An Iterative Technique for Fast and Accurate Measurement of Power System Frequency

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Abstract: This paper describes the design, computational aspects and implementation of an iterative technique for measuring power system frequency. The technique provides accurate estimates to a resolution of 0.01-0.02 Hz for near-nominal, nominal and off-nominal frequencies in about 20 ms. Computation requirements are modest and the technique has been implemented on a modern digital signal processor. The proposed technique was extensively tested using voltage signals obtained from a dynamic frequency source and from a power system. Some test results are presented in the paper.

Keywords: Power system frequency, Frequency measurement, Frequency relaying, Microprocessor-based relays.

I. INTRODUCTION

Frequency is an important operating parameter of a power system. A number of frequency measurement algorithms suitable for microprocessor-based relays have been proposed in the past. These algorithms use a variety of approaches including Discrete Fourier Transform and recursive phasor phase change [1], least error squares technique [2,3], Kalman filtering [4]. Recently, a Newton-type algorithm has been proposed [5] which requires considerable computational resources and, therefore, cannot be economically implemented with the presently available technology. For generator protection, where frequency is to be estimated over a wide range, the use of adaptive sampling intervals is suggested in [6] and an algorithm with a variable window length is proposed in [7]. An adaptive algorithm for measuring power system frequency over a wide operating range is suggested in [8]. However, this algorithm is adversely affected by presence of harmonics at non-nominal frequencies and, therefore, the use of pre- and post-filters are suggested which results in a slower response.

This paper presents an iterative technique for frequency estimation that provides accurate estimates in about 20 ms and requires modest computations. The proposed technique is capable of estimating frequency over a wide operating range. Theoretical basis, analysis, practical issues and implementation of the technique are described. The proposed technique was tested using voltage signals obtained from a dynamic frequency source and from the SaskPower system. Test results demonstrating the performance of the proposed technique are included.

PE-778-PWRD-0-04-1997 A paper recommended and approved by the IEEE Power System Relaying Committee of the IEEE Power Engineering Society for publication in the IEEE Transactions on Power Delivery. Manuscript submitted February 13, 1996; made available for printing March 26, 1997.

II. ERROR ANALYSIS AND BASIS

The real and imaginary parts of the fundamental frequency component are used to compute its peak value and phase angle. It is also possible to use consecutive phase angle estimates for computing the frequency. Generally, the orthogonal filters used for estimating the real and imaginary parts are designed by assuming the nominal frequency (60-Hz or 50-Hz) as the fundamental frequency. However, use of these filters will cause errors in the phasor and frequency estimates when the fundamental frequency deviates from the assumed nominal frequency. This is illustrated as follows:

Consider that two orthogonal digital filters, shown in Fig. 1, provide outputs of $K_s A\sin(wt)$ and $K_c A\cos(wt)$ for an input signal of $A\sin(wt+\theta)$. K_s and K_c are the gains of the filters corresponding to the frequency of w rad/s.

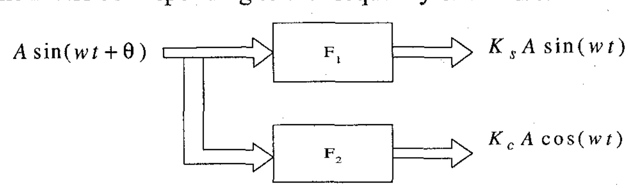


Fig 1. Output of the orthogonal digital filters.

The output of the orthogonal filters can be combined as follows to determine the estimated peak value, A_e.

$$A_e = \{[K_s A\sin(wt)]^2 + [K_c A\cos(wt)]^2\}^{1/2} \tag{1}$$

This can be further reduced to

$$A_e = \left\{ \left[\frac{(K_s^2 A^2 + K_c^2 A^2)}{2}\right] + \frac{1}{2}(K_c^2 A^2 - K_s^2 A^2)\cos(2wt) \right\}^{1/2} \tag{2}$$

The peak value estimates are correct for frequencies at which $K_S = K_C = 1$. At other frequencies, the outputs are timevariant. Estimates will be in the range of $K_s A$ and $K_c A$. The orthogonal filters are designed to have gains equal to one at the nominal frequency. Therefore, the estimates for signals having non-nominal frequencies will be in error. Fig. 2(a) shows the range of estimated peak values of the signals having a peak value of 1.0 and frequencies between 40-70 Hz. The peak values are estimated by using orthogonal filters whose frequency responses are given in Fig. 2(b). It can be observed that peak value estimate for 60 Hz signal is not in error because the gain of orthogonal filters is equal to 1 at 60 Hz.

The outputs of the orthogonal filters can also be used to estimate the phase angle, θ_e, as follows:

$$\theta_e = \tan^{-1}\left[\frac{K_s A}{K_c A}\tan(wt)\right] = \tan^{-1}\left[\frac{K_s}{K_c}\tan(wt)\right] \qquad (3)$$

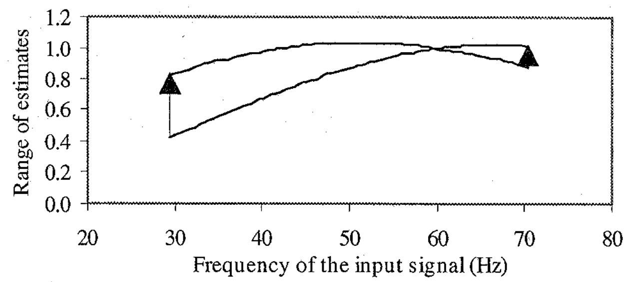
It is clear from Eqn. 3 that phase angle estimate is correct when K_S and K_C are equal. Therefore, at non-nominal frequencies, the estimates will be in error. This also has implications for frequency estimation procedure which uses rotation (phase change) of the phasor to estimate the frequency. At non-nominal frequencies, the difference of phase angle from one sampling instant to the next sampling instant will not provide the correct value of the phasor rotation and, therefore, frequency estimates will be in error. Fig. 3 shows the estimated frequency for a signal of 55-Hz when orthogonal filters having frequency responses of Fig. 2(b) are used. Some techniques use averaging filters to reduce the oscillations in the frequency estimates, however, this increases the measurement time.

Presence of harmonics in the signals will cause further errors because the filters may be designed to eliminate harmonics of the nominal frequency and there are nulls at these frequencies in the frequency response of these filters as is seen in Fig. 2(b). However, when the frequency deviates from the nominal frequency, the harmonics of non-nominal frequencies are not eliminated by these filters and, therefore, will cause errors in the phasor and frequency estimates.

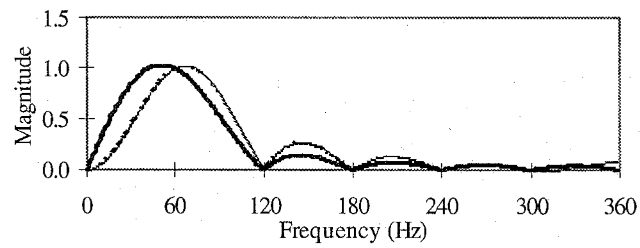
From the above discussion, it is clear that a possible solution is to adjust the orthogonal filters in response to the frequency deviation. The filters should be adjusted such that they provide a gain of unity at the existing fundamental frequency and nulls at its harmonic components. If this could be achieved, the phasor and frequency estimates would be correct. Also, the window length should not change for ease of implementation and sampling rate should be kept constant allowing use of synchronised sampling. A technique using the fore-mentioned premise has been developed.

III. THE PROPOSED TECHNIQUE

The technique considers that samples of a signal having fundamental frequency of f_0 Hz are taken at a pre-specified sampling rate. It is possible to design orthogonal filters for



(a)



(b)

Fig. 2. (a) Range of peak value estimates (b) Frequency response of orthogonal filters.

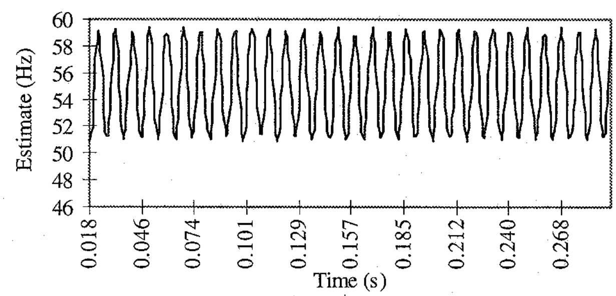


Fig. 3. Frequency estimates of a 55 Hz input signal by using filters whose frequency responses are shown in Fig. 2(b).

extracting the real and imaginary parts of the fundamental frequency component of the signal. The real and imaginary parts, V_{rn} and V_{in}, computed using samples corresponding to the *nth* data window can be used to estimate the phase angle, θ_n of the fundamental frequency phasor corresponding to the *nth* window by using Eqn. 4.

$$\theta_n = \tan^{-1}(V_{in} / V_{rn}) \qquad (4)$$

As the next sample arrives, the data window is shifted by one sample. The phase angle, θ_{n+1}, of the fundamental frequency phasor corresponding to the *(n+1)th* data window can be computed using the data from the *(n+1)th* window and coefficients of the orthogonal filters. The phase angle difference, $(\theta_{n+1} - \theta_n)$, represents the rotation of the phasor in one sampling interval. This rotation for, a phasor having fundamental frequency of f_0, will equal to

$$(\theta_{n+1} - \theta_n) = (2\pi f_0)/f_s. \qquad (5)$$

Eqn. 5 is only true if the phase angles, θ_{n+1} and θ_n, are computed by using the orthogonal filters which are designed by assuming the fundamental frequency of the signal being equal to f_0. In spite of this, Eqn. 5 forms the basis for frequency estimation and estimate of the frequency, $\hat{f}$, can be obtained as follows:

$$\hat{f} = (\theta_{n+1} - \theta_n)/(2\pi / f_s) \qquad (6)$$

One of the following two situations can exist:

(A) The estimated frequency is equal to the fundamental frequency assumed for designing the orthogonal filters that are used to compute phase angles, θ_n and θ_{n+1}. This means that the estimated frequency is also the fundamental frequency of the signal.

(B) The estimated frequency is not equal to the fundamental frequency assumed for designing the orthogonal filters. This means that the estimated frequency is not the fundamental frequency of the signal. However, to achieve Situation A, the phase angles need to be calculated using orthogonal filters designed by assuming the fundamental frequency being equal to the fundamental frequency of the signal. This can be achieved by using an iterative procedure as follows:

(i) Design new orthogonal filters by assuming the fundamental frequency of the signal being equal to the latest estimate of the frequency obtained from Equation 6.

(ii) Compute the phase angles θ_n and θ_{n+1} by using the orthogonal filters designed in step (i) and the samples corresponding to data windows *n* and *(n+1)*.

(iii) Estimate the frequency using Equation 6 and the phase angles computed in step (ii).

(iv) Check if the estimated frequency from step (iii) is equal to the fundamental frequency assumed for designing the filters in step (i). If it is, the estimated frequency in step (iii) is the fundamental frequency of the signal. Otherwise, the process reverts to step (i).

It must be noted that an initial estimate of the frequency is assumed for starting the procedure for the first time.

IV. PRACTICAL ISSUES

The technique for estimating power system frequency is explained in the previous section. Practical considerations for implementing the proposed technique are:

1. As has been discussed previously, the estimation process may require design of new orthogonal filters at every iteration. Design of filters requires considerable amount of computations. These computations may not be completed within one sampling interval which is available for performing calculations. Therefore, the filters should be designed off-line and their coefficients be stored for use in estimating the frequency.

2. The iterative procedure for estimating frequency is terminated when the estimated frequency obtained from Eqn. 6 is equal to the fundamental frequency assumed for designing the orthogonal filters. However, in practice, a margin should be allowed to account for errors arising from truncations during calculations, data acquisition etc.

3. Number of iterations which can be performed in one sampling interval will be limited by the capability of the digital processor.

V. COMPUTATIONS

The technique performs computations in off-line and on-line modes.

A. Off-Line Computations

Off-line calculations include the design of orthogonal filters that assume the fundamental frequency of the signal being equal to 34 Hz to 75 Hz in steps of 1 Hz. The coefficients of the resulting sine and cosine filters are stored in the form of a look-up table for use during the on-line calculations. This means that the coefficients of filters corresponding to the above mentioned frequencies are directly available, however, for filter coefficients corresponding to other frequencies, interpolation must be performed. It was found that this arrangement facilitates the estimation of frequencies in the range of 40-70 Hz by using the proposed technique. This range covers the frequencies that can exist during the power system operation, however, it can be extended by storing coefficients of filters covering a wider range.

B. On-line Calculations

The digitized values of samples of the signal taken at a pre-specified rate are the inputs. Following steps are performed to estimate the frequency of the signal:

1. Obtain a new sample representing the signal and store it in the processor's memory.
2. Check if the all the samples representing the first two windows have been acquired. If they are, proceed to step 3; otherwise revert to step 1.
3. Assume the value of the initial estimate of the signal frequency and assign this value to a variable, FL. Also, set the iteration counter to be 1.
4. Obtain the coefficients of the filters corresponding to the frequency assigned to variable FL. For some values of the frequency, the coefficients are available directly, and for other values of frequency, interpolation must be performed to determine the coefficients.
5. Compute the real and imaginary parts of the signal corresponding to the latest two windows by using coefficients of the filters determined in step 4.
6. Calculate the phase angles corresponding to the latest two windows. Then, estimate the signal frequency using Eqn. 6 and update the value of variable FL with the estimated frequency. However, if the estimated frequency is outside the range for which filter coefficients are stored, the variable,

FL, is assigned the same value as in step 3 i.e. the initial estimate of the signal frequency.

7. Check if the absolute value of the difference between the estimated frequency obtained in step 6 and the frequency used for obtaining the coefficients of the filters in step 4 is less than a pre-specified threshold. If it is, the signal frequency is the estimated frequency and proceed to step 9; otherwise, proceed to step 8.

8. Increment the iteration counter by one. Check if the iteration counter is less than a pre-specified value, if it is, revert to step 4; otherwise, proceed to step 9.

9. Reset the iteration counter to be 1. Wait for the next sample and revert to step 4 after obtaining it.

VI. PARAMETER SELECTION

The proposed technique was programmed using a digital computer in MATLAB programming environment. The orthogonal filters were designed by using the Least Error Squares (LES) technique [9] and by assuming the fundamental frequency of the signal being equal to 34 to 75 Hz, in steps of 1 Hz. The following design parameters were chosen.

- a sampling rate of 720 Hz
- window length of 13 samples
- time reference coinciding with the middle of the window
- signal is assumed to contain dc, fundamental frequency and harmonic components up to 5th harmonic of the fundamental frequency

The coefficients of the LES filters were stored in a tabular form for use by the on-line portion of the technique.

Analysis was conducted using software generated data. The data were samples, taken at 720 Hz, of the synthesized voltages having frequencies in the 40 to 70 Hz range. No intentional noise was added as these data were used to study the characteristics of the proposed technique. Effects of the initial estimate and number of iterations were studied and are reported in this section.

A. Initial Estimate

On-line calculations of the technique start by assuming an initial estimate of the fundamental frequency of the signal. This estimate has an effect on the time taken by the technique to converge to the true value of the signal frequency. This aspect was studied by using the proposed technique to estimate the frequency of 40, 50 and 60 Hz signals when initial estimates of 40, 50 and 60 Hz are used. The number of iterations per sampling interval were kept to be 1 for all cases so that this parameter does not bias the results.

Figure 4 shows the estimates of frequency when an input signal of 40 Hz and initial estimates of 40, 50 and 60 Hz are used. Figures 5 shows estimates of the frequency when input signals of 70 Hz frequency, and initial estimates of 40 50 and 60 Hz are used. It can be seen from Figures 4 and 5 that all estimates converge to the true values. However, it takes about 3-4 extra sampling intervals to converge when initial estimate of the frequency is not close to the frequency of the input signal. Further testing using input signals of other frequencies and different initial estimates confirmed this observation. However, it must be kept in mind that this conclusion is valid only when one iteration is performed in each sampling interval. The effect of number of iterations is reported in the next section.

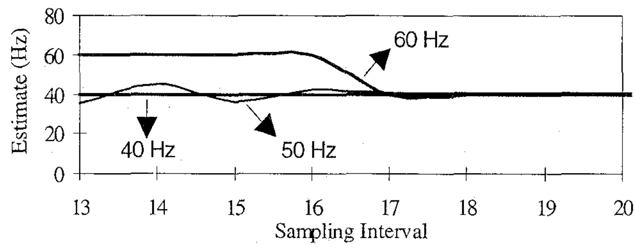


Fig. 4. Frequency estimates of an input signal of 40 Hz when initial estimates of 40-, 50- and 60-Hz are used.

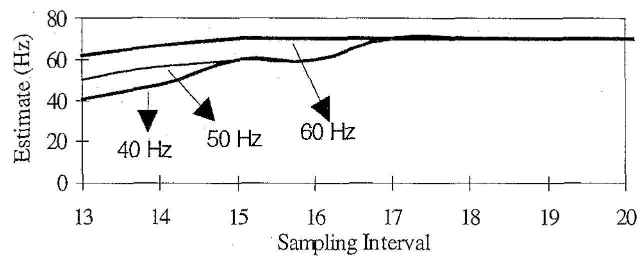


Fig. 5. Frequency estimates of an input signal of 70 Hz when initial estimates of 40-, 50- and 60-Hz are used.

B. Number of Iterations per Sampling Interval

Computing power of a digital processor limits the number of iterations that can be performed in one sampling interval. However, this impacts the frequency tracking capability and convergence of estimates. This aspect was studied by estimating the frequency of signals when number of iterations that can be performed in one sampling interval are kept to be 1, 3, 5 and 7. Initial estimate was assumed to be the nominal frequency of the system i.e. 60 Hz.

Figure 6 shows the frequency estimates when input signal has a frequency of 65 Hz. It can be seen from this figure that the time for convergence decreases as the number of iterations increase. However, there is no further significant improvement when the number of iterations are increased beyond 5. Similar observations were made when input signals of other frequencies were used. It was also seen that the frequency estimates for an input signal of 60 Hz converge in the same time irrespective of the number of iterations per sampling interval. This is due to the initial estimate being equal to the signal frequency.

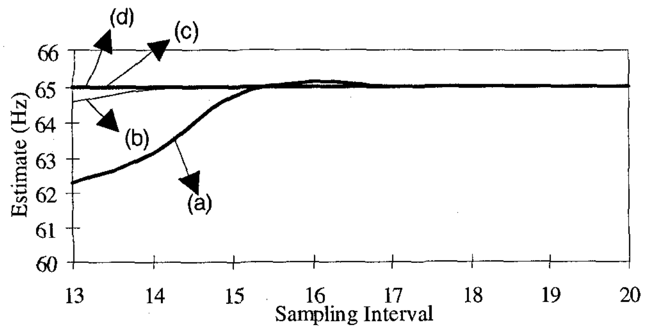


Fig. 6. Frequency estimates for an input signal of 65 Hz when number of iterations per sampling interval are set equal to (a) one (b) three (c) five and (d) seven.

C. Recommended Values

It is clear from the above analysis that convergence is achieved faster and less number of iterations are needed when initial estimate is closer to the frequency of the signal. However, without any prior knowledge, it is recommended that an initial estimate equal to the nominal frequency be used.

As seen earlier, for a step change in the signal frequency, the number of iterations should be kept at five to achieve faster convergence. However, for situations when the estimate is close to the signal frequency, the number of iterations per sampling interval can be less than five. In fact, this would be the case for most situations because the algorithm will be continuously tracking the frequency. Therefore, if a digital processor cannot perform five iterations in one sampling interval, the number of iterations per sampling interval can be set at less than five depending on the capability of the processor. The only effect of this is that it may take one or two additional sampling intervals for convergence in situations where there is a step change in the frequency or initial estimate is not very close to the signal frequency.

VII. IMPLEMENTATION

The proposed technique was implemented on a DSP-based general-purpose hardware which has been designed and developed at the University of Saskatchewan for implementing microprocessor-based relays [10]. The hardware consists of isolation and analog scaling and microcomputer blocks. The isolation and analog scaling block consists of eight pluggable modules; four modules for processing voltages and four modules for processing current signals. The functions of this block are to reduce the levels of the signals, convert current signals to equivalent voltage signals and isolate the microcomputer block from the power system. The microcomputer block consists of two data acquisition cards, a DSP card and MS-DOS-based host computer. Each data acquisition card can sample and digitize four signals. The DSP card is based on Texas Instruments TMS32030 floating point digital signal processor.

Data acquisition and implementation software was written in 'C' language and, after compilation and assembly it was downloaded to the DSP card. Floating-point arithmetic was used in the implementation. Bench-mark tests were performed to determine the execution time of the software implementing the proposed frequency estimation technique. It was found that the software executed in about 0.33 ms when the number of iterations performed in one sampling interval was set at three. This is well within the available inter-sampling time of (1/720) s i.e. 1.3889 ms and leaves ample time to perform other functions. Software was, therefore, set to perform three iterations in a sampling interval.

VIII. TEST RESULTS

After implementing the proposed technique, it was tested to determine its measurement accuracy and speed. Effectiveness of the technique was checked for frequencies in the range of 40 Hz to 70 Hz. The test signals were obtained from a dynamic frequency source as well as from a power system.

The output of the dynamic frequency source can be adjusted from 0 to 140 V with a frequency in the range from 40 to 79.99 Hz. The voltage was set at 110 V and frequency was selected in the 40 to 70 Hz range. The voltage level was reduced to 3V peak value using a module of the isolation and analog scaling block of the general-purpose hardware. The reduced voltage signal was applied to a channel of the data acquisition card. The signal was sampled at 720 Hz and digitized using a 5V, 12-bit A/D converter of the data acquisition card. The voltage magnitude and frequency of the dynamic frequency source are accurate within one and 0.001 percent respectively. The A/D converter used has an accuracy of 0.025 percent in the full range. The voltage signals from the SaskPower system were also reduced in level, quantized at 720 Hz and digitized. The digitized values were stored in the memory of the DSP board and were used by the software that implemented the proposed frequency measurement technique. Some results are discussed in the following paragraphs.

Figure 7(a) shows the frequency estimates of a 59 Hz signal. The figure shows that the frequency is measured to resolution of 0.01 Hz with a delay of about 20 ms. The fluctuations about the true value are due to noise in the input data. If better resolution is desirable, this effect of noise could be further reduced by using appropriate smoothing filters or by increasing the data window length selected for designing LES filters. However, it must be noted that these actions will increase the measurement time. Figures 7(b) and 7(c) show the frequency estimates of 52 Hz and 65 Hz signals. It can be seen from the figures that off-nominal frequencies are measured to a resolution of 0.02 Hz with about 20 ms delay.

Figure 8 shows the estimated frequency when data recorded from a power system was used. As the true frequency of the power system was not known, therefore, estimation errors cannot be specified. However, the quality of estimates can be judged from the their consistency. The maximum variation in the estimates is about ± 0.01 Hz.

It can be seen from Figures 7 and 8 that the measurement time to provide frequency estimates to a resolution of 0.01 Hz for near-nominal frequencies and 0.02 Hz for off-nominal frequencies is about 20 ms. Above results represent an improvement, in terms of accuracy and speed, in comparison to the results previously reported in the literature. For example, Reference 8 reports a resolution of 0.01-0.04 Hz with a measurement time of 32 ms under steady-state conditions.

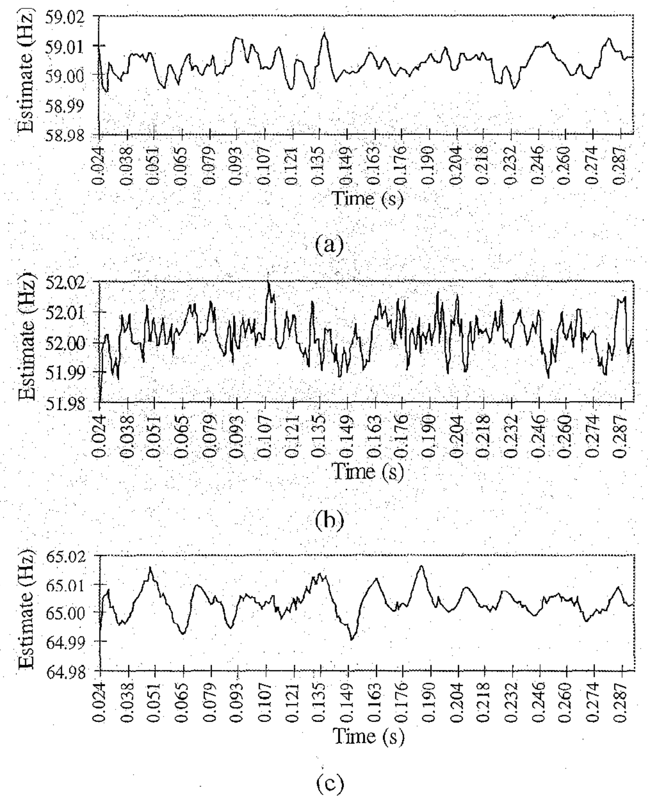


Fig. 7. Frequency estimates of input signals of (a) 59 Hz (b) 52 Hz and (c) 65 Hz.

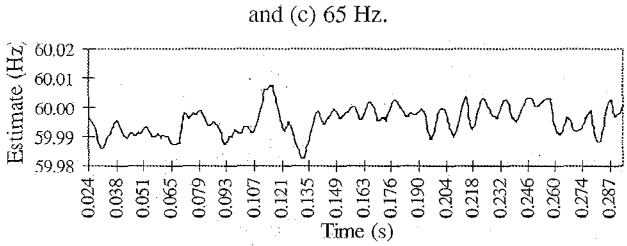


Fig. 8. Frequency estimates of a voltage signal recorded from the SaskPower system.

A. Performance in Presence of Harmonics

To demonstrate the effectiveness of the proposed technique for estimating frequencies in presence of harmonics, an input signal having fundamental frequency of 61 Hz, a 20% second harmonic component (122 Hz) and a 10% third harmonic component (183 Hz) was synthesized. Frequency of the synthesized signal was estimated by using the proposed technique. Figure 9 shows the voltage signal and the frequency estimates. The estimated values are equal to the true value despite the presence of harmonic components. If presence of other non-harmonic components is anticipated, the LES filters to be used in the technique should be designed by including these components in signal model.

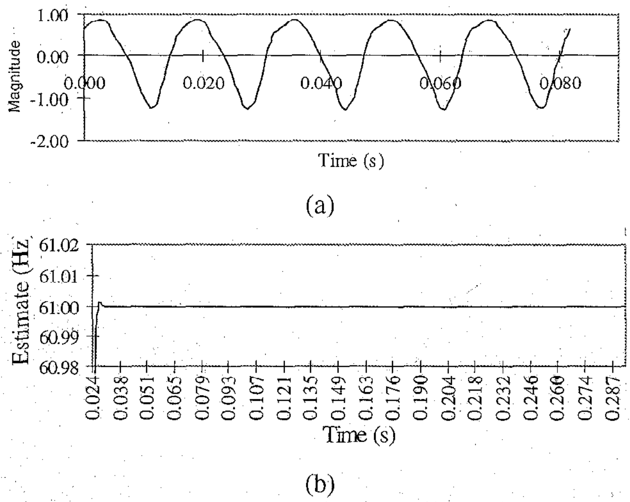


Fig. 9. (a) A 61 Hz voltage signal containing second and third harmonics and (b) its frequency estimates.

B. Performance During Frequency Decay/Rise

Design of the proposed technique assumes that frequency of the signal remains constant within a data window. However, this may not be the case when power system frequency is decaying or rising. The effectiveness of the proposed technique during frequency decay/rise was tested by synthesizing a signal whose frequency oscillates by ± 3 Hz at a frequency of 5 Hz starting from 60 Hz. The synthesized signal was sampled at the rate of 720 Hz. The sampled data were used by the proposed technique for estimating the frequency of the signal. Figure 10 shows the actual frequency of the signal and the frequency estimates given by the proposed technique. It can be seen from this figure that the estimates provided by the proposed technique follow the signal frequency extremely closely with a delay of about 20 ms. The tests were repeated by using other decay and rise rates and similar results were obtained.

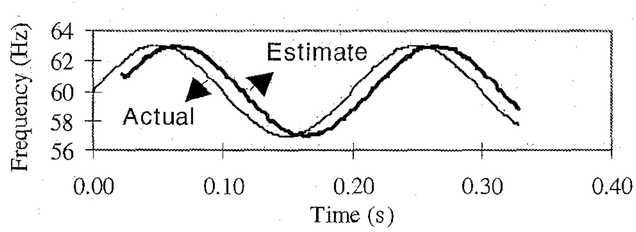


Fig. 10. Actual and estimated frequency of an input signal having dynamic frequency oscillating at 5 Hz.

C. Performance in Presence of Noise

Figs. 7 and 8 show the frequency estimates of signals that contain noise levels normally expected in the power systems. The effect of noise introduced by the data acquisition system is also present. Performance of the technique was further studied by estimating the frequency of signals that contained varying degree of noise. Signals of 50, 60 and 70 Hz were synthesized and a varying degree of random noise having zero-mean and Gaussian distribution was added to them. Fig. 11 shows the maximum errors observed in frequency estimates when input signals of 50, 60 and 70 Hz having SNR of 40, 50, 60, 70 and 80 dB were used. Measurement time for results shown in Fig. 11 is about 20 ms. It is clear that errors increase with the amount of noise. However, as noted earlier, the proposed technique provides accurate estimates for noise levels normally expected in the power system signals (Fig. 8). If higher levels of noise are expected, strategies like use of smoothing filters or larger data windows can be used to achieve desirable resolution.

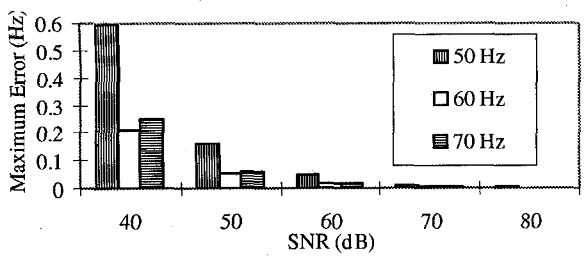


Fig. 11. Maximum estimation errors for noisy input signals.

IX. CONCLUSIONS

Design, practical issues, computational aspects and implementation of an iterative technique for measuring the operating frequency of a power system have been described in the paper. The proposed technique is suitable for measuring near-nominal, nominal and off-nominal frequencies. The beneficial features of the proposed technique include fixed sampling rate, fixed data window size and easy implementation. Fixed sampling rate makes the use of this technique attractive for synchronized measurement systems. A by-product of this technique is accurate phasor measurements when system frequency deviates from the nominal frequency.

Performance test results, that used signals from a dynamic frequency source and from a power system, indicate that the proposed technique can provide accurate estimates within about 20 ms. Maximum estimation errors observed during testing are of the order of 0.01 Hz for nominal and near-nominal frequencies and are in the range of 0.02 Hz for off-nominal frequencies. The technique requires modest computations and is suitable for use in microprocessor-based relays.

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XI. BIOGRAPHIES

Tarlochan S. Sidhu received the B.E. (Hons.) degree from the Punjabi University, Patiala, India in 1979 and the M.Sc. and Ph.D. degrees from the University of Saskatchewan, Saskatoon, Canada in 1985 and 1989 respectively. He worked for the Regional Computer Center, Chandigarh, India from 1979 to 1980 and for the Punjab State Electricity Board, India from 1980 to 1983 After obtaining the Ph.D. degree, he joined Bell-Northern Research Ltd., Ottawa, Canada and worked on a software development project for about one year. He joined, in 1990, the University of Saskatchewan where he is presently Associate Professor in the Department of Electrical Engineering. His areas of research interest are power system protection, monitoring and control.

Mohindar S. Sachdev was born in Amritsar, India, in 1928. He received the B.Sc. from the Baneras Hindu University, India, the M.Sc. degrees from the Punjab University, Chandigarh, India and the University of Saskatchewan, Saskatoon, Canada and, the Ph.D. and the D.Sc. degrees from the University of Saskatchewan, Saskatoon, Canada. He worked for the Punjab P.W.D. Electricity Branch and the Punjab State Electricity Board, India from 1950 to 1968. In 1968, he joined the University of Saskatchewan where he is currently Professor Emeritus of the Electrical Engineering. His areas of interest are power system analysis and power system protection. He is a Fellow of the Institution of Engineers (India) and a Fellow of the Institution of Electrical Engineers, London (England). He is also a Registered Professional Engineer in the Province of Saskatchewan and a Chartered Engineer in the UK.