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A Method of Least Squares Error with Iterative Resampling for Frequency Tracking in Smart Grid

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ABSTRACT

Phase angle and amplitude of a phasor can be provided by method of Least Squares Error (LSE). The LSE adopted iteratively is able to track the frequency and amplitude of power system in steady states and in kinds of non-steady ones. Iterative LSE with resampling is analyzed in conditions of off-nominal input, nominal input with harmonics and decaying Direct Current (DC) offset and additive Gaussian white noise. In the circumstances of frequency and phase step changes, performance of resampling LSE is compared with traditional LSE. Resampling LSE has better performance than the traditional one in frequency tracking ability and can provide less mean square error.

Key words: Power system, frequency tracking, least squares error, resampling

INTRODUCTION

Frequency is one of the most important situation variables and operation parameter in power system. It is a good indicator of integrity of a power system facing separation and islanding. Phasor Measurement Unit (PMU) provides an indication of lost generation, as an example, a frequency drop of 0.1 Hz is typical in the Western Electricity Coordinating Council for 800 MW generation loss (NAERC., 2010). In the past several decades researchers have paid much attention to frequency measurement and analysis in power engineering. Types of frequency estimation methods have been reported, such as zero crossing (Begovic *et al.*, 1993; Nguyen and Srinivasan, 1984), demodulation technique (Begovic *et al.*, 1993), Newton algorithm (Terzija *et al.*, 1994), Kalman filter (Wood *et al.*, 1985; Routray *et al.*, 2002; Siavashi *et al.*, 2009), prony approach (Lobos and Rezmer, 1997), artificial neural network (Vianello *et al.*, 2010), etc. in time-domain. In the transform-domain, the Discrete Fourier Transform (DFT)/fast fourier transform (Yang and Liu, 2000; Zeng *et al.*, 2011) is

widely adopted as well as discrete wavelet transform (Chaari *et al.*, 1996; Huang *et al.*, 1999; Lin *et al.*, 2002; Ren and Kezunovic, 2011).

Frequency can be calculated by Least Squares Error (LSE) with a pseudo-inverse of a matrix that is determined by a frequency ω_0 and samples at time t_m ($m = 1, 2, \dots, M$, samples at $t_1 \sim t_M$ constituting a data window) of the input signal (Sachdev and Baribeau, 1979). Using LSE iteratively is able to track system frequency sample-by-sample (Sidhu and Sachdev, 1998). Its estimation accuracy is determined by the number of iterations per sample interval and by computation ability of hardware in real-time application.

In section II, we firstly presented basic idea of method of LSE, then we put forward a guide line on designing coefficients matrix for signal containing harmonics. In section III, we narrated the process of frequency tracking by resampling LSE. Simulation results of above algorithms in different scenarios were compared and some specialties of iterative LSE were given in section IV. Finally we concluded this study.

MATERIALS AND METHODS

Traditional LSE method: Suppose the nominal voltage or current signal of a power system is:

$$s(t) = \sum_{i=1}^H A_i \cos(\omega_i t + \phi_i) \quad (-\infty < t < \infty) \quad (1)$$

where, H is the order of highest harmonic and $A_i, \omega_i = i \times \omega_1, \phi_i$ are the amplitude, angular frequency and initial phase angle of each harmonic. According to rule of Nyquist, least sampling frequency $f_s \geq 2 \times H f_1$ should be chosen to eliminate aliasing and thanks to low pass filter, components with frequency higher than $H f_1$ are filtered, so we have:

$$V = A \times X \quad (2)$$

where, $V = [s(t_1), s(t_2), \dots, s(t_{2H})]^T$, in which $s(t_1), s(t_2)$ and $s(t_{2H})$ are the samples taken at time t_1, t_2 and t_{2H} (at least 2H samples are taken); $X = [X_1, X_2, \dots, X_{2H-1}, X_{2H}]^T$, in which $X_1 = A_1 \cos \phi_1, X_2 = A_1 \sin \phi_1, \dots$ and $X_{2H-1} = A_H \cos \phi_H, X_{2H} = A_H \sin \phi_H$ and the matrix:

$$A = \begin{bmatrix} a_{11} & a_{12} & \dots & a_{1_{2H-1}} & a_{1_{2H}} \\ \vdots & \vdots & \ddots & \vdots & \vdots \\ a_{2H_1} & a_{2H_2} & \dots & a_{2H_{2H-1}} & a_{2H_{2H}} \end{bmatrix}$$

in which $a_{11} = \sin(\omega_1 t_1), a_{12} = \cos(\omega_1 t_1), a_{13} = \sin(\omega_2 t_1), a_{14} = \cos(\omega_2 t_1), \dots, a_{1_{2H-1}} = \sin(\omega_H t_1), a_{1_{2H}} = \cos(\omega_H t_1), \dots$ and $a_{2H_1} = \sin(\omega_1 t_{2H}), a_{2H_2} = \cos(\omega_1 t_{2H}), a_{2H_3} = \sin(\omega_2 t_{2H}), a_{2H_4} = \cos(\omega_2 t_{2H}), \dots, a_{2H_{2H-1}} = \sin(\omega_H t_{2H}), a_{2H_{2H}} = \cos(\omega_H t_{2H})$. When, a fault or a disturbance occurs, the current signal consists of exponentially decaying DC offsets in electrical power system (ElRefaie and Megahed, 2010; Benmouyal, 1995; Kezunovic *et al.*, 1992; Sidhu *et al.*, 2003). The decaying rates depend on the time constants determined by the inductive reactance to resistance ratio (X/R ratio) of the system (Diniz de Oliveira *et al.*, 2012; Balamourougan and Sidhu, 2006; Lin and Liu, 2002; Belega and Petri, 2011). The larger the X/R ratio, the slower the DC component decays (Kang *et al.*, 2009; Gu and Yu, 2000; Lee *et al.*, 2009). Signal with the nominal component and a decaying DC offset is represented as:

$$s(t) = A_1 \cos(\omega_1 t + \phi_1) + A_{dc} e^{-t/\tau} \quad (0 \leq t \leq T_{DC}) \quad (3)$$

where A_{dc} and τ are the amplitude and time constant of the decaying DC offset component. The T_{DC} is the effecting period of decaying DC offset, since the decaying DC offset exists

only in several cycles and then disappears. It is a non-periodic signal and its frequency spectrum encompasses all the frequencies which cannot be removed by anti-aliasing low pass filter. Matrix A has to be rewritten according to input signal as:

$$\begin{bmatrix} a_{11} & a_{12} & \dots & a_{15} \\ a_{21} & a_{22} & \dots & a_{25} \\ \vdots & \vdots & \ddots & \vdots \\ a_{m1} & a_{m2} & \dots & a_{m5} \end{bmatrix}$$

where, $a_{m1} = 1, a_{m2} = t_m, a_{m3} = t_m^2, a_{m4} = \sin(\omega_1 t_m)$, at time t_m ($m \geq 5$) for iterative LSE. In order to obtain X, we have:

$$X = [A^T A]^{-1} A^T V = A^+ V \quad (4)$$

We get phasor with amplitude Y^1 and phase angle θ^1 :

$$Y(t_1) = Y^1 e^{j\theta^1} \quad (5)$$

Where:

$$\theta^1 = \text{atan}[Y(t_1)_{\text{image}}/Y(t_1)_{\text{real}}] \quad (6)$$

where, atan is the inverse tangent function. The $Y(t_1)_{\text{image}}$ and $Y(t_1)_{\text{real}}$ are the imaginary part and real part of phasor $Y(t_1)$ and:

$$Y^1 = \sqrt{X_1^2 + X_2^2}$$

In order to get $Y(t_2)$, we have to sample the input signal at time t_2 and t_3 , whose time interval is T_s . The data window to get $Y(t_2)$ is shifted by one T_s from that of $Y(t_1)$. So the angular frequency is calculated by $f_1 = (\theta^2 - \theta^1)/(2\pi T_s)$. Generally we have frequency at t_m :

$$f_1^m = (\theta^{m+1} - \theta^m)/(2\pi T_s) \quad (7)$$

Traditional LSE algorithm uses fixed sampling frequency of 720 Hz as specified in Sachdev and Baribeau (1979) and Sidhu and Sachdev (1998). However, frequency of the input may change under different conditions, such as off-nominal signal in kinds of transient states or under sudden step changes. In these circumstances one data window may contain non-integer number of samples that leads to spectrum leakage and error of estimation. If we change the rate of sampling according to previous estimated frequencies, there may be integer number of sample in one

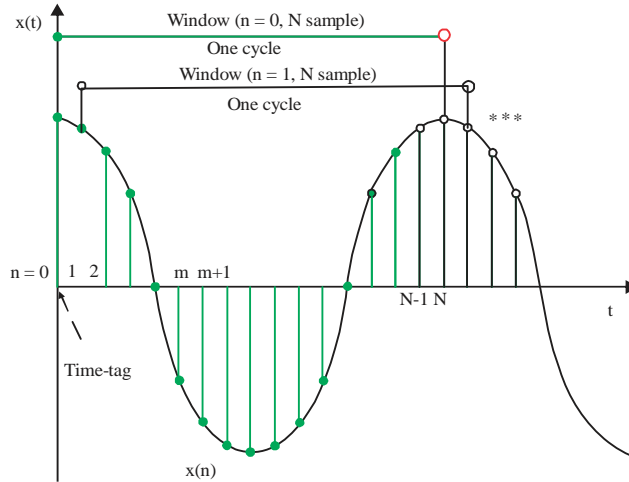


Fig. 1: Illustration of shifting windows of a cosine waveform with time-tag at starting time of t = 0 sec

data window and then the error of frequency would be minimized. Also the nominal signal may be polluted by harmonics, inter-harmonics, decaying DC offsets, additive Gaussian with noise, etc. Window-based data acquiring process is shown in Fig. 1.

Suppose frequency of input signal is an off-nominal one. By Eq. 7, we can estimate the frequency iteratively by resetting the matrix A with different estimated frequency f'. In reality, sampling rate is set properly to fulfill the requirement of Nyquist's rule and the criterion of sampling in Terzija *et al.* (1994). Secondly, an over-determined linear set is built to give a more precise estimation with more samples. Steps of iterative LSE method are:

- Step 1:** Sample the signal with frequency f_s and form a data window
- Step 2:** Obtain matrix A according to sampling frequency at time $t_m, m = 1, 2, \dots, M$, where M is the length of data window. M is bigger than length of vector X to construct an over-determined linear set
- Step 3:** Obtain X according to Eq. 4
- Step 4:** Obtain phase angle of phasor $Y(t_1)$ and $Y(t_2)$ by Eq. 6 and then estimated frequency f' by Eq. 7. Number of iteration is 1
- Step 5:** Resample the input signal and reset the matrix A according to estimated frequency f' at time $t_m, m = 1, 2, \dots, M$. Repeat steps 1~4 and get f''. Number of iteration is 2
- Step 6:** If $|f'' - f'| \leq \delta$, iteration process converge, where δ is a preset minimal number; and also if number iteration approaches the maximal number N_{ite_max} , iteration process stops

Frequency estimation by iterative resampling-LSE method:

In order to track the frequency sample-by-sample, we have to use the process of the iterative LSE per sampling time interval. The initial frequency to get matrix A in the following interval is also the estimated one of previous interval. Number of iterations per sampling interval, limited by computation ability of a digital processor, influences frequency tracking ability and convergence speed of the estimations process. Method of LSE adopts iteration process to get the off-nominal frequency more and more precisely. More iteration is accomplished during one time interval, more accurate we get but more time is consumed. Flow chart of resampling LSE is in Fig. 2.

RESULTS

Suppose the input signal contains a 3rd order harmonic and a decaying DC offset lasting in several cycles at the beginning of simulation as shown in Eq. 8. Parameters used in simulation are listed in Table 1:

$$X(t) = A_1 \cos(\omega_1 t + \phi_1) + A_3 \cos(\omega_3 t + \phi_3) + A_{dc} e^{-t/\tau} (0 \leq t) \quad (8)$$

We can design the matrix A with the help of Taylor expansion as:

$$A = \begin{bmatrix} a_{11} & a_{12} & \dots & a_{17} \\ a_{21} & a_{22} & \dots & a_{27} \\ \vdots & \vdots & \ddots & \vdots \\ a_{m1} & a_{m2} & \dots & a_{m7} \end{bmatrix}$$

where, $a_{m1} = 1, a_{m2} = t_m, a_{m3} = t_m^2, a_{m4} = \sin(\omega_1 t_m), a_{m5} = \cos(\omega_1 t_m), a_{m6} = \sin(\omega_3 t_m), a_{m7} = \cos(\omega_3 t_m)$ at time

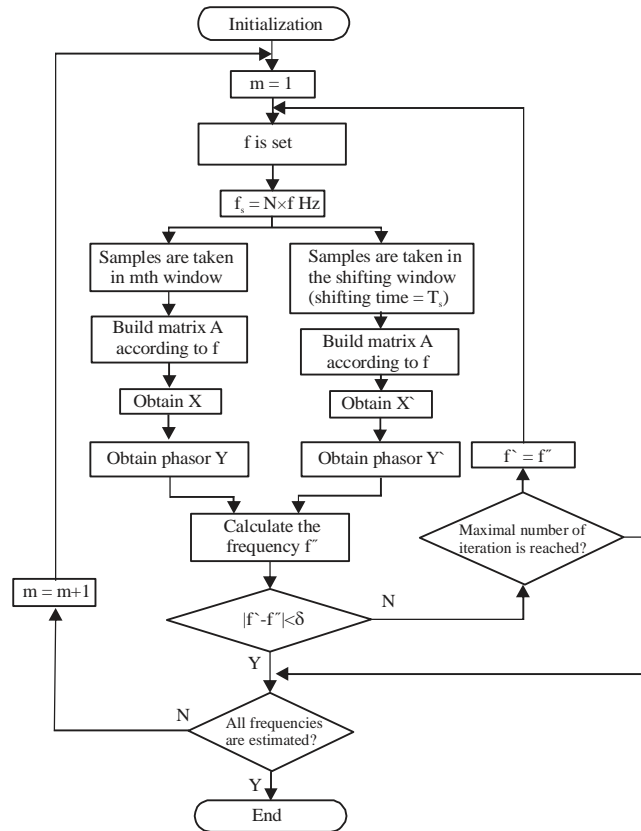


Fig. 2: Flow chart of resampling LSE

Table 1: Basic parameters utilized in simulation

Name	Notation	Value	Unit
Nominal frequency	f_1	60	Hz
Nominal angular frequency	ω_1	$2\pi f_1$	radian
Initial sampling frequency	f_s	720	Hz
Initial sampling interval	T_s	1/720	Second
Length of one cycle*	T	1/60	Second
Shifting frequency off the nominal one	Δf	[-5,5]	Hz
Signal to noise ratio	SNR	[20,60]	dB
Highest order of harmonics	N	3	-
Length of data window	L	14,19	-
Amplitude of nominal component	A_1	1	p.u.
Initial phase angle of nominal component	ϕ_1	0	deg
Amplitude of 3rd order harmonic	A_3	0.1	p.u.
Initial phase angle of 3rd order harmonic	ϕ_3	45	deg
Frequency of 3rd order harmonic	f_3	180	Hz
Amplitude of decaying DC offset	A_{dc}	[0.1,0.5]	p.u.
Time constant of decaying DC offset	τ	[0.1T,2T]	Second

*The cycle is calculated according to the nominal frequency

$t_m (m \geq 7); \omega_3 = 2\pi f_3$ and $X = [X_1, X_2, \dots, X_7]^T$, where $X_1 = A_{dc}$, $X_2 = -A_{dc}/\tau$, $X_3 = A_{dc}/(2\tau^2)$, $X_4 = A_1 \cos\phi_1$, $X_5 = A_1 \sin\phi_1$, $X_6 = A_3 \cos\phi_3$, $X_7 = A_3 \sin\phi_3$.

At first, we choose the best sampling rate and data window for LSE algorithms, since the performance of these algorithms are highly connected with them. Former studies show that lower sampling rate and longer data window help

to obtain a better result (Sachdev and Baribeau, 1979; Sidhu and Sachdev, 1998). The f_{input} denotes the frequency of input signal.

In Fig. 3, we find that the number of iterations during each sampling interval is set to be 2 or 3, which is good enough to acquire a high accuracy for resampling LSE and comparing with the traditional one, the resampling LSE method tends to converge more robustly. In Fig. 4, we know that tracking accuracy of resampling LSE is higher than that of the traditional one. The dynamic range of resampling LSE is smaller than that of the traditional one. Figure 5 shows the Mean Square Error (MSE) of two LSE methods with different variables such as time constant and amplitude of decaying DC offset component. When, the amplitude of decaying DC offset is 0.2 p.u. and the time constant is 0.1 T, the MSE are 1.2106 and 0.5222 for traditional LSE and resampling LSE, respectively.

Figure 6 tell us that resampling LSE does not work well when signal to noise ratio (SNR) is less than 25~30 dB. In the practice of power system, SNR is always higher than 30 dB. So Gaussian white noise could not do too much harm to resampling LSE in practice.

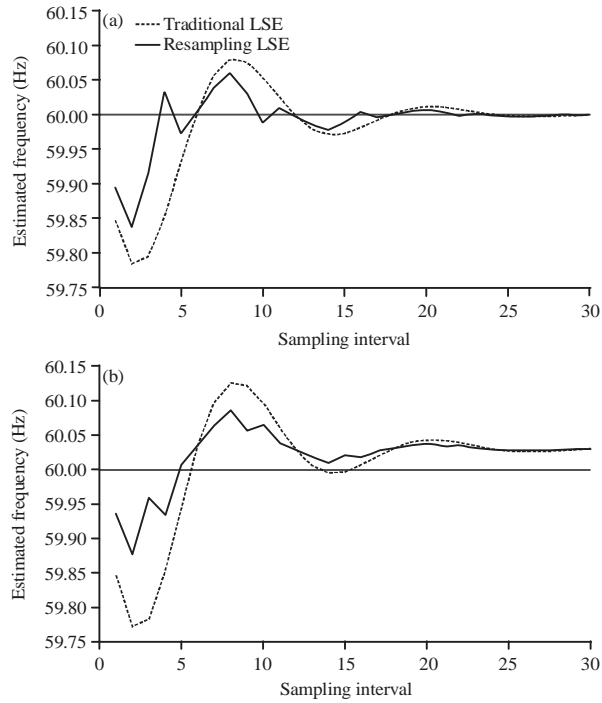


Fig. 3(a-b): Performance of traditional and resampling LSE methods with nominal input frequency (Initial sampling rate $f_s = 720$ Hz, data window $L = 14$, $f_{input} = 60$ Hz, first 30 estimated frequencies are plotted), (a) Number of iteration $N_{ite} = 2$ in each sampling interval and (b) Number of iteration $N_{ite} = 3$ in each sampling interval

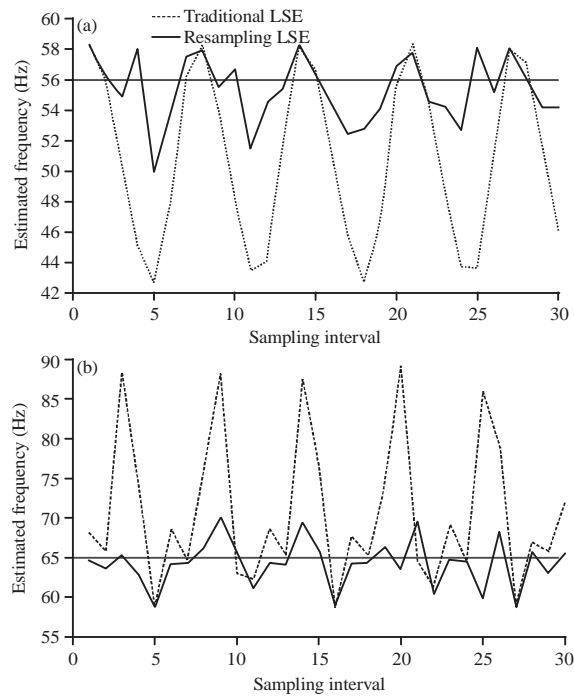


Fig. 4(a-b): Performance of traditional and resampling LSE methods with off-nominal input frequency (Initial sampling rate $f_s = 720$ Hz, data window $L = 14$, Number of iteration $N_{ite} = 2$, first 30 estimated frequencies are plotted), (a) $f_{input} = 55$ Hz and (b) $f_{input} = 65$ Hz

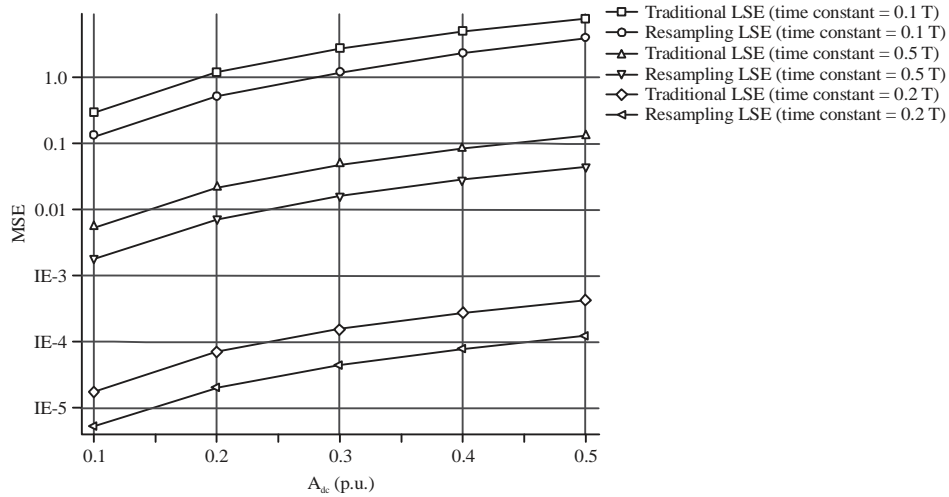


Fig. 5: Performance of traditional and resampling LSE methods with different decaying DC offset (Initial sampling rate $f_s = 720$ Hz, data window $L = 14$, Number of iteration $N_{ite} = 2$, $f_{input} = 60$ Hz, first 30 estimated frequencies are used for MSE calculation)

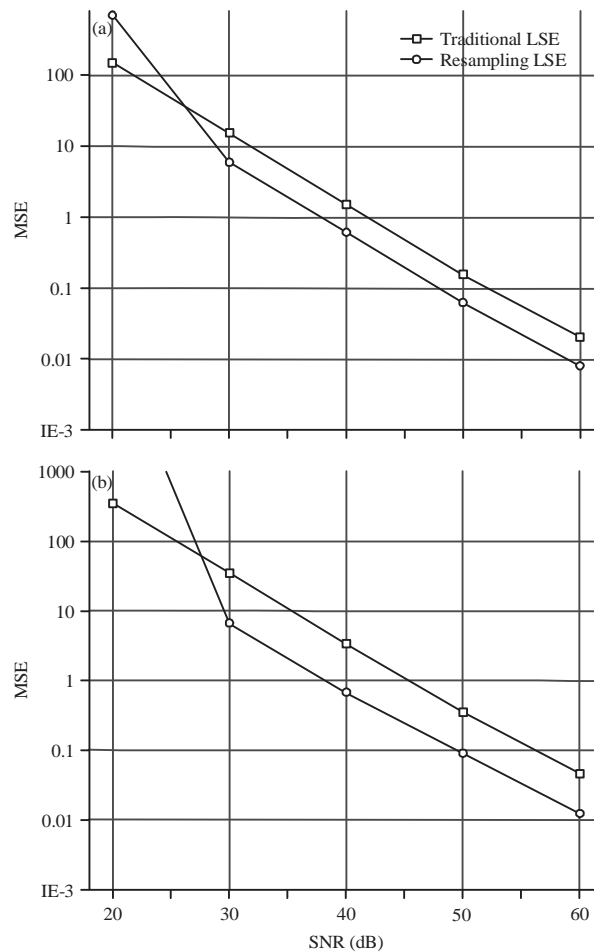


Fig. 6(a-b): Performance of traditional and resampling LSE methods with noise (Initial sampling rate $f_s = 720$ Hz, data window $L = 14$, $f_{input} = 60$ Hz, first 30 estimated frequencies are used for MSE calculation and averaged in 1000 times), (a) Number of iteration $N_{ite} = 2$ and (b) Number of iteration $N_{ite} = 3$

DISCUSSION

To study the dynamic tracking ability of LSE, step changes of input signals under various conditions are proposed by IEEE Std. The C37.118 (IEEE Standard, 2006, 2011). We compare the result of resampling LSE with the traditional one in the following parts, numerical comparison are listed in the following tables and figures.

Frequency step change:

- **Scenario 1:** Nominal input signal. The input signal contains only the nominal sine signal. The first 1 Hz frequency change occurs at $2.6 \times T$, the second -1 Hz frequency change occurs at $6.9 \times T$

Table 2: Performance of traditional and resampling LSE methods with number of iterations (window length $L = 19$)

MSE	Number of iterations			
	1	2	3	4
Traditional LSE	10.1792	44.9310	113.1579	228.2406
Resampling LSE	10.1792	8.6464	18.4575	9.8293
Gain	-	7.1571	7.8751	13.6587
Traditional LSE	11.7281	51.5753	130.2425	265.0047
Resampling LSE	11.7281	11.3092	15.3735	20.3517
Gain	-	6.5901	9.2798	11.1465

MSE: Mean square error

- **Scenario 2:** Nominal input signal with DC offset and 3rd order harmonics, without noise
- **Scenario 3:** Nominal input signal with DC offset, 3rd order harmonics and additive Gaussian white noise, SNR = 30 dB

For Table 2 and 3 and Fig. 7a-c, we find resampling LSE algorithms has 6~11 dB gains comparing with traditional LSE. Because of iterations in sampling interval, there are still jitters in frequency tracking as shown in above figures. The overshoots appear around the time that step changes happen, however the peak value and MSE of resampling LSE is less than that of traditional LSE. We choose $L = 19$ as the length of data window which is longer than that of (Sidhu and Sachdev, 1998), since longer data window can minimize the influence of Gaussian white noise and also can improve the accuracy of frequency tracking ability but more computation burden and storage requirement.

Table 3: Performance of traditional and resampling LSE methods with number of iterations (window length $L = 19$; 1000 time average)

MSE	Number of iterations			
	1	2	3	4
Traditional LSE	11.9846	52.5273	132.4945	268.9973
Resampling LSE	11.9820	12.2859	18.6639	19.0251
Gain	-	6.3098	8.5120	11.5042

MSE: Mean square error

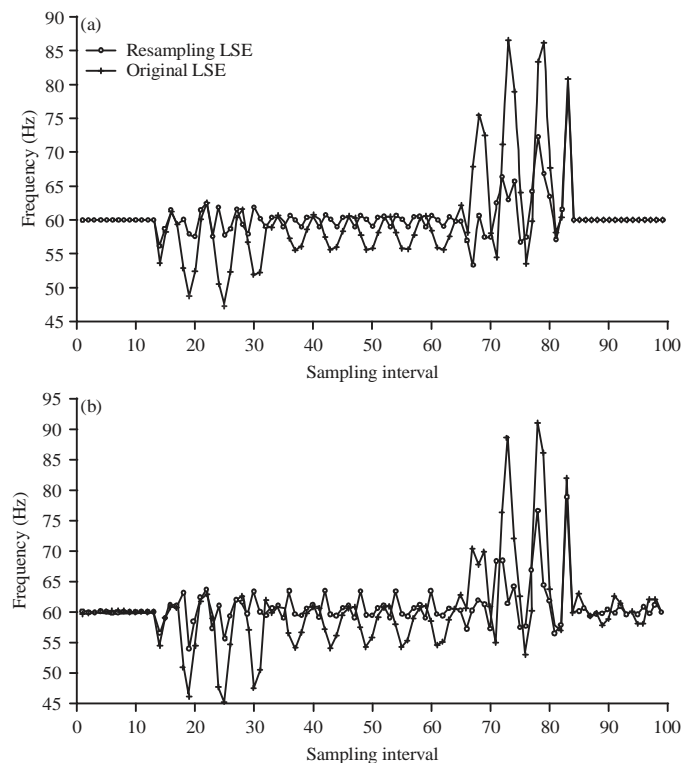


Fig. 7(a-f): Continue

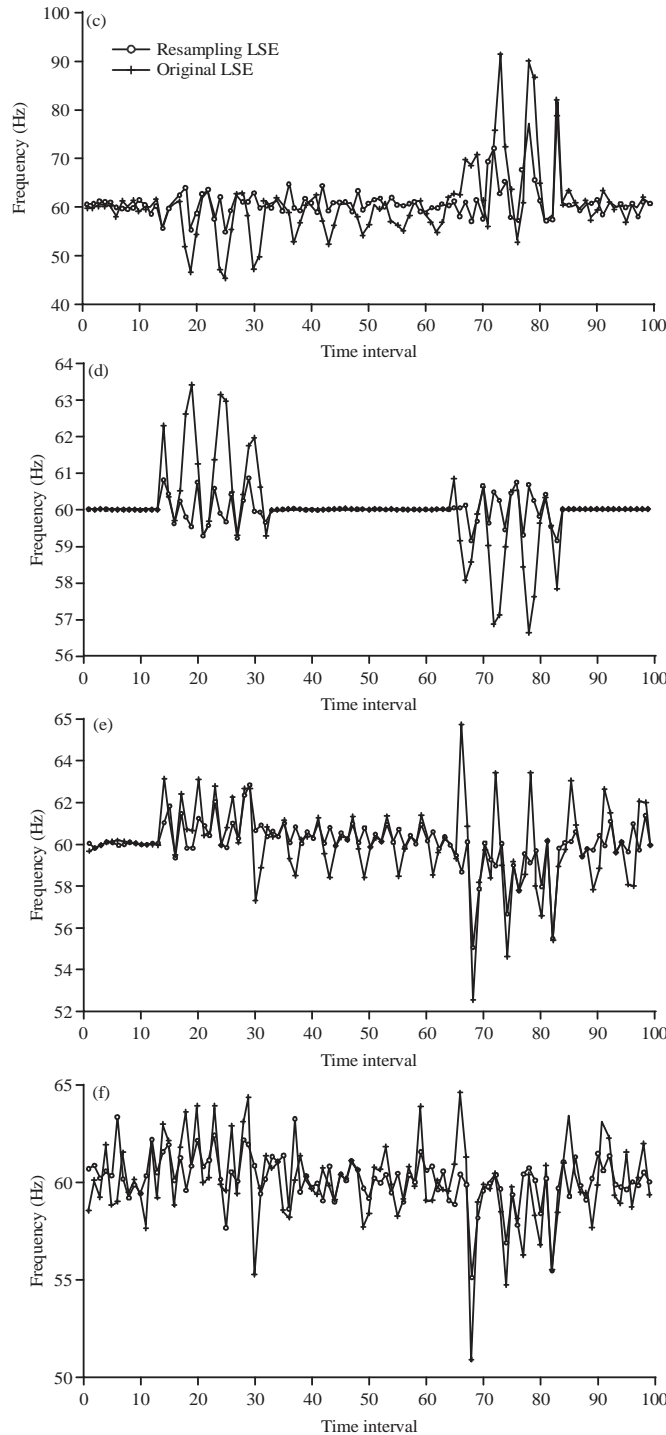


Fig. 7(a-f): Comparison of resampling LSE and traditional LSE (Number of iteration $N_{ite} = 2$), (a) Scenario 1, (b) Scenario 2, (c) Scenario 3, (d) Scenario 4, (e) Scenario 5 and (f) Scenario 6

Phase step change:

- **Scenario 4:** Nominal input signal. The first 5° phase change happens at 2.6 T, the second -5° phase change happens at 6.9 T

- **Scenario 5:** Nominal input signal with DC offset and 3rd order harmonics, without noise
- **Scenario 6:** Nominal input signal with DC offset, 3rd order harmonics and Gaussian white noise. SNR = 30 dB

Table 4: Performance of traditional and resampling LSE methods with the number of iterations, frequency is off nominal one (window length L = 19)

MSE	Number of iterations			
	1	2	3	4
Traditional LSE	0.2737	1.0948	2.4650	4.3880
Resampling LSE	0.2737	0.0967	0.4896	0.3566
Gain	-	10.5391	7.0198	10.9009
Traditional LSE	0.8277	3.2766	7.3082	12.9005
Resampling LSE	0.8277	1.2127	2.5305	4.0641
Gain	-	4.3176	4.6060	5.0164

MSE: Mean square error

Table 5: Performance of traditional and resampling LSE methods with the number of iterations, frequency is off nominal one (window length L = 19; 1000 time average)

MSE	Number of iterations			
	1	2	3	4
Traditional LSE	1.0869	4.3166	9.5839	17.1295
Resampling LSE	1.0897	1.8489	3.6819	5.5834
Gain	-	3.6823	4.1547	4.8685

MSE: Mean square error

Form Table 4 and 5 and Fig. 7d-f, we can draw the following conclusions. There are huge overshoot near the point of frequency step change and jitter phenomenon when the input frequency is the off-nominal one. There are certain specialties of iterative LSE: Design of matrix A should conform to input signal, in another word, designer has to estimate how many harmonics are contained in the signal after low pass filter. Otherwise, the result of LSE would be error, which is difficult to be revised by iteration. Data window of LSE is defined by the number of unknown parameters in every row of matrix A. A longer data window contains more data which are more easily distorted by dynamic change of power system and it also causes longer delay for reporting PMU to phasor data concentrator. Performance of LSE and convergence speed are affected by designing of matrix A. If several decaying DC offset components are contained simultaneously in the signal, it is a huge problem for designing matrix A. Method of LSE is sensitive to signal distortions and signal step change caused by faults of power system, since it is a time-domain algorithm without abilities of de-noising and anti-aliasing.

CONCLUSIONS

We proposed a method of resampling LSE, whose performance was analyzed and compared with the traditional one, when the incoming signal containing harmonics and decaying DC offset. The key point of resampling LSE is that it resamples the input signal according to frequency calculated in previous sampling interval and the iterations processes are accomplished in each interval, which is limited by the computation ability of digital signal processor chips. Iterative LSE can track the nominal frequency and the off-nominal one in presence of additive Gaussian white noise with harmonics.

The resampling one is less sensitive to DC offset than traditional one. Resampling LSE shows better performance of frequency tracking comparison with the old one with less MSE and better performance in dynamic states of a power system.

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