplitude. The adaptive of the notch filter coefficients are obtained by a recursive least mean squares algorithm. The performance of the proposed algorithm is computationally efficient and it produces an accurate estimate of the fundamental frequency in the presence of harmonics and noise, which is a prerequisite to frequency relaying in power systems. Several computer simulation results are presented in the paper to show the effectiveness of the algorithm. The algorithm is simple and suitable for real time implementation. Further the accuracy of this approach in presence of harmonics and noise is improved considerably.

Introduction

For monitoring, control and protection of power systems using microprocessors, it is normally important to carry out on-line calculation of supply frequency and its variations. Variations in frequency from a normal value can be, for example, indicative of unexpected system disturbances for which some corrective action is to be taken.

If a voltage or current signal measured at a power system bus is a pure sinusoidal at fundamental frequency, then the measurement of system frequency by digital means or otherwise is simple and accurate. However, in reality, such measured signals are distorted due to presence of non-linear loads or sources such as rectifiers and inverters used in HVDC links. A large number of numerical techniques, dedicated to frequency measurement by digital means have been described [1,2], under the assumption of stationary statistics for the power system signal during the observation period. Amongst these Discrete Fourier transforms, Least Mean Squares (LMS), Least Squares, Kalman filters are known signal processing techniques [3,11], used for frequency measurement of signal whose amplitude or phase vary with time. All the above algorithms are very much influenced by the presence of harmonics, exponentially decaying dc component and random noise and are prone to error. Recently an iterative Newton's procedure combined with least square approach, is described [12] to produce fast and accurate estimation of power system frequency. However, this algorithm is less attractive due to the selection

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of a rigorous starting point and large computational overhead for real-time implementations.

Infinite Impulse Response (IIR) filter structure has been reported as an efficient tool for extraction of multiple sinusoids from a composite signal. These structure show substantially reduced number of computations and quick convergence. The notable examples are the *resonators-in-a loop* structure [13], *cascaded* IIR structure [14], and multi resolution techniques [15]. However, the problem associated with resonator-in a loop structure is that, the number of sinusoids present in the composite signal must be known correctly to have accurate frequency estimation becomes unviable due to the tremendous computational overhead.

In this paper we have investigated a modified version of the cascaded IIR structure with multirate sampling to estimate fundamental frequency of power system voltage waveform. The algorithm makes use of serial combination of two single notch filters in which the later uses a down sampled version of the enhanced signal obtained from the first filter.

Frequency Estimation Model

The power system is assumed to be a *quasi-steady state* representation which is having a strong fundamental component with time varying harmonics and a random noise sequence. We represent the power system voltage waveform of which the frequency is to be estimated as,

$$y(t)=\sum_{i=1}^{p} A_i(t).\sin(\omega_i t+\phi_i)+v(t) \qquad (1)$$

where *v(t)* is a random sequence and $A_i(t)$ $\phi_i(t)$ ω_i are the amplitude phase and frequency of the respective sinusoids present in the signal. The algorithm using adaptive IIR notch filter is described below :

Algorithm

Input signal : $y_i(t)$

Output signal : $e_i(t), y_{en}^i(t)$

For a second order notch filter the RML algorithm can be described as follows [14] :

The transfer function of a second order adaptive IIR notch filter is given by,

$$H_i(z)=\frac{1+a_i z^{-1}+z^{-2}}{1+ra_i z^{-1}+r^2 z^{-2}} \qquad (2)$$

where parameters a_i and r are adaptively tuned. The centre-frequency and 3dB rejection bandwidth *BW* are obtained as,

$$\omega_N^i=\cos^{-1}(-a_i/2)$$

$$BW=\pi(1-r) \qquad (3)$$

If the notch input is $y_i(t)$ and the output is $e_i(t)$ respectively,then a_i is updated by minimizing the cost function,

$$J(a_i,t)=\sum_{n=1}^{t}\lambda^{t-n}e_i(n) \qquad (4)$$

The notch parameter having the frequency information, i.e. a_i is adaptively tuned as per the following equations

$$a_{i+1}=a_i+p_i\psi_i e_i \qquad (5)$$

where,

$$p_{i+1}=p_i/(\lambda+p_i\psi_i^2) \qquad (6)$$

and, $\psi_i(t)$ is the pseudo-sensitivity function which is related to $y_i(t)$ as

$$H_s(z)=\frac{\psi(z)}{Y(z)}=\frac{z^{-1}(rH(z)-1)}{1+ra_i z^{-1}+z^{-2}} \qquad (7)$$

The enhanced output $y_{en}^{i}(t)$ is obtained as,

$$y_{en}^{i}(t) = y_i(t) - e_i(t) \qquad (8)$$

The notch is initialized with a r value of 0.8 and subsequently updated as,

$$r = r_0 . r + (1 - r_0) r_\infty \qquad (9)$$

where, r_0 and r_∞ are adaptation factor and final notch radius chosen for a particular application.

While dealing with a composite harmonic signal we only need to estimate the fundamental frequency because all other frequencies will be integral multiples of the fundamental frequency. Some useful observations while dealing composite harmonic signal like power system voltage waveform can be summarized as below :

- For power system signal it is pertinent to estimate the fundamental frequency only.
- It has been shown that a cascaded form of IIR notch structures converge to the frequencies of the sinusoids depending on their strengths. Hence, the notch structure would converge to the fundamental frequency first and other notches would converge subsequently; as the fundamental signal power is invariably larger than any of the harmonic powers; so it is worthwhile to use only a single notch.
- The excursion of coefficient a_i which is related to normalized 0dB or the centre-frequency f_i^n as $a_i = -2\cos(2\pi f_i^n)$ is limited between $\pm 10\,\text{Hz}$ of the nominal power system frequency. This is because of that fact that power system frequency hardly crosses such limits and the system is directed to trip for security reasons.

We, therefore propose a single notch filter which is adaptively tuned and the notch centre-frequency is bounded between $\pm 10\,\text{Hz}$ of the nominal power system frequency. The notch parameters settle down to the local minima asymptotically. To get an accurate estimate of frequency the radius of the notch filter r must be brought close to unity radius notch filter and instead use a second notch filter which use the enhanced output of the first notch to obtain a more exact frequency estimate. The second notch filter can be realized by using the sensitivity output $\psi_i(t)$ and one more adder [14]. Thus additional computational burden is almost negligible.

Further, the multi resolution technique[15] which is known for its frequency enhancement capabilities can be effectively used here. Here, the subsequent notch stages are fed with down-sampled version of the enhanced signal obtained from the previous stage. The general multirate adaptive notch is described in Fig.3. If a signal *y(t)* having a z-transform *Y(z)* is down-sampled by a factor D then the resulting signal $Y_D(z)$ is obtained as,

$$Y_D(z) = \frac{1}{D} . \sum_{k=0}^{D-1} Y(z^{\frac{1}{D}} w^{-k}) \qquad (10)$$

Thus, if the original signal has a spectrum of 2π/D after down-sampling has the spectrum width of 2π, thus making it easier to detect. However, the down-sampling factor must be carefully chosen to retain proper adaptability and selectivity of the filter. A large down sampling factor results in degradation of the adaptation convergence rate since the coefficients are adapted in every D samples. Also, the effective radius of the notch filter $r_{eff} = r^{\frac{1}{D}}$ thereby restricting adaptability. However, the convergence rate can be retained by updating the second notch filter at the same rate as the sampling rate. This is done by replacing z^{-1} by z^{-D} in the equations (1-8). The realization without altering the sampling frequency is shown in Fig.4. A sinusoid having frequency f_0 will be mapped to a down-sampled estimate f_D as,

$$f_D = D f_0 \bmod(2\pi) \qquad (11)$$

Thus knowing f_D, f_0 can be calculated as,

$$f_0 = \left(\frac{k}{D} + \frac{f_D}{D}\right) \qquad (12)$$

where, k = $INT\,[D f_D]$

and, *INT*[ ] denotes the nearest integer value towards zero.

Simulation Results

We show some simulated but realistic tests for the above mentioned algorithm.

Tracking Non stationary Signals

A. The input signal *y(t)* is defined as

y(t)=

$$1.0\sin(\omega_0 t + \pi/6) + 0.03.\sin(3\omega_0 t + \pi/3)$$
$$+0.03.\sin(5\omega_0 t + \pi/9) + 0.01.\sin(1\omega_0 t + \pi/12)$$
$$+0.01.\sin(13\omega_0 t + 5\pi/12) + 0.1.rand(t)$$

where, *rand(t)* is a white noise sequence having zero-mean and unity variance.

Where,

$$\omega_0 = \begin{cases} 100\pi & for\ \ t < 0.078125 \\ 102\pi & for\ \ 0.078125 \le t < 0.15625 \\ 100\pi & for\ \ t > 0.15625 \end{cases}$$

The resulting frequency estimate is shown in Fig.5(i), from the figure it is clear that though the 1st stage was able to reach near the actual frequency, the estimate may not be suitable for most of the pratical applications. The 2nd notch structure which takes the enhanced signal from the first stage and uses a down-sampling factor of 8, shows considerable improvement over the first stage.

B. In another case the frequency is ramed after for t ≥ 0.078125 , and ω_0 is given as,

$$\omega_0 = \begin{cases} 100\pi & for\ \ t < 0.07125 \\ 100\pi - 100(t - 0.078125) & for\ \ t \ge 0.078125 \end{cases}$$

In this case, the first stage shows steady state error, though it is quick to respond to the ramp. The 2nd stage predictably takes time to respond but show excellent steady state response. The frequency estimate for this case is shown in Fig. 5(ii).

C. In the third simulation test, we take-up a single-machine infinite bus system having a non-linear load and an induction machine. The mechanical power is suddenly changed at t=0.25, resulting in terminal voltage and frequency oscillations. Fig.5(iii) shows the actual frequency and tracked frequencies by the two stages of the notch filter. The corresponding system is shown in Fig.5(iv). The frequency tracking in this case can be considered satisfactory, because the machine terminal voltage also shows transient variations.

Conclusion :

In this paper we have presented a two-stage adaptive notch filter structure for estimation of power system frequency. The approach uses only two notch structures and multirate technique. The technique gives fair amount of accuracy while tracking fundamental frequency from power system voltage signal distored by harmonics and random noise. The algorithm described is computationally efficient and hence suitable for real-time frequency relaying. The new approach yields the system frequency in nearly one cycle of the fundamental waveform. This procedure can be suitably extended for harmonic measurements and power quality problems.

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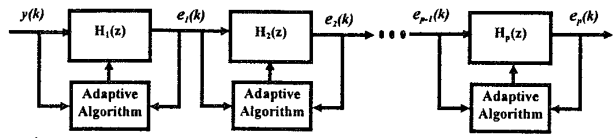

Fig.1 : Cascaded Notch structure

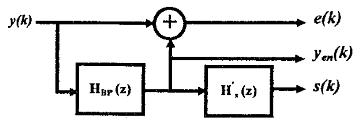

Fig.2 : A single notch realization

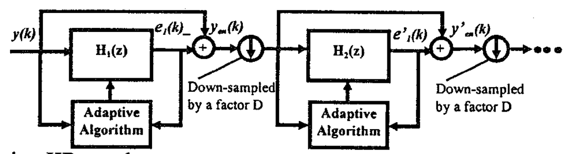

Fig.3 : Multirate adaptive IIR notch structure.

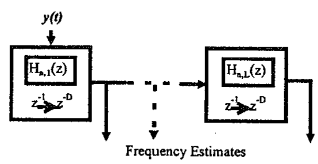

Fig. 4: Adaptive IIR notch filter without sampling rate alteration.

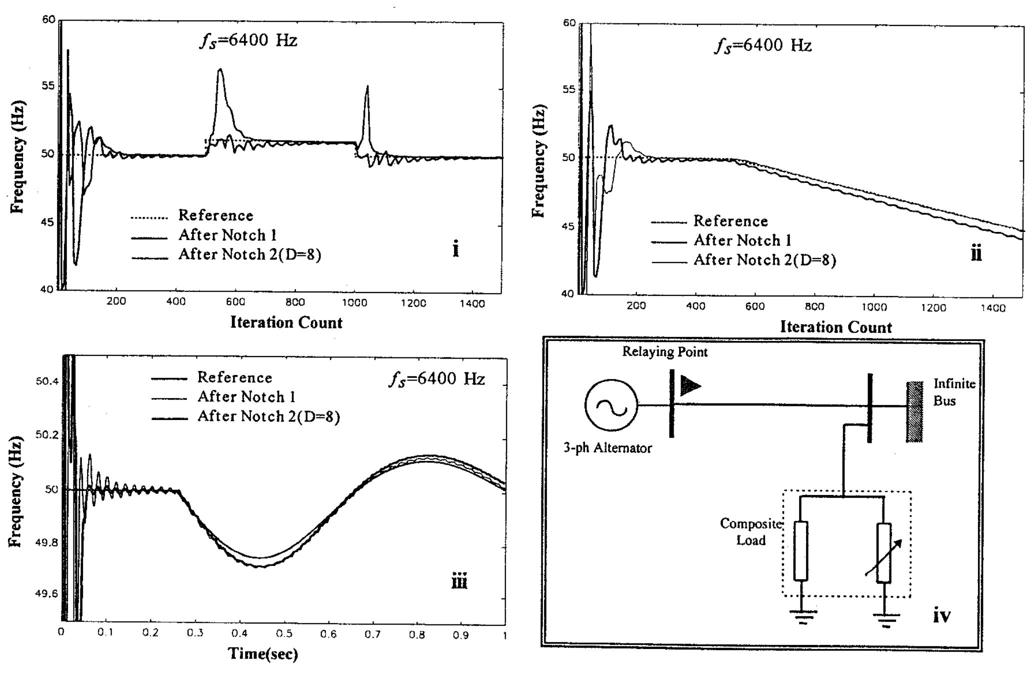

Fig 5: Simulation Test Results . Figures show tracking of (i) step changes, (ii) a ramp change and, (iii) oscillatory changes in system frequency. Fig. 5(iv) shows the model used for simulating the reference voltage and frequency used in Example C (Fig. 5(iii)).