



A Fast Fault Classification Technique for Power Systems

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ABSTRACT: This paper proposes a fast fault classification technique using three phase current signals for power systems. Digital Fourier Transform, the ‘Least Square’ method or the Kalman Filtering technique are used to extract fundamental frequency components of three phase fault currents. Fast fault classification can be achieved using the fault probability of three phases. Results from simulation work on EMTP have validated the proposed fault classification technique. The response time of the fault classification technique using the ‘Least Squares’ method is 1.875ms (3 samples) for single-phase-to-earth fault, two-phase-to-earth fault, two-phase-fault and three-phase fault.

KEYWORDS: Fault classification, Fourier transforms, least squares method, Kalman filter

I. INTRODUCTION

Fault classification is an important function for power systems protection and fault locators. A modular methodology [1] is used for fast fault classification. Girgis [2-3] uses an adaptive Kalman filters technique and a probabilistic technique for fault classification. One Kalman filter assumes the features of the faulted phase while the other has features of an un-faulted phase. The condition of the phase, faulted or non-faulted, is then decided from the computed probabilities. Barros and Drake [4] implement 8-bit microprocessors for fault classification based on Kalman filters. The response time is less than 20 ms for all possible types of ground fault. Artificial intelligent techniques are also used for fault classification, such as the neuro-fuzzy approach [5], probabilistic neural networks [6], neural networks [7] and combined unsupervised/supervised neural networks [8]. Zhao [9] applies a wavelet transform based scheme for fault classification.

This paper proposes a fast fault classification technique for fault locators. Digital Fourier Transform (DFT), or the Least Squares Method (LSM), or Kalman Filtering Technique (KFT) is used to extract fundamental frequency components of three phase currents. Fault classification of three phases can be made by using of the fault probability method. Results from EMTP/ATP simulations are presented. The fault classification technique using the ‘Least Squares’ method can operate correctly within 3 samples (1.875ms) for single line-to-ground (SLG) faults, double line-to-ground (2LG) faults, line-to-line (LL) faults and three-phase (3 ϕ) faults.

II. FAULT CLASSIFICATION

Fault classification is determined by fault probabilities of three phases. Fault probability of phases depends upon the amplitude of the fundamental components of three phase currents, $I_a(t)$, $I_b(t)$ and $I_c(t)$, which are calculated using fundamental frequency component extracting techniques, such as DFT, LSM or KFT.

(a) Fault probability function

There is a relation between fault probability and amplitude of fundamental frequency current I_s ($S = A, B, C$). When I_s is a load current, its fault probability is low (close to 0) and if I_s is a fault current, the fault probability will be high (close to 1.0 i.e.100%). So, it is reasonable to assume that the fault probabilities are as follows (there are also other similar functions):

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$$\begin{aligned}
 fp(S) &= f(I_s) = 0 && (I_s < 1.0 \text{ p.u.}) \quad (S = A, B, C) \\
 &= f(I_s) = (I_s - 1.0) && (1.0 < I_s < 2.0 \text{ p.u.}) \\
 &= f(I_s) = 1.0 && (I_s > 2.0 \text{ p.u.})
 \end{aligned}$$

Per unit based on maximum load current is used. The probability function is shown in fig 1.

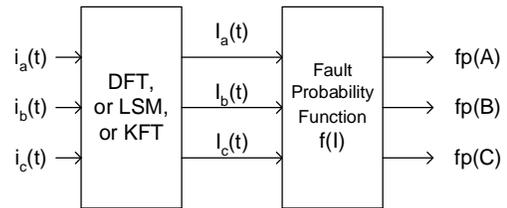
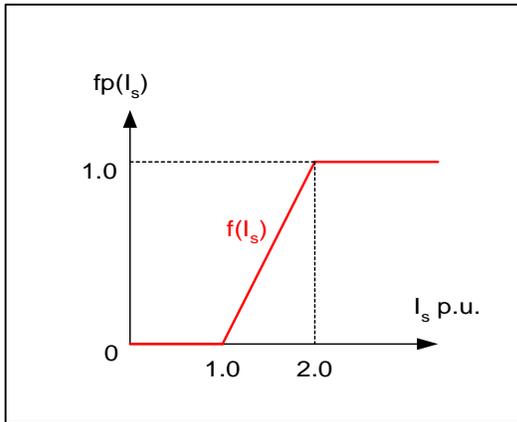


Fig. 1 Fault probability function Fig. 2 Block diagram of fault classification

(b) Functional block diagram

A block diagram of fault classification is shown in fig 2. The input signals are three phase current $i_a(t)$, $i_b(t)$ and $i_c(t)$. Amplitudes of fundamental frequency current $I_a(t)$, $I_b(t)$ and $I_c(t)$ are calculated using mathematical algorithm DFT, or LSM, or KFT. Fault probability $fp(A)$, $fp(B)$ and $fp(C)$ can be obtained from fault probability function $f(I)$.

III. SIMULATION RESULTS

To validate the proposed fault classification technique, a two-source system [4] is used. Parameters of the two-source system are $Z_{SA} = 0 + j21.5$ ohms; $Z_{OA} = 0 + j24.4$ ohms; $Z_{SB} = 0 + j25.6$ ohms; $Z_{OB} = 0 + j27.8$ ohms. Line impedance $Z_L = 3.4 + j53.4$ ohms and $Z_{OL} = 57.0 + j241$ ohms. The source voltage is $E_A = 225.17 + j0$ kV and $E_B = 169.86 - j169.86$ kV. A sample rate of 32 samples per cycle ($\Delta t = 0.625$ ms) is used.

All simple fault types, such as single-phase-to-earth faults (LG), two-phase-to-earth faults (2LG), two-phase-short circuit faults (LL) and three-phase-short-circuit faults (3ϕ) are taken into account. Fault resistance with zero and 10 ohms are considered at faultpoint. The simulation results are listed in table I and II. In these tables, the minimum sample is counted from fault occurrence (sample=0) to the sample at which fault probability of the faulty phase has reached 100%, i.e. values with 1.0.

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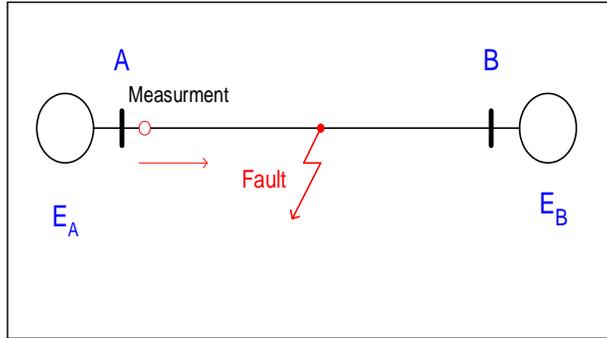


Fig. 3 Two-source simple system for validation

TABLE I- MINIMUM SAMPLES FOR FAULT CLASSIFICATION WITH $R_f=0$ OHM

	DFT	LSM	KFT
SLG	10	2	5
2LG	15	2	10
LL	9	3	5
3 ϕ	12	2	8

TABLE II- MINIMUM SAMPLES FOR FAULT CLASSIFICATION WITH $R_f=10$ OHMS

	DFT	LSM	KFT
SLG	17	2	5
2LG	15	2	9
LL	9	3	5
3 ϕ	12	2	8

Figs 4 to 15 present fault classification simulation results with zero fault resistance for Table I.

(a) Digital Fourier Transform (DFT)

For A-phase-to-earth fault, the fault probability of A phase reaches 1.0 after 10 samples when the fault occurs. The result is shown in fig 4. It is the same for B-phase-to-earth fault (10 samples) and C-phase-to-earth fault (9 samples). Therefore there are 10 samples in table I. In a similar way, fig 5 shows fault classification results for A B-phase-to earth fault, fig 6 for A-B-phase short circuit fault and fig 7 for three-phase fault.

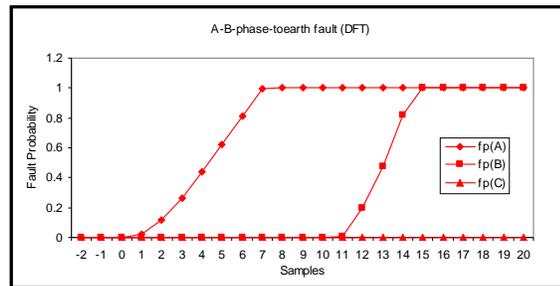
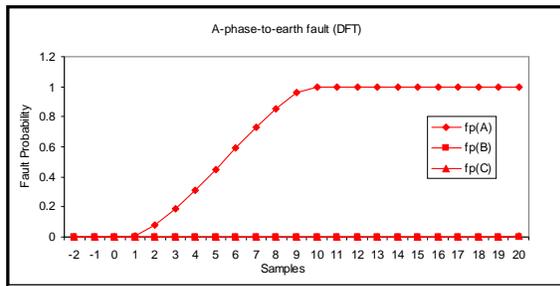


Fig. 4 A-phase-to-earth fault Fig. 5 A-B-phase-to-earth fault

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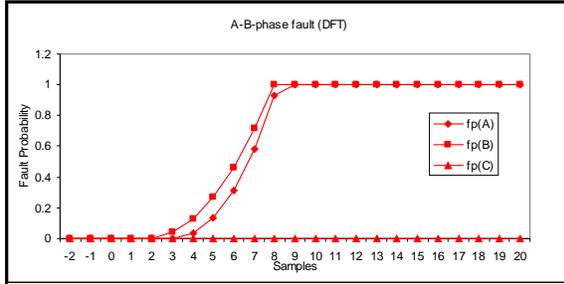


Fig.6 A-B-phase fault

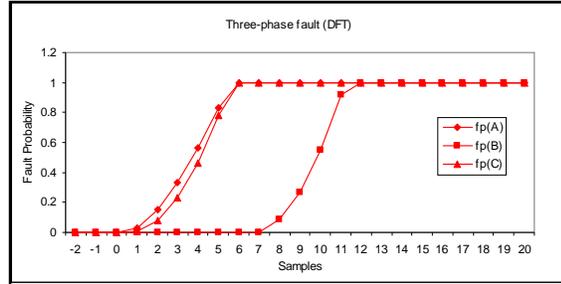


Fig. 7 Three-phase fault

(b) Least Squares Method (LSM)

Figs. 8 to 11 show fault classification results using LSM. It is easy to find that 2 samples are needed for correctly classifying A-phase-to-earth fault, 2 samples for A-B-phase-to-earth fault, 3 samples for A-B-phase short circuit and 2 samples for three-phase-short circuit fault.

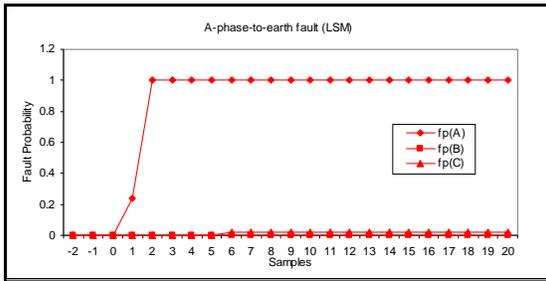


Fig. 8 A-phase-to-earth fault

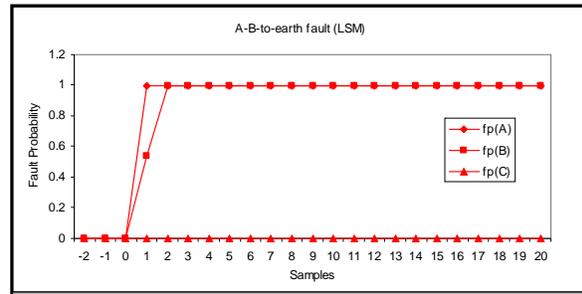


Fig. 9 A-B-phase-to-earth fault

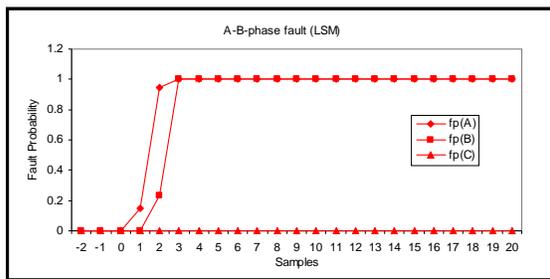


Fig. 10 A-B-phase fault

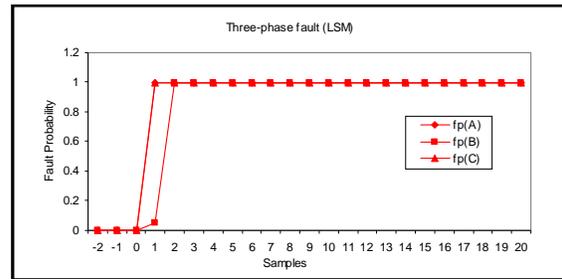


Fig. 11 Three-phase fault

(c) Kalman Filtering Technique (KFT)

Figs 12 to 15 show fault classification results using KFT. It can be seen that 5 samples are needed for correctly classifying A-phase-to-earth fault, 10 samples for A-B-phase-to-earth fault, 5 samples for A-B-phase short circuit and 8 samples for three-phase-short circuit fault.

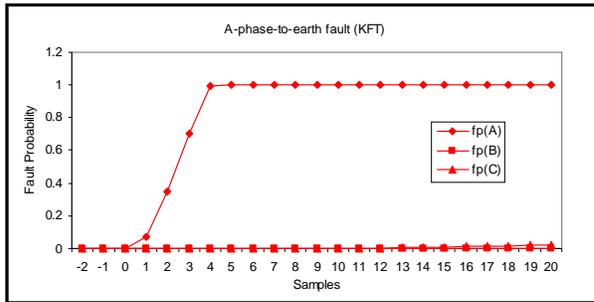


Fig. 12 A-phase-to-earth fault

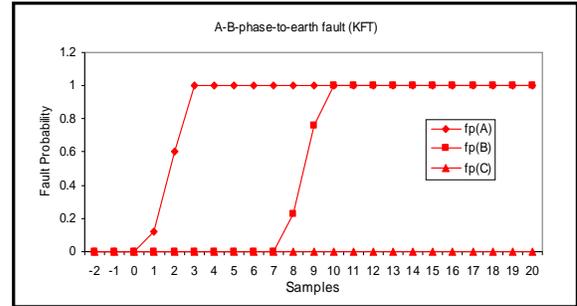


Fig. 13 A-B-phase-to-earth fault

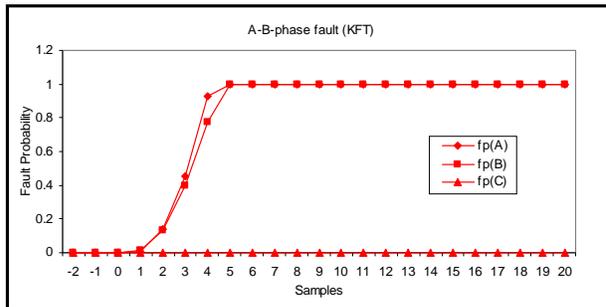


Fig. 14 A-B-phase fault

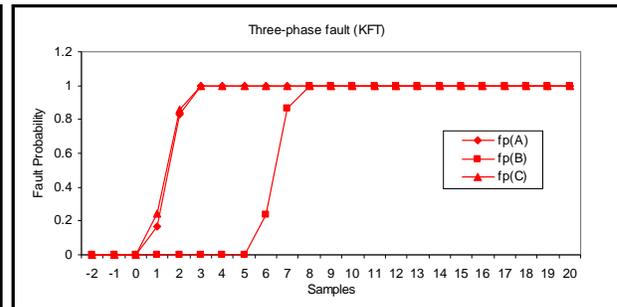


Fig. 15 Three-phase fault

IV.COMPARISON

(1) Among the three methods, performance of simulation results using LSM with 6 samples is the most effective. Fault classification using LSM can give correct fault phase detection within 3 samples (1.875ms) for all four types of simple fault. On the other hand, KFT needs 10 samples (6.25 ms) and DFT needs 15 samples (9.375 ms) to represent correct fault types.

(2) There are no significant changes for 0-10 ohms fault resistance at fault point. The reason may be that the fault resistance affects the amplitude of fundamental frequency fault current, but the fault current is much more than the threshold value (2.0 p.u.) for a fault probability of 1.0.

V.CONCLUSION

This paper proposes a fast fault classification technique using current signals for fault locators in power systems. Digital Fourier Transform, the Least Square Method or the Kalman Filtering Technique are used to extract fundamental frequency components of three phase fault currents. Fault classification can be made using fault probability of the three phases. Results from simulation work on an EMTP are presented. The fault classification technique using the 'least square' method can operate correctly within 3 samples (1.875ms) for single line-to-ground faults, double line-to-ground faults, line-to-line faults and three-phase faults.

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Appendix: DFT, LSM and KFT

A-1 Discrete Fourier transform

The Discrete Fourier transform (DFT) is defined as

$$X[k] = \sum_{n=0}^{N-1} x[n] e^{-j \frac{2\pi kn}{N}} \quad (A1-1)$$

where $x[n]$ is a sequence obtained by sampling the continuous time signal $x(t)$ every time step Δt for N samples. $X(k)$ is the frequency-domain representation of $x(n)$. It is reasonable to assume that current signal $i(t)$ has the following form:

$$i(t) = \frac{a_0}{2} + \sum_{n=1}^{\infty} a_n \cos n\omega_0 t + \sum_{n=1}^{\infty} b_n \sin n\omega_0 t \quad (A1-2)$$

Let I_x and I_y be the real and imaginary parts, respectively, of the phasor that represents the fundamental component of the faulted current waveform $i(t)$.

$$I_x = \frac{2}{N} \sum_{j=0}^{N-1} i(t_j) \cos\left(\frac{2\pi j}{N}\right) = \frac{2}{N} \sum_{j=0}^{N-1} i(t_j) \cos\omega_0 t_j = \frac{2}{N} \sum_{j=0}^{N-1} W_{x,j} i(t_j) \quad (A1-3)$$

and

$$I_y = \frac{2}{N} \sum_{j=0}^{N-1} i(t_j) \sin\left(\frac{2\pi j}{N}\right) = \frac{2}{N} \sum_{j=0}^{N-1} i(t_j) \sin\omega_0 t_j = \frac{2}{N} \sum_{j=0}^{N-1} W_{y,j} i(t_j) \quad (A1-4)$$

The amplitude I_m and phase angle θ_I of the fundamental frequency current are:

$$I_m = \sqrt{I_x^2 + I_y^2} \quad (A1-5)$$

$$\theta_I = \tan^{-1}\left(\frac{I_y}{I_x}\right) \quad (A1-6)$$

A-2 Least squares method

LSM can minimise the fitting error, and can be used to extract the fundamental component of voltage and current waveforms. It is assumed that current (or voltage) has a following form:



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$$i(t) = \sum_{m=0}^{2M} I_m s_m(t) + \varepsilon(t) \quad (A2-1)$$

where the signals $s_m(t)$ are assumed known but the coefficients I_m are unknown. The obvious choices for the signals include:

$s_0(t) = e^{-\frac{t}{\tau}}$	the exponential offset
$s_1(t) = \cos \omega_0 t$ and $s_2(t) = \sin \omega_0 t$	fundamental frequency components
$s_3(t) = \cos 2\omega_0 t$ and $s_4(t) = \sin 2\omega_0 t$	the second harmonics
.....	other harmonics

where M is the highest harmonic considered, τ is the time constant of the decaying DC component and ω_0 is the angular frequency of the fundamental component.

The aim then is to estimate the unknown coefficients, $I_0, I_1, I_2, \dots, I_{2M}$ from the N measurements, that is: if current is sampled as $i(t_1), i(t_2), \dots, i(t_n), \dots, (t_n = n\Delta t)$. The N measurements are put into equation (A2-1), and written as matrix form:

$$\begin{bmatrix} i(t_1) \\ i(t_2) \\ \vdots \\ i(t_n) \end{bmatrix} = \begin{bmatrix} s_0(t_1) & s_1(t_1) & \cdots & s_{2M}(t_1) \\ s_0(t_2) & s_1(t_2) & \cdots & s_{2M}(t_2) \\ \vdots & \vdots & \ddots & \vdots \\ s_0(t_n) & s_1(t_n) & \cdots & s_{2M}(t_n) \end{bmatrix} \begin{bmatrix} I_0 \\ I_1 \\ \vdots \\ I_{2M} \end{bmatrix} + \begin{bmatrix} \varepsilon(t_1) \\ \varepsilon(t_2) \\ \vdots \\ \varepsilon(t_n) \end{bmatrix}$$

In short: $\mathbf{i} = \mathbf{S} \mathbf{I} + \mathbf{e}$ (A2-2)

The \mathbf{I} that minimize can be obtained by taking the partial derivatives of equation (A2-2) with respect to the component of \mathbf{I} and equating to zero,

$$\frac{\partial(\mathbf{e}^T \mathbf{e})}{\partial \mathbf{I}} = 0$$

The estimation result of unknown \mathbf{I} is

$$\hat{\mathbf{I}} = (\mathbf{S}^T \mathbf{S})^{-1} \mathbf{S}^T \mathbf{i} \quad (A2-3)$$

After $\hat{\mathbf{I}}$ is obtained, the amplitude of the fundamental frequency current can be calculated:

$$I_{amp} = \sqrt{I_1^2 + I_2^2}$$

A-3 Kalman Filtering Technique

The advantage of the Kalman-filtering-based algorithm is not only the expeditious rate of convergence of the estimated states to the exact values, but also the low computer burden of the computation of the states. A mathematical model of the states to be estimated is assumed to be in the form

$$x_{k+1} = \Phi_k x_k + w_k \quad (A3-1)$$

The observation (measurement) of the process is assumed to occur at discrete points of time in accordance with the relation

$$z_k = H_k x_k + v_k \quad (A3-2)$$

The covariance matrices for system and measurement noise vectors, w_k and v_k are given by

$$w_k w_k^T = Q_k, \quad i = k; \quad w_k w_k^T = 0, \quad i \neq k \quad (A3-3)$$

$$v_k v_k^T = R_k, \quad i = k; \quad v_k v_k^T = 0, \quad i \neq k \quad (A3-4)$$

$$w_k v_k^T = 0, \text{ for all } i \text{ and } k \quad (A3-5)$$



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Recursive Kalman Filter Equations

The Kalman filter gain and the covariance matrix may be computed off line and stored in the memory using equation (A3-6), (A3-7) and (A3-8).

(a) Compute the Kalman filter gain K_k

$$K_k = P_k^{-1} H_k (H_k P_k^{-1} H_k^T + R_k)^{-1} \quad (A3-6)$$

(b) Compute error covariance for updated estimate:

$$P_k = (I - K_k H_k) P_k^{-1} \quad (A3-7)$$

(c) Project error covariance

$$P_{k+1}^{-1} = \Phi_k P_k \Phi_k^T + Q_k \quad (A3-8)$$

Using the Kalman filter gain and an initial estimate, the states of the current can be recursively estimated using equation (A3-9) and (A3-10).

Update estimate with measurement z_k

$$\begin{bmatrix} \hat{x}_{1k} \\ \hat{x}_{2k} \\ \hat{x}_{3k} \end{bmatrix} = \begin{bmatrix} \hat{x}_{1k}^{-1} \\ \hat{x}_{2k}^{-1} \\ \hat{x}_{3k}^{-1} \end{bmatrix} + \begin{bmatrix} K_{1k} \\ K_{2k} \\ K_{3k} \end{bmatrix} [z_k - [\cos \omega_0 k \Delta t, -\sin \omega_0 k \Delta t, 1] \begin{bmatrix} \hat{x}_{1k}^{-1} \\ \hat{x}_{2k}^{-1} \\ \hat{x}_{3k}^{-1} \end{bmatrix}] \quad (A3-9)$$

$$\text{Project estimates } \begin{bmatrix} \hat{x}_{1k+1}^{-1} \\ \hat{x}_{2k+1}^{-1} \\ \hat{x}_{3k+1}^{-1} \end{bmatrix} = \begin{bmatrix} \hat{x}_{1k} \\ \hat{x}_{2k} \\ \hat{x}_{3k} \cdot e^{-\beta \Delta t} \end{bmatrix} \quad (A3-10)$$