# MICROCONTROLLER BASED FREQUENCY RELAY

**A DISSERTATION**

*Submitted in partial fulfilment of the requirements for the award of the degree*
*of*
**MASTER OF TECHNOLOGY**
*in*
**ELECTRICAL ENGINEERING**
**(With Specialization in Measurement & Instrumentation)**

*By*

**Lt. Col. JAGDISH PATI**



DEPARTMENT OF ELECTRICAL ENGINEERING
INDIAN INSTITUTE TECHNOLOGY ROORKEE
ROORKEE-247 667 (INDIA)

JUNE, 2005

## CANDIDATE'S DECLARATION

I, declare that the work presented in the dissertation titled **"DEVELOPMENT OF MICROCONTROLLER BASED FREQUENCY RELAY"** submitted in partial fulfillment of the requirement for the award of degree of Master of Technology with specialization in **Measurement and Instrumentation** in the department of Electrical Engineering, IIT Roorkee, Roorkee-247667, is an authentic record of my work under the guidance of **Dr R P Maheshwari**, Assistant Professor, Electrical Engineering Department, IIT Roorkee.

I have not submitted the matter embodied in this dissertation for the award of any other degree.

Date: 01 June 2005

Place: Roorkee

**Jagdish Pati**
**Lt Col**

---

This is to certify that, the above statement made by the student is correct to the best of my knowledge and belief.

1.06.05

**Dr. R .P. Maheshwari**
Assistant Professor
Deptt of Electrical Engineering
IIT Roorkee
Roorkee – 247667

## ACKNOWLEDGEMENT

I take this opportunity to express my sincere gratitude and thanks to all those who extended full cooperation and all possible help to me, throughout the duration of the dissertation.

First and foremost, I am indebted to my guide Dr R P Maheshwari, Assistant Professor, Electrical Engineering Department, for his continuous support, encouragement, assistance and able guidance. His knowledge and practical inputs have helped me to make a conclusive report on the subject.

The support of my family and friends has been valuable in timely completion of the work.

This acknowledgement would be incomplete if I don't thank the laboratory and other staff of the department who have rendered their support all the way.

# PREFACE

Frequency is an important operating parameter of power system. Frequency of a power system remains constant if sum of all loads plus losses equals total generation in the system. However, frequency starts to decrease if total generation is less than the sum of loads and losses. On the other hand, the system frequency increases if total generation exceeds the sum of loads and losses. The frequency changes that can be tolerated in a system are governed by the characteristics of the rotating equipment in the system.

Reliable frequency measurements are therefore a pre-requisite for effective power system protection and control. These measurements should produce frequency readings as accurately and as fast as possible.

Typically, both under and over frequency relays are used for Generator protection, Feeder protection Load shedding, EHV line schemes, Synchronism check and so on.

A micro controller (8051) based frequency relay has been developed utilizing Zero crossing technique and also simulation of one of the frequency measurement technique, namely 'Demodulation' method has been incorporated in this dissertation.

# CONTENTS

# CHAPTER 1

# INTRODUCTION

The capital investment involved in a power system for the generation, transmission and distribution of electric power is so great that the proper precautions must be taken to ensure that the equipment not only operates as nearly as possible to peak efficiency, but also that it is protected from accident.

Providing reliable electricity is an enormously complex technical challenge, even on the most routine days. It involves real-time assessment, control and coordination of electricity production at all the generators, moving electricity across an interconnected network of transmission lines, and ultimately delivering the electricity to millions of customers by means of a distribution network.

Electricity is produced at lower voltages at generators from various fuel sources, such as nuclear, coal, oil, natural gas, hydropower, and so on. The produced electricity is "Stepped up" to higher voltages for transportation in bulk over transmission lines. Operating the transmission lines at high voltages reduces the losses of electricity from conductor heating and allows power to be shipped economically over long distances. Transmission lines are interconnected at switching stations and substations to form a network of line and stations called the power "grid". Electricity flows through the interconnected network of transmission line from the generators to the loads in accordance with the laws of physics - along the "paths of least resistance" in much the same way that water flows through a network of canals. When the power arrives near a load center, it is "stepped down" to lower voltages for distribution to customers. The bulk of power system is predominantly an alternating current (ac) system because of the ease and low cost with which voltages in system can be converted from one level to another.

Reliable operation of the power grid is complex and demanding for fundamental reasons: -

First, electricity flows at the speed of light and it is not economically storable in large quantities. Therefore electricity must be produced the instant it is used.

Second, the flow of (alternating current) electricity cannot be controlled like a liquid or gas by opening or closing a valve in pipe, or switched like calls over a long-distance telephone network. Electricity flows freely along all available paths from the generators to the loads in accordance with the laws of physics- dividing among all connected. flow paths in the network, in inverse proportion to the impedance (resistance plus reactance) on each path.

Maintaining reliability is a complex enterprise that requires trained and skilled operators, sophisticated computers and communications, and careful planning and design.

One of the key aspects to maintain system reliability is to balance power generation and demand continuously. Demand is somewhat predictable, appearing as a daily demand curve. Failure to match generation to demand causes the frequency of an ac power system (nominally 50 cycles per second or 50 Hz) to increase (when generation exceeds demand) or decrease (when generation is less than demand). Random, small variations in frequency are normal, as loads come on and off and generators modify their out put to follow the demand changes. However, large deviations in frequency can cause the rotational speed of generators to fluctuate, leading to vibrations that can damage generators turbine blades and other equipments. Extreme low frequencies can trigger automatic under frequency "load shedding" which takes blocks of consumers off-line in order to prevent total collapse of the system.

The purpose of protective relays and relaying systems is to operate the correct circuit breakers so as to disconnect only the faulty over loaded equipment from the systems as quickly as possible, thus minimizing the trouble and damage caused by the fault when they do occur.

# CHAPTER 2

# LOAD FREQUENCY RELATIONSHIP

When a total or partial loss of generation occurs with in the system, the first indicators are a drop in voltage and in frequency. However, given that voltage drops can also be caused by system faults, it is generally recognized that a drop in frequency is more reliable indication of loss of generation. A sudden loss of generation in the system will result in a reduction in frequency at a rate of change, which depends on the size of the resultant overload and the inertia constant of the system.

## 1. Frequency – Load relationship

Variation of frequency with time following a sudden variation in load and/or generation

$$\frac{GH}{\pi f_0} \times \frac{d^2\delta}{dt^2} = P_A \qquad (1)$$

G = Nominal MVA of machine under consideration

H = Inertia constant

$\delta$ = Generator torque angle

$f_0$ = Nominal frequency

$P_A$= Net power accelerated or decelerated

The speed of machine at any instant (W)

$$W = W_0 + \frac{d\delta}{dt}$$

$$= 2\pi f \qquad (2)$$

$W_0$ = synchronous speed is nominal speed at rated frequency.

Differentiating equation (2) w.r.t time

$$\frac{dW}{dt} = \frac{d^2\delta}{dt^2}$$

$$= 2\pi \frac{df}{dt} \qquad (3)$$

substitute (3) in (1)

$$\frac{df}{dt} = \frac{P_A f_0}{2GH} \qquad (4)$$

In case of group of generators

$$H = \frac{H_1 MVA_1 + H_2 MVA_2 + \ldots.. + M_n MVA_n}{MVA_1 + MVA_2 + \ldots\ldots. + MVA_n}$$

The accelerating power $P_A$ in equation (4) is responsible for frequency variation

$$P_A = P_M - P_E \qquad (5)$$

$P_M$ = Mechanical Power entering the generator

$P_E$ = Electrical Power leaving the generator

Under stable conditions $P_A = 0$, no frequency variation. In case of over loads, $P_E > P_M$, thus $P_A < 0$, hence drop in system frequency.

## CHAPTER 3

# FREQUENCY MEASUREMENT TECHNIQUES

The increased growth of power system in both size and complexity imposes the requirement of high-speed relays to limit damage to major equipments or, in order to preserve system stability. Many techniques have been developed to measure the power system frequency such as:

(a) Zero crossing.
(b) Level crossing.
(c) Least Error Squares.
(d) Fast Fourier Transform (FFT).
(e) Digital Fourier Transform (DFT).
(f) Kalman Filtering.
(g) Numerical Optimization (Newton type).
(h) Demodulation.
(j) Phase locked loop.
(k) Quadrature phase locked loop.
(l) Iterative DSP techniques.

Although many methods have been developed with various modifications only some of the important techniques have been mentioned above. In the following sections some these techniques will be elaborated.

### 1. Least Error Squares (LES)

The technique is based on least error squares curve fitting method, which was earlier used for impedance protection of transmission lines transformer and differential protection of transformer (1). The method is extended to simultaneously measure frequency and amplitude of the system voltages from sampled values. Simultaneous measurement of voltage and frequency is useful in detecting over excitation of transformer. This is also suitable for protection of generators during start up, shut down operations, and while operating off-line for spinning reserve.

### 1.1 Algorithm

Assuming that system frequency does not change during a data window used for measurement, the system voltage can be represented as:

$$V(t) = Vm \; Sin(\omega t + \theta) \qquad (3.1)$$

The above expression for system voltage can further be expanded using basic trigonometric identities

$$V(t) = Vm\,(Cos\theta)\; Sin\;(2\pi ft) + Vm\;(Sin\theta)\,Cos\,(2\pi ft) \qquad (3.2)$$

Using the Taylor series, Sin ($\pi$ft) and Cos ($2\pi$ ft) can be expanded in the neighborhood of an expected value $f_o$. Restricting the series to first three terms, Sin ($2\pi$ ft)and Cos ($2\pi$ ft) can be approximated by

$$Sin\,(2\pi ft) = Sin\,(2\pi f_ot) + 2\pi t\,(f - f_o)\,Cos\,(2 f_ot) - (2\pi t/2)^2 (f - f_o)^2\, Sin\,(2\pi f_ot) \qquad (3.3)$$

$$Cos\,(2\pi f t) = Cos\;(2\pi f_ot) - 2\pi t\;(f-f_o)\,Sin\;(2\pi f_ot) - (2\pi t/2)^2\,(f-f_o)^2\,Cos\,(2\pi f_ot) \qquad (3.4)$$

By substituting equation 3.3 & 3.4 in 3.2 we can get

$$\begin{aligned} V(t) = {} & (Sin\,(2\pi f_ot))\,Vm\,Cos\theta + (2\pi t\,Cos\,(2\pi f_ot))\,(f-f_o)\,Vm\,Cos\,\theta + \\ & (t^2\,Sin\,(2\pi f_ot))\,(-2\pi/2)^2 f^2 + (2\pi)^2 f\,fo - (2\pi)^2 f_o^2)\,Vm\,Cos\,\theta \\ & + (t^2\,Cos(2\pi fot))\;(-2\pi/2)^2\,f^2 + (2\pi)^2 f\,fo\,(-(2\pi)^2/2)\,f_o^{\,2})\,Vm\,Sin\,\theta \end{aligned} \qquad (3.5)$$

Assuming $\Delta t$ as the sampling interval, a set of 'm' samples may be designated as V ($t_1$), V($t_1 + \Delta t$), V($t_1$+2$\Delta t$) .... V ($t_1$+ m $\Delta t$) where $t_1$ is an arbitrary time reference. The voltage sampled at t=$t_1$ can now be expressed by substituting $t_1$ for t in equation (3.5). By mathematical manipulation

$$V(t_1) = a_{11}\,x_1 + a_{12}\,x_2 + a_{13}\,x_3 + a_{14}\,x_4 + a_{15}\,x_5 + a_{16}\,x_6 \qquad (3.6)$$

Where $x_1 = V_m \cos\theta$

$x_2 = (f-f_o) V_m \cos\theta$

$x_3 = V_m \sin\theta$

$x_4 = (f-f_o) V_m \sin\theta)$

$x_5 = \left(-\frac{(2\pi)^2}{2} f^2 + (2\pi)^2 f f_o - \frac{(2\pi)^2}{2} f_o^2\right) V_m \cos\theta$

$x_6 = (-(2\pi)^2 f^2 + (2\pi)^2 f f_o + (2\pi)^2 f_o^2) V_m \sin\theta$

$a_{11} = \sin(2\pi f_o t_1)$

$a_{12} = 2\pi t_1 \cos(2\pi f_o t_1)$

$a_{13} = \cos(2\pi f_o t_1)$

$a_{14} = 2\Pi t_1 \sin(2\pi f_o t_1)$

$a_{15} = t_1^2 \sin(2\pi f_o t_1)$

$a_{16} = t_1^2 \cos(2\pi f_o t_1)$

Voltage sample at $t_2 = (t_1 + \Delta t)$ can be expressed as

$$V(t_2) = a_{21} x_1 + a_{22} x_2 + a_{23} x_3 + a_{24} x_4 + a_{25} x_5 + a_{26} x_6$$

Proceeding in this manner, m digital values of the voltage sampled from the system can be expressed as m equations in six unknowns. These equations in matrix form:

$$\underset{M \times 6}{[A]} \quad \underset{6 \times 1}{[X]} = \underset{m \times 1}{[V]} \qquad (3.7)$$

The elements of the matrix (A) depend on the time reference and sampling rate used and can be predetermined in an off line mode. Six samples of the voltage would be required to determine the six unknowns.

Assuming 'm' is greater than six the matrix (A) is now a mx6 rectangular matrix. Pre-multiplying both sides of the equation 6 with the left pseudo – inverse of (A), values of the unknown can be determined as

$$(X) = (A)^{+} (V)$$

$$\text{Where } (A)^{+} = [(A)^{T} (A)]^{-1} [A]^{T}$$

By determining all the elements of (x), the system frequency and frequency deviation can be calculated as under

$$f - fo = \frac{x_2}{x_1} \quad \text{or} \quad f\text{-}fo = \frac{x_4}{x_3}$$

$$f = fo + \frac{x_2}{x_1} \quad \text{or} \quad f = fo + \frac{x_4}{x_3}$$

Another approach to use all the variables $x_1$, $x_2$, $x_3$, and x4 to estimate f-fo is

$$(f\text{-}fo)^2 = \frac{x_2^2 + x_4^2}{x_1^2 + x_3^2}$$

## 2. Extension of Measurement Range of the LES Technique

In LES technique, the measurement error increases as the frequency deviation form the nominal value increases. In the proposed technique, the error expected to be present in the estimated values of the true frequency are pre-determined and correct values of the frequency are stored in a look up table. When frequency deviations are estimated, the corresponding value of the true frequency is obtained from look up table. This approach increases the useful measurement range of the technique substantially (3). This method is useful in designing digital meters and relays that need to measure system parameters accurately over a large frequency range.

## 3. Level Crossing Method

The method is based on a generalization of thee zero crossing detection to a level crossing detection. This yields several estimates of the frequency within one cycle. A composite best estimate is obtained by an appropriate-weighted average of these estimates. The method is insensitive to amplitude variations unbalance and is capable of tracking small frequency deviations in as short a duration as a fraction of a cycle. The measurement scheme is particularly suitable for integration with the digital schemes for real time monitoring, control and protection of power system [4].

The explicit computation of the time period between zero crossings is supplemented by multiple computations of the time periods between various non-zero voltage levels crossings. The different time periods computed are of different degrees of precision for non-triangular waveforms. In practice, those near the zero crossing are the most precise and a weighting function inversely related to the expected errors is used to obtain a best average estimate of the time period of signal. In case of a three-phase signal, the number of estimates to be averaged is an arbitrary multiple of three, and over any fraction of cycle. Immunity to amplitude variations is obtained by normalizing the signals at every sampling instance. Crystal controlled clock is used for sampling the waveforms at precise fixed time intervals.

The block diagram of the frequency measurement scheme is given in *Fig 1.* This method of computation of frequency from time period deviation estimation is particularly attractive when implemented with other power system on line tasks involving analog- to-digital conversion of voltage and waveforms.

This method yields the same result as the conventional zero- crossing time interval technique under ideal conditions. The major differences in this approach are: -

(a) Frequency estimate is available at fraction of cycle in case of single-phase measurement.

(b) Immunity to high frequency and random waveform variations is achieved by a weighted averaging process of several estimates.

(c) Frequency estimate is available at each sampling instant in case of three-phase measurement.

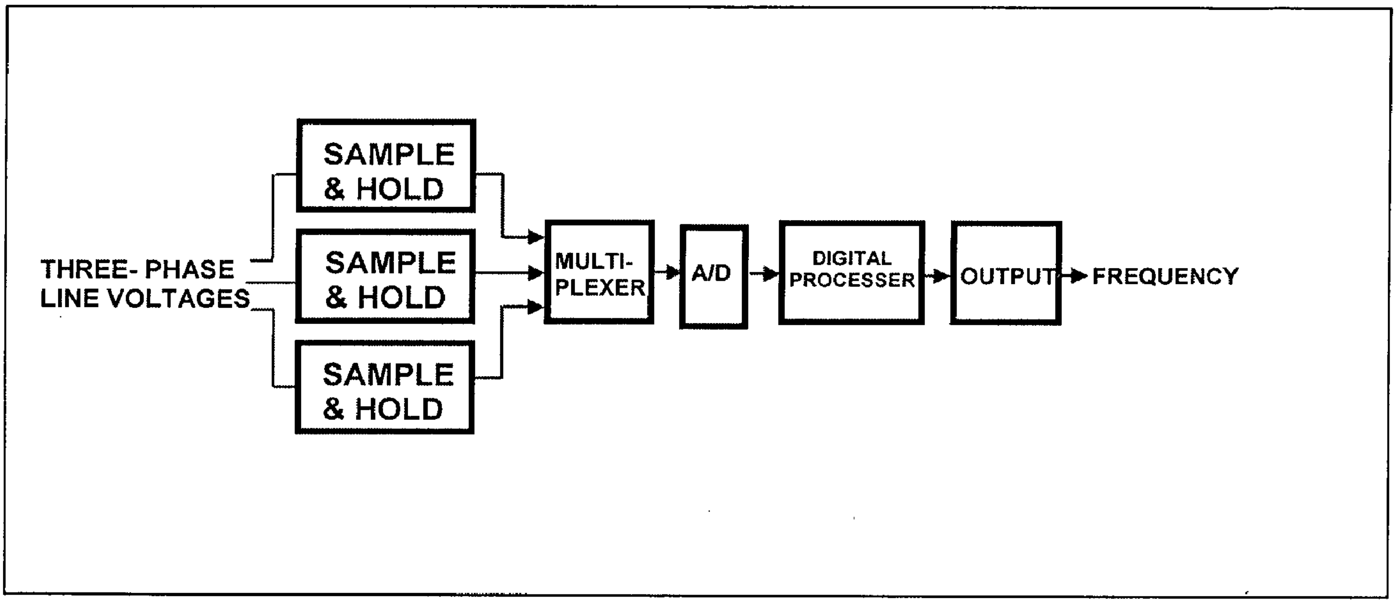



**FIG 1 - A FREQUENCY MEASUREMENT SCHEME**

## 4. Fast Fourier Transform Method

The change in power system frequency is detected by relating it to the leakage coefficient in the FFT. This method in used to detect the deviation of the power system frequency from the fundamental frequency and the best estimates of its rate of change. Leakage is one the major pitfalls associated with the use of FFT(5). In this method, the leakage is used advantageously for detecting fluctuations in the fundamental frequency of the system. Considering the oscillatory nature, this method is developed which also optimally estimates the mean frequency and its average rate of change. The optimal estimates are provided by Kalman Filter. (6)

**Determination of frequency fluctuations.** Frequency fluctuation is determined by the leakage coefficient. Leakage coefficient is defined as under: -

$$5 = \left(\left(\sum_{k=0}^{N-1} |X(k)| - |X(1)|\right)\right) / |X(1)|$$

X (k) denotes the DFT and K=1 corresponds to the fundamental frequency. If 5 =o, no leakage has occurred, which implies that the frequency component at the nominal frequency has not deviated. Whenever the nominal frequency component deviates by $\pm \Delta f$, the leakage coefficient (5) will be non-zero. Leakage coefficient gives the magnitude of frequency deviation only. To determine if the fundamental frequency component has increased or decreased the real part of X(1) must be examined. If real (X (1)) < (0), then the fundamental component has decreased, whereas if real (X (1))> (0), indicates that it has increased.

**Best Estimate of frequency and Rate of change** To minimize the errors introduced by the frequency oscillations an algorithm based on Kalman filtering is used to compute the best estimate of the smoothed frequency and its average rate of change.

## 5. Iterative DSP technique

It is necessary to detect under frequency and over frequency conditions very fast so that necessary corrective actions can be taken. Quick estimates of frequency would provide extra time at the hands of the operators to take corrective action. The protection should be so designed that the time taken is inversely proportional to amount of deviation by frequency.

The technique uses digitized values of samples of the voltage taken at a pre-specified sampling rate. As in LES technique, it is possible to extract the real and imaginary parts of the fundamental frequency component of a signal. The real and imaginary parts computed using samples corresponding to the $n^{th}$ data window can be used to compute phase angle $\theta n$ of the fundamental frequency phasor (corresponding to the nth window). As the next sample arrives, the data window in shifted by one sample. The phase angle, $\theta_{n+1}$, of the fundamental frequency phasor corresponding to the (n+1)th window and coefficients of the sine and cosine filters (7 ).

The phase angle difference, $(\theta_{n+1}-\theta n)$, represents the rotation of the phasor in one sampling interval. This difference can be used to obtain an estimate of the frequency as follows: -

$$f = (\theta_{n+1}-\theta n)/ (2\pi/fs)$$

fs = is the sampling rate.

If the estimated frequency is equal to the fundamental frequency assumed for designing the LES filters that are used to compute phase angles $\theta n$ & $\theta_{n+1}$, this implies that the estimated frequency is also the fundamental frequency of the signal. In case estimated frequency is not equal to fundamental frequency, phase angles need to be calculated using LES filters designed by assuming the fundamental frequency being equal to fundamental frequency of the signal. This can be achieved by using an iterative procedure.

The proposed technique when tested using signals from a dynamic frequency source, has shown that accurate estimate can be done within 20 ms. Maximum frequency estimation error observed during testing are of the order of 0. 01 Hz for nominal and near nominal frequency and are in the range of 0.03-0.05 Hz for off nominal frequency. This technique is general in form and can be designed to provide accurate estimate when presence of harmonic components is anticipated in the input signal.

## 6. Quadrature Phase Locked Loop (QPLL)

The technique accommodates the inherent non-linearity of the frequency estimation problem. This makes the method capable of providing a fast and accurate estimate of the frequency when its deviation from the nominal value is incremental or large. The non-liner structure of the method permits direct estimation of the frequency and its rate of change without linearisation process or simplifying assumptions on the signal's model. The method is robust with respect to the external disturbances e.g., switching type transients, as well as the internal parameters of the estimator (8). QPLL The general block structure of QPLL is given in *Fig 2.*

The phase detector of (PD) of the conventional phase locked loop (PLL) system is replaced with an alternative mechanism, which enables the QPLL to provide an on-line estimate of the phase -locked fundamental component. QPLL is also capable of estimating the frequency of the input signal and its rate of change. The 90° phase shift included in the PD can be implemented by the VCO in analog implementation. In digital implementations, where look-up table or DDS is used to generate the sine function, this phase shift can be implemented by proper adjustment of the look up table. Behavior of the QPLL is controlled by two internal parameters $K_a$ & $K_f$. $K_a$ controls the speed of convergence of the amplitude of the response while $K_f$ controls that of the frequency for a given set of parameters, time-constants of the frequency. For a given set of parameters, time-constants of the QPLL are proportional to $T_a = 1/K_a$ and $T_f=1/K_f$. A larger value of each parameter provides a faster convergence of the corresponding variable. However, faster convergence is accompanied by a larger steady state error. Steady state error is due to presence of noise and distortions in the input signal of the estimator. The PD of the QPLL is comprised of two branches for estimating in phase and quadrature phase amplitudes of the fundamental component of the input signal, which in combination with the estimated frequency generate the fundamental component. The three state variables of QPLL system are the two amplitudes and frequency. These variables are assumed to have relatively slower time variations in comparison with the input signal variation.

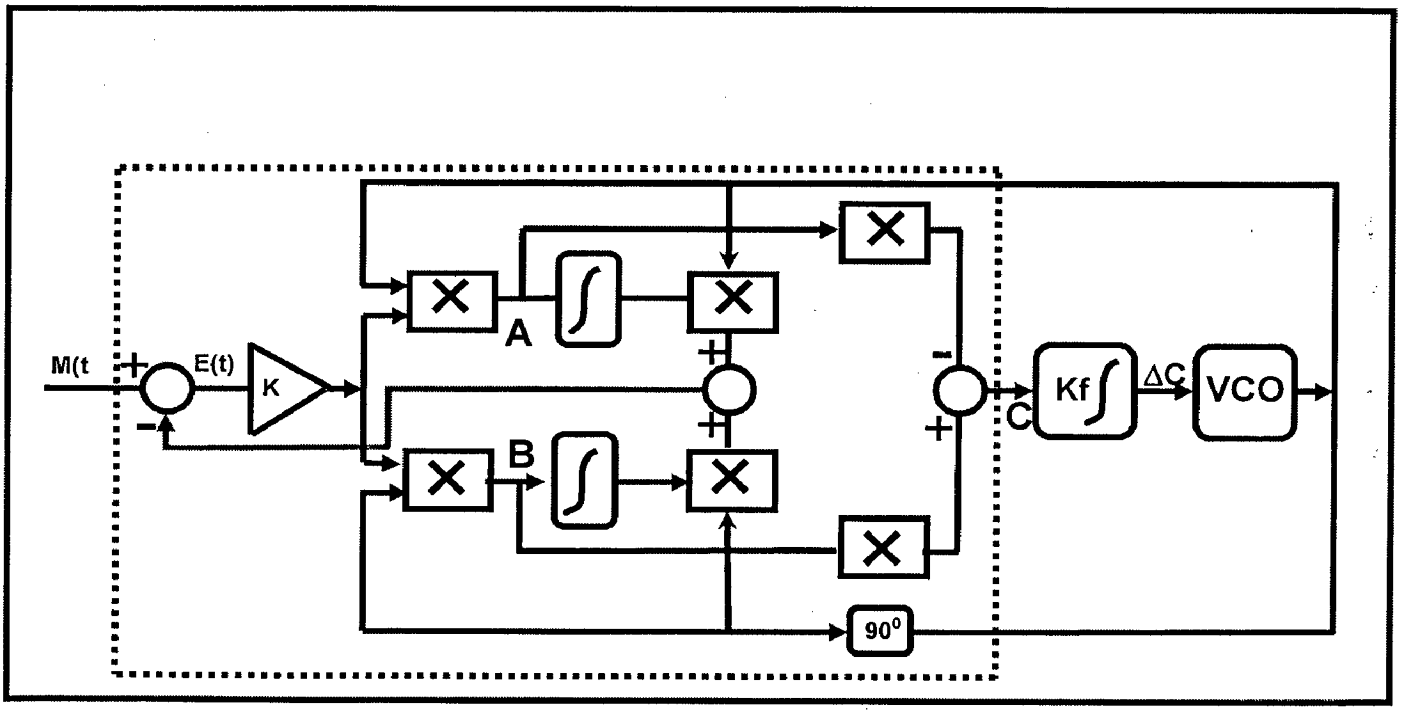



FIG 2 QUADRATURE PHASE LOCK LOOP

Employment of appropriate low pass filters (LPFs) enables the estimator to operate with adequate speed and accuracy in highly polluted environment. LPFs are located at three points A, B, C, as shown on fig. LPFs are first order filters with cut off frequency of about 30Hz. A band pass filter (BPF) is used at the input terminal to improve the performance of the frequency estimator. A second- order BPF with the cut off frequency of 10Hz & 110 Hz provides the desired characteristics. The main advantages of such a filtering stage are, prevention of the adverse impacts of the dc components and enhanced robustness of the estimator with respect to harmonics and noise.

## 7. Measurement from Estimates of Voltage and Current Phasors

The cosine of the sampling angle, angle between phasors estimated at two consecutive sampling instants, is calculated from three and five consecutive real components of phasors. The information is than used to estimate the frequency (9 ).

The phasor quantities are used for estimating successive sampling angles. To accurately estimate the phasor quantities a specially designed least error square (LES) filter in conjunction with a Decimation filter is used.

The mathematical basis of the algorithm is :-

The voltage phasors at the rth & (r+1)th instants can be written as.

$$V(r) = Vc^{(r)} + j\,Vs^{(r)}$$

$$V(r+1) = Vc^{(r+1)} + jVs^{(r+1)}$$

Where Vr = cosine/real component of the phasor

Vs = Sine/ imaginary component of the phasor

If $\Delta T$ in the sampling interval, moving the data window one sample forward advances the voltage phasor in a counter clockwise direction by the sampling angle. The phasor at the (r+1)th instant can therefore be expressed as

$$V(r+1) = e^{j(\omega/fs)}\,V(r)$$

By using three consecutive estimates of the phasor, and using real/cosine and imaginary/sine components of the voltage phasor.

$$\text{Cos}^{(\omega/fs)} = \frac{Vc^{(r+1)}+Vc^{(r-1)}}{2Vc^{(r)}} \quad - \qquad (1)$$

$$\text{Cos}\left(^{(\omega/fs)}\right) = \frac{Vs^{(r+1)}+Vs^{(r-1)}}{2Vs^{(r)}} \quad - \qquad (2)$$

By using five consecutive samples

$$\text{Cos}^{(\omega/fs)} = \frac{Vc^{(r-2)}-Vc^{(r+2)}}{2(Vc^{(r-1)}-Vc^{(r+1)})} \quad - \qquad (3)$$

$$\text{Cos}\left(^{(\omega/fs)}\right) = \frac{Vs^{(r-2)}-Vs^{(r+2)}}{2(Vs^{(r-1)}-Vs^{(r+1)})} \quad - \qquad (4)$$

Using Equations 1 and 4 cosines of the sampling angle can be estimated. Subsequently, frequency can be calculated either using real or imaginary components of the phasor.

The test result of this technique has shown that the algorithm is particularly useful for monitoring sub- transmission and distribution system and for calculating frequency when harmonic components can be signification by high. It can be implemented in real time application where several channels can be successfully monitored and the frequency for all at every sampling instant.

Among the first techniques for frequency measurements were those based on Zero crossing. They were gradually abandoned due to their sensitivity to noise, presence of DC Components in the signal, and harmonics. However, their inherent simplicity cannot be matched by any other technique. When combined with a data smoothing technique, Zero crossing may produce surprisingly good performance. A variation of the same method involves frequency multiplication of the measured signal using PLL, which reduces the measurement time, but does not very good resolution or dynamic properties.

# CHAPTER 4

# FREQUENCY MEASUREMENT :DEMODULATION METHOD

### Frequency Estimation by Demodulation of Two Complex Signal

In power system analysis, the $\alpha\beta$ transform is used to convert three phase quantities to a complex quantity where the real part is the in phase component and the imaginary part is the quadrature component, This complex signal is demodulated with a known complex phasor rotating in opposite direction to the input. The advantage of this method in that the demodulation does not introduce a double frequency component. For signals with high signal to noise ratio, the filtering demand for the double frequency component can often limit the speed of the frequency estimator.

The idea to multiply two complex signals bears resemblance with single side band modulation [11].

### Traditional Demodulation

The idea of traditional demodulation is to multiply the scalar input with a sine and cosine signal – that can be interpreted as a complex exponential – with a known frequency. The method is shown in *Fig 3.*

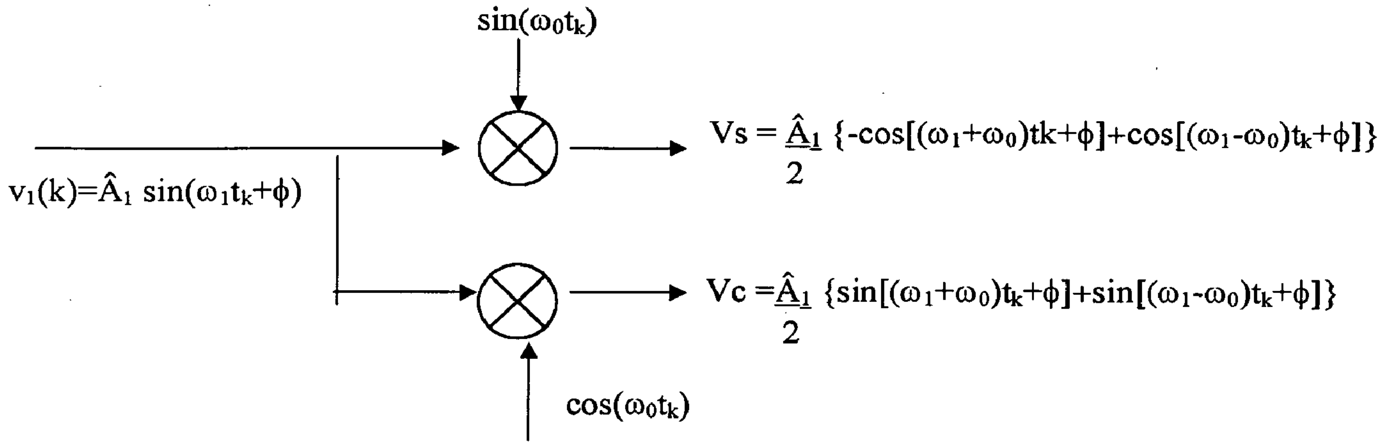



**Figure 3. Normal demodulation (Correlation)**

The resulting signal has two parts, Vs and Vc, that each contains two components of different frequency. One component of the frequency difference and one with the sum. The signal is then low-pass filtered, or band-stop filtered, to reduce the double frequency component. The remaining component in Vs and Vc is used to estimate the unknown frequency. More details about demodulation and aspects on filter design are given in [12, 13, 14].

The paper [12] describes a study to find a frequency estimation algorithm that could be used for the Electricité de France's defence plan against loss of synchronism. An extensive study of different algorithms was done. The final choice of algorithm was demodulation with a fixed filter to reduce harmonics and an adaptive filter to remove the double frequency component of demodulation.

The paper [13] concludes that frequency estimation by demodulation is "capable of transient performance expected for monitoring of real time power dynamics. Further improvements are possible with thorough analysis of the filtering options of the demodulation method".

This motivates that demodulation is a promising method for frequency estimation. It is also stressed that the filtering of the double frequency component in demodulation needs thorough analysis and can very well be the limiting factor of demodulation. This is also reflected in [12] where a large part of the paper describes alternative designs of these filters.

**New demodulation of two complex signals: Algorithm**

Consider $v_1(k)$, $v_2(k)$, $v_3(k)$ to be discrete samples of the three phase voltages that have antialiasing filter

$$V_i(k) = \hat{A}_i \sin(\omega_1 t_k + \phi] + ei(k), \ i = 1, 2, 3 \tag{1}$$

where $e_i$ is a general noise term that can be any combination of white noise and harmonics.

The $\alpha$, $\beta$-components are defined as the complex voltage $V(k) = V\alpha\ (k) + jV_\beta(k)$ (2)

where the real and imaginary parts are calculated from

$$\begin{pmatrix} V\alpha\ (k) \\ V\beta\ (k) \end{pmatrix} = \sqrt{2/3} \begin{pmatrix} 1 & -\frac{1}{2} & -\frac{1}{2} \\ 0 & \sqrt{3/2} & -\sqrt{3/2} \end{pmatrix} \begin{pmatrix} v1\ (k) \\ v2\ (k) \\ v3\ (k) \end{pmatrix} \tag{3}$$

In the literature, the complete transformation is often called the $\alpha\beta$ – transform and then also includes the zero sequence components. In the application only the two perpendicular parts, $\alpha$ and $\beta$ are used and the zero sequence components is left out in the transformation.

$V_\alpha$ and $V_\beta$ can contain plus and minus sequence voltage, but not any zero sequence components. Hence, harmonics that are mainly zero sequences - such as the third harmonic - are blocked by the $\alpha\beta$ – transformation.

To make the analysis straightforward it is assumed that the input voltages $v_1$, $v_2$, $v_3$ do not have any negative sequence voltage nor any noise. We then have

$$V(k) = A[\cos(\omega_1 t_k+\phi]j\sin(\omega_1 t_k+\phi)] \tag{4}$$

$$= A\, e^{j(\omega 1 tk+\phi)}$$

Where A is the phase-to-phase RMS-value.

The demodulation is done with a complex signal Z, that rotates in the opposite direction, i.e., negative sequence, compared to the input signal Vas shown in *Fig 4.*

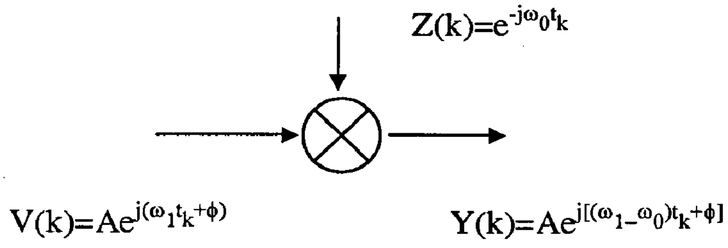



**Figure 4. New demodulation of two complex signals.**

The signal Z with a known frequency $w_0$ is

$$Z(k)=\cos(-\omega_0 t_k)+j\sin(-\omega_0 t_k)=e^{-j\omega_0 t_k} \tag{5}$$

The resulting signal, Y, after the multiplication becomes

$$Y(k)=V(k).Z(k) = Ae^{j(\omega_1 t_k+\phi)}e^{-j\omega_0 t_k}$$

$$=Ae^{j[(\omega_1-\omega_0)t_k+\phi]}$$

$$= A\{\cos[(\omega_1-\omega_0)t_k+\phi]+j\sin[(\omega_1-\omega_0)t_k+\phi]\} \tag{6}$$

Note that the demodulation does not create the double frequency component. Hence, the demodulation does not add demands to filter away the double frequency component. however, there still might be a need to filter due to noise. The frequency estimation is done as in [13]. To find the phase difference, we define the complex variable U as.

$$U(k)=Y(k).Y(k-1)^* \qquad (7)$$

where * stands for conjugate. We separate Y in real and imaginary part and find that

$$\begin{aligned} U(k) &= Re[Y(k)]Re[Y(k-1)]+Im[Y(k)]Im[Y\{k-1)] \\ &+j\{Im[Y(k)]Re[Y(k-1)]-Re[Y(k)]Im[Y(k-1)]\} \end{aligned} \qquad (8)$$

The phase difference between two consecutive samples is calculated from the real and imaginary part of U.

$$\gamma(k)-\gamma(k-1)\cong\arctan\left(\frac{Im[U(k)]}{Re[U(k)]}\right) \qquad (9)$$

The deviation in angular frequency is estimated from

$$\Delta\omega(k\text{-}0.5) \cong \frac{1}{\Delta t}[\gamma(k) - \gamma(k\text{-}1)]= fs[\gamma(k) - \gamma(k-1)] \qquad (10)$$

The time index k-0.5 is used to point out that the estimate is best in the middle of the time interval [k-1,k]. For real-time application we are restricted to causal relations and get

$$\Delta\omega(k) \cong \frac{1}{\Delta t}[\gamma(k) - \gamma(k\text{-}1)]= fs.[\gamma(k) - \gamma(k-1)] \quad (11)$$

The unknown frequency for the signal V is estimated as

$$f(k)\cong f_0+\frac{f_s}{2\pi}\arctan\left(\frac{Im[U(k)]}{Re[U(k)]}\right) \qquad (12)$$

where $f_0$ is the nominal, and $f_s$ is the sampling frequency.

# CHAPTER 5

# SIMULATION OF THE ALGORITHM

## 1. Simulation

The suggested technique has been implemented with Matlab code. The programme is given in Appendix.

Strengths and weaknesses of the suggested technique are illustrated with the following examples.

(a) **Example 1** The phase voltages are: -

$vi(k) = \sqrt{2}$ Arms Sin $(\Phi i(k)) + Ni(m,\sigma)$, $i = 1,2,3$

Where the angles are calculated from

$\Phi 1\ (0) = \Phi i\ (k-1) + \omega(k)\,\Delta t$ ; for $k \geq 1$

With the initial values

$\Phi_1(0) = 0,\ \Phi_2(0) = \frac{-2\pi}{3},\ \Phi_3(0) = \frac{2\pi}{3}$

The frequency is time varying

$$\omega(k) = 2\pi\left[50 + \text{Sin}\,(2\pi.1.t_k) + 0.5\ \text{Sin}\,(2.\ \pi.6.t_k)\right]$$

The notation N $(m,\sigma)$ is used for normally distributed white noise. $\sigma$ is standard deviation.

$\sigma = 0.01$ and mean $m = 0$

Signal is distorted with :-

(i) Negative sequence of 1%

(ii) White noise with SNR 40dB ;

(iii) $3^{rd}$ harmonic, 5% mainly zero sequence ;

(iv) $5^{th}$ harmonic, 2% mainly negative sequence.

(b) **Example 2** Same test signal as in example 1, but non causal filter.

## 2. Analysis of the output

### 2.1 Example 2

Simulation result is attached as Fig 5 (a) to 5 (d).

This illustrates the application of the algorithm for real time applications, such as relay protection.

### 2.2. Example 2

*Fig 5 (e)* shows the simulation result.

The filtering is performed in the Matlab package by the command *Filt filt* .The phase shift is reduced by applying forward filtering followed by backward filtering.

## 3. Summary:

The outputs of the above examples show that.

(a) This method is suitable for signals with high SNR.

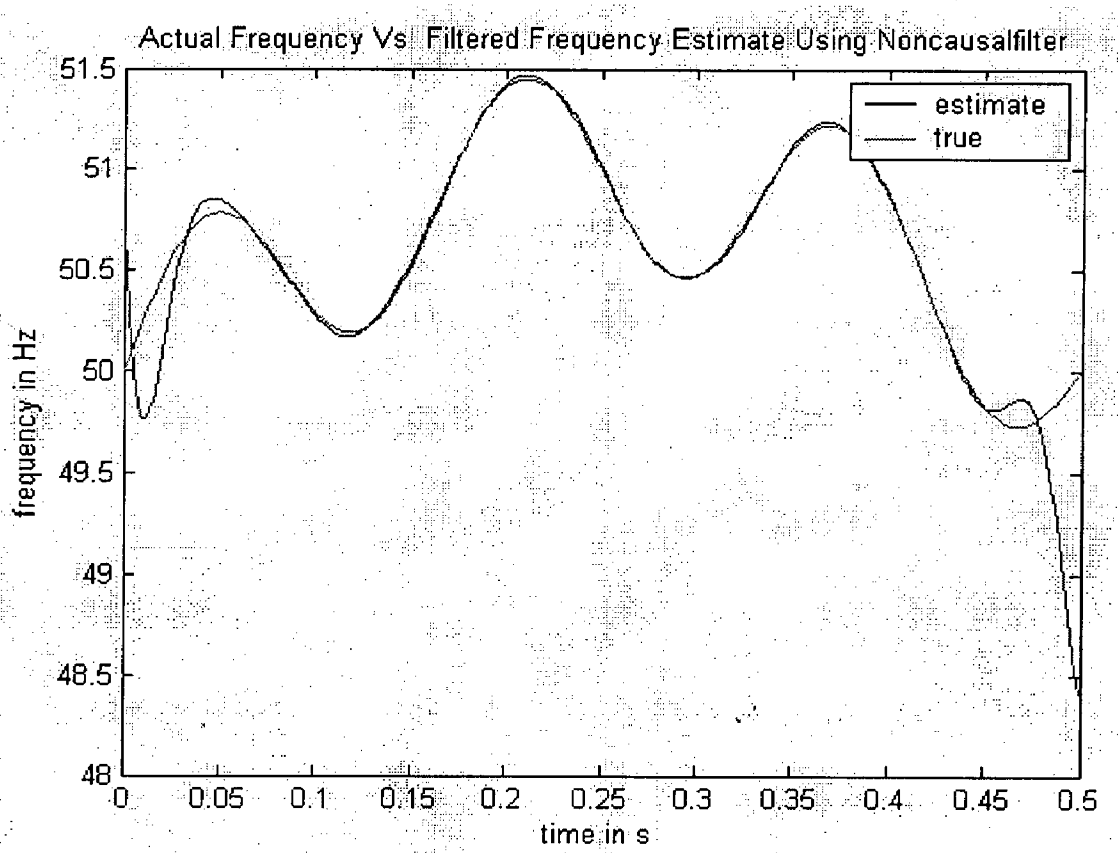
(b) The frequency estimate needs filtering when the SNR value falls below 80 dB.

(c) Casual filtering introduces a time delay. For off line calculation we can use non-causal filters to reduce the time delay.

## 4. Shortcomings

(a) Unsymmetric phase voltages-If the input contains negative sequence, the demodulation introduce a double frequency component that is proportional to the negative sequence amplitude. This gives the same type of double frequency component as the traditional demodulation method. At unsymmetrical faults, the negative sequence component can be large. In this situation the proposed method will give a double frequency component with large amplitude and will work similar to the old demodulation method.

(b) Three phase calculation - All three phase are used for the calculation. A method that uses all three phases independently can give a better estimate by using the mean value.



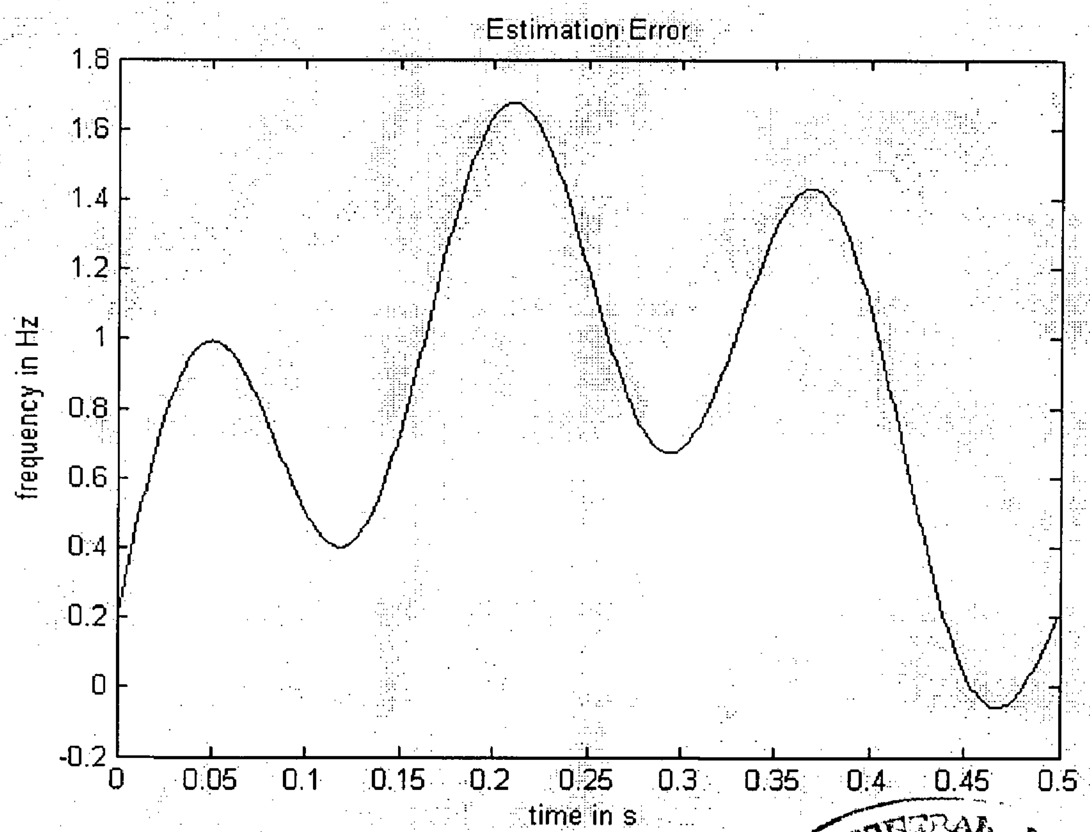




**Fig 5 (a) and Fig 5 (b)**

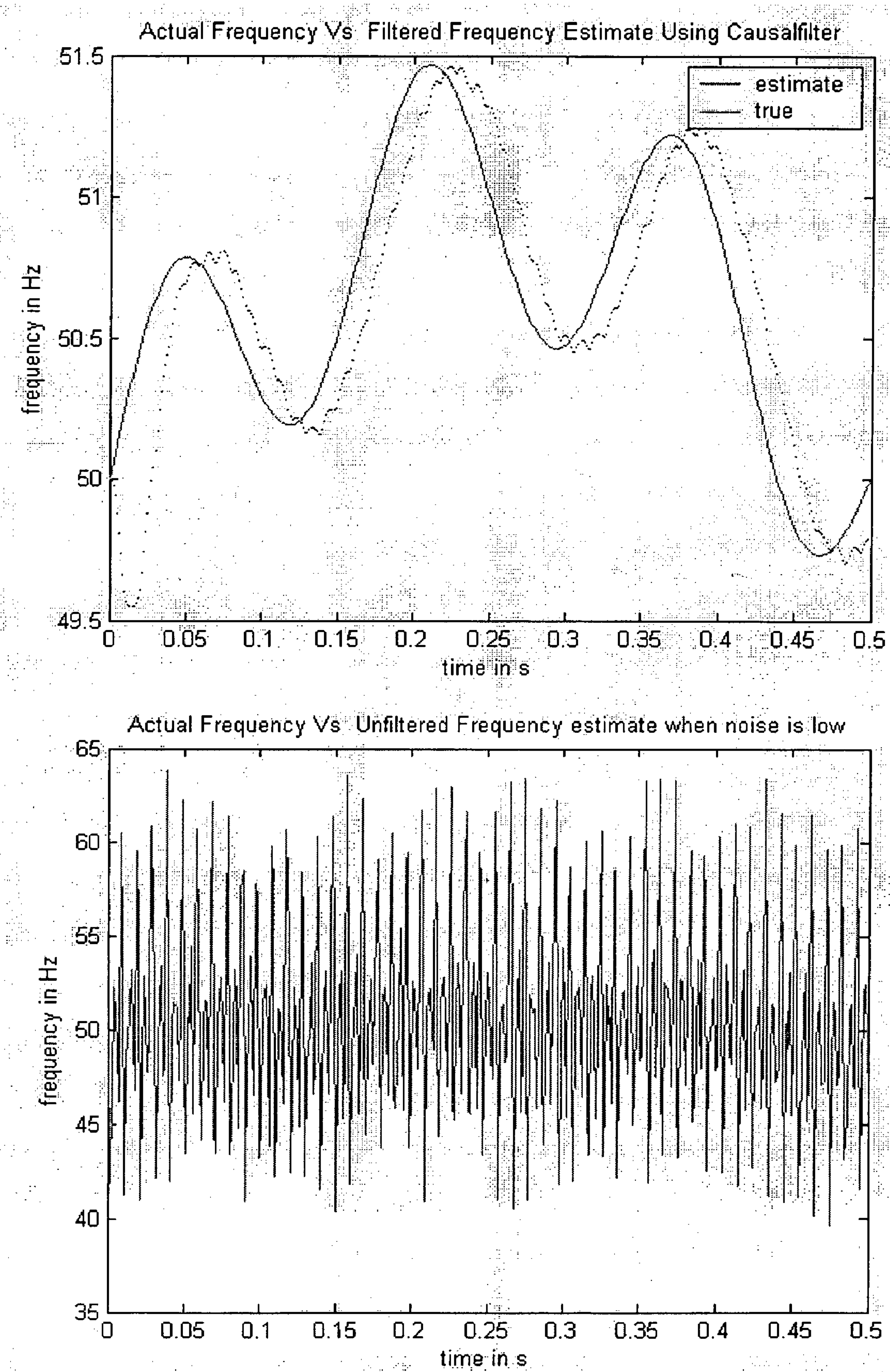

Actual Frequency Vs Filtered Frequency Estimate Using Causalfilter
estimate
true
frequency in Hz
time in s
Actual Frequency Vs Unfiltered Frequency estimate when noise is low
frequency in Hz
time in s

**Fig 5 ( c) and 5 (d)**

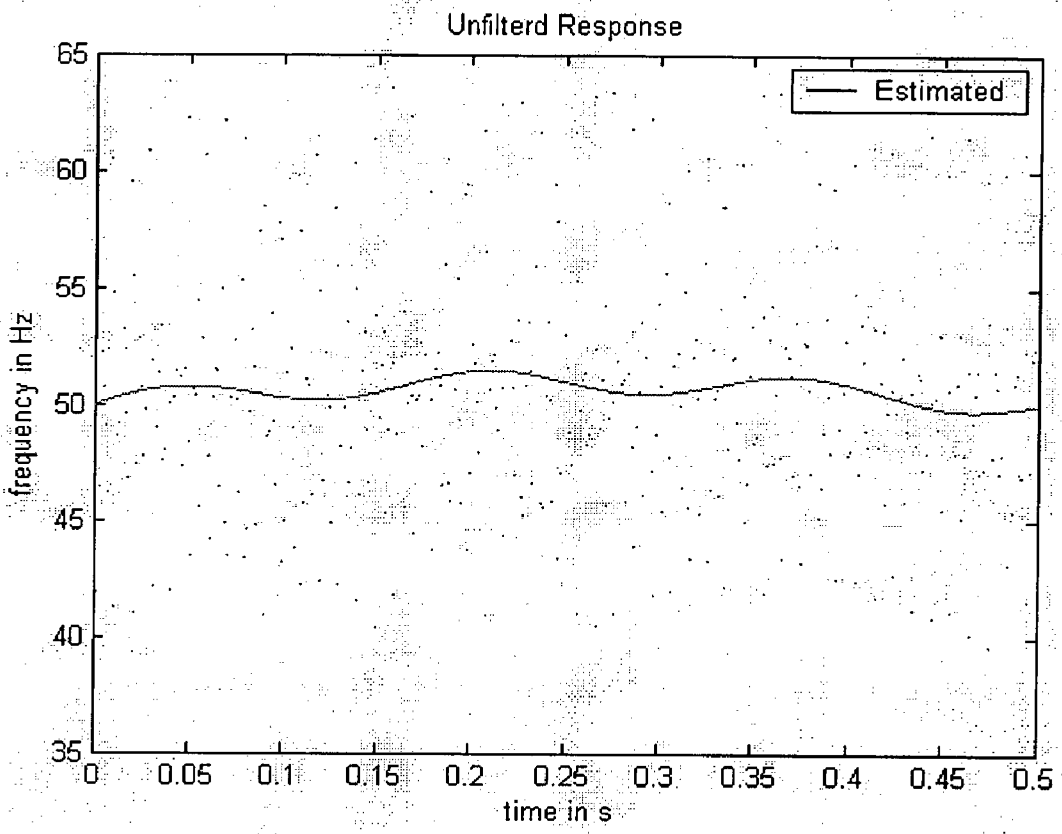

Unfilterd Response
Estimated
frequency in Hz
time in s
65
60
55
50
45
40
35
0
0.05
0.1
0.15
0.2
0.25
0.3
0.35
0.4
0.45
0.5

**Fig 5 (e)**

## CHAPTER 6

# MICRO CONTROLLER

. Micro controllers have appeared as a major off shoot in the evolution of micro coprocessor technology. As the micro processors developed in the direction of greater word widths (ie 16/32/64 bits) incorporating increasingly powerful CPUs running at higher speed, a branch a development took to combining RAM, ROM, Timers, A/D and D/A and various I/O facilities along with the CPU on a single chip. This single chip architecture is commonly referred to as Micro controllers. They are also sometimes called embedded controllers because in most common applications, micro controllers lie embedded in the machine controlling its various functions without the user getting to know about it. Simple examples of such machines are automatic washing machines, microwave oven and so on.

The on chip resources provide on integrated approach to a variety of real time control tasks. However if the on chip resources fall short for a particular application, the micro controllers can be operated in expanded mode, through which the chip resources are augmented and other required resources are added. The micro controllers have a clear edge over microprocessor in control applications and other diverse jobs.

### 1. Family overview of MCS-51

The Intel MCS-51 generic part number actually includes a whole family of micro controllers that are available in NMOS and CMOS construction. Pin out diagram for the 8051, 8751 and 8031 are shown in *Fig 6*. The devices include the following features:-

- Single supply 5 Volts operation.
- 8 bit CPU with registers A and B.
- KB of program memory on chip (not on 8031).
- 128 bytes of Data memory on chip.

- Four register banks, each containing eight registers.
- Sixteen bytes which may be addressed at bit level.
- Eight bytes of general purpose date memory.
- 128 user defined software flags.
- Data and program memory expandable to 64 KB.
- One micro second instruction cycle with 12 MHz crystal.
- 32 Bi-directional I/O lines organized as four 8- bit ports.
- Multiple mode high-speed programmable serial port.
- Two multiple mode, 16-bit time/counters.
- Two level prioritized interrupt structures.
- Direct byte and bit addressing.
- Powerful 111 instructions instruction set.

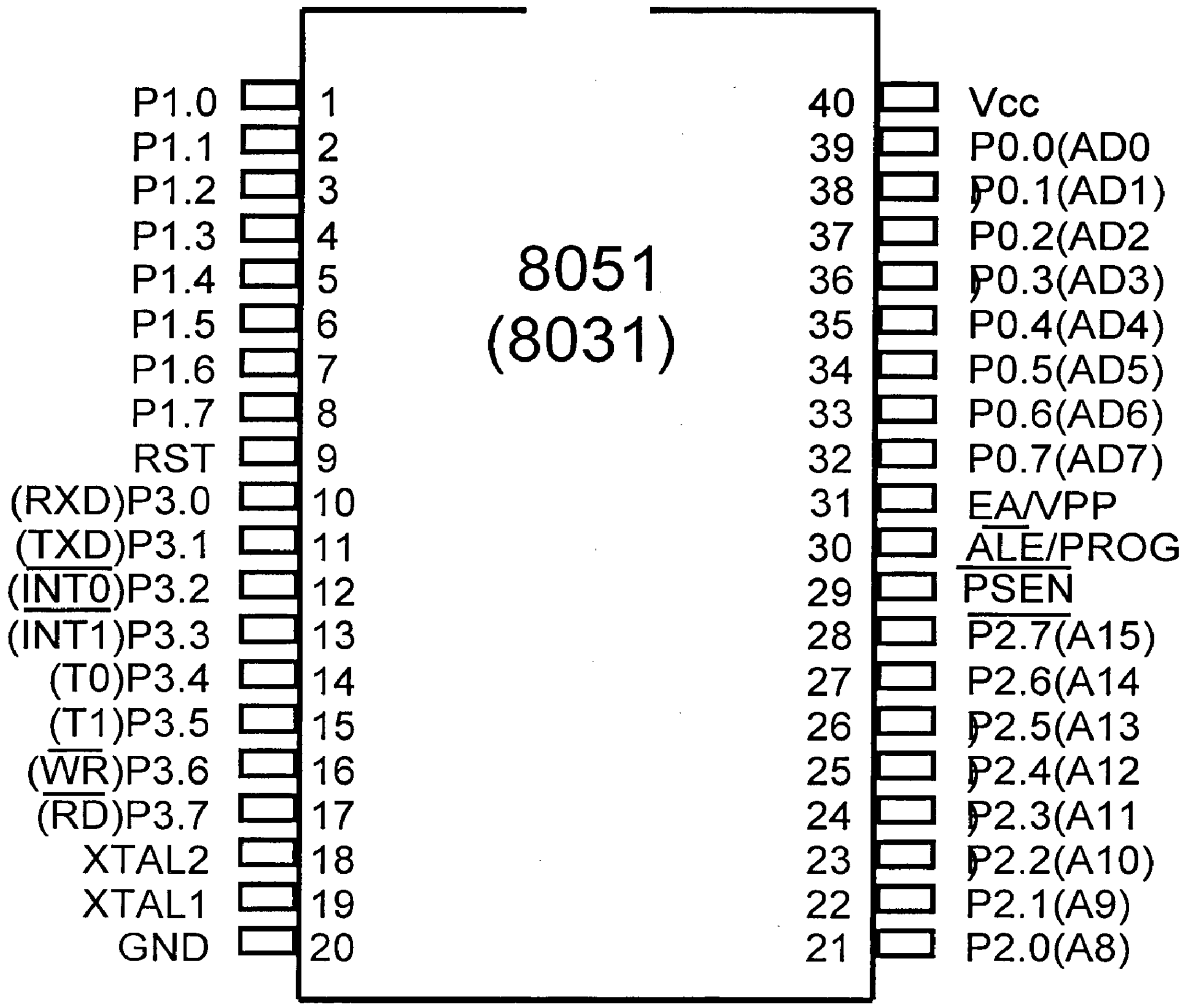



**Fig 6 - Pin Description of the 8051**

## 2. MICE-51

It is an entry-level emulator development system that provides a powerful, inexpensive set of software and hardware tools for designing with the 8051 family of micro controllers. MICE-51 comprises three components, the MICE-51 in-circuit emulator hardware, the MBUG symbolic debugger and the MBUG cross assembler. Together, these components provide the engineer with a complete microcontroller design tool set for working with microcontrollers in the 8051 family in a cost-effective package.

Using MICE-51 and MBUG one can assemble a source program, download the object code in to the MICE-51 Emulator, load the source program into the MBUG Debugger, emulate the microcontroller in the target system under the control of the MBUG Debugger, run and debug the program while it is running in the target system, set breakpoints, watch on-chip variables, single step the program and display the contents of all on-chip registers, all on-chip and all off-chip RAM.

## 3. System specifications

- CPU : 8031/8051/8731
- MEMORY ; Total memory expandable up to 64K.
- RAM : 8K Byte Space for further expansion.
- ROM : 8K Byte of EPROM
- I/O : 24 I/O Lines using 8255 expandable up to 48
- TIMER : 16 Bit timer/counter using 8253
- KEY BOARD ; 32 Hex Keys
- INTERFACE ; EPROM Programmer/ 8 Analog in put using ADC 0809/1 Analog output using DAC 0800
- EMULATOR POD ; 40 Pin Header for 8031/8051/8751

# CHAPTER 7

# IMPLEMENTATION OF FREQUENCY RELAY ON MICRO CONTROLLER

## 1. IMPLEMENTATION

Various algorithms presented in previous chapters were evaluated for their real time implementation using microcontroller. For real time implementation various aspects which need to be addressed are:

- Processor
- Clock frequency
- Functional integration of processors
- Software support

1.1 **Processor** – as algorithm to be implemented in real time, to that end, time available for processing is limited. Hence, processor to be chosen should have sufficient processing capabilities such as hard-wired multiplication and division, so that, designated task(s) can be accomplished within the available time.

1.2 **Clock Frequency,** as discussed above, while selecting the clock frequency, due care must be taken so that sufficient processing speed is achieved to complete the execution of relaying algorithm before next set of data is made available for processing. The other aspect is the maximum clock speed of the processor chosen and for this, processors are operated generally at its maximum clock speed.

1.3 **Functional Integration of Processor**: In general, speed of hard-wire besides being dependent upon clock frequency, is also affected by external device(s), which need to be interfaced. As, while interfacing, external devices such as additional gates, flip-flops and buses are to be used. All these result in propagation delay. Therefore, processors with on board integration of peripheral devices are considered to be the best choice for an application like protective relaying. Yet another aspect to be taken care of is capability of processors, which allows single bit manipulation. To that end, for our application micro controller 8051 is the best choice. Functional details of this microcontroller has been given at *Fig 7.*

1.4 **Software Support:** Despite having all the best features available, without proper software support no microcontroller can be used. Keeping this aspect also in mind, due to availability of C compiler of 8051 microcontroller, it was an automatic choice for the implementation of the microcontroller based frequency relay.

## 2. Frequency Measurement Scheme

Among the various schemes available for the measurement of frequency of power system, zero crossing detector method is considered to be the best. In this method, time period for one cycle of unknown frequency signal is measured using clock frequency signal. Since the clock frequency signal is usually of very high value, therefore, a significantly good resolution is obtained. The scheme consists of a hard-wired zero crossing detector and software for measuring time period for one cycle of input wave formed.

## 3. Zero Crossing Detectors (ZCD)

ZCD is wired around general-purpose operational amplifier 741 whose circuit diagram is given at *Fig. 8*. The input and output waveforms of ZCD are shown at *Fig 9*.

## 4. Software for Time Period Measurement

Flow Chart for the time period measurement is shown at *Fig 10*. All though the flow chart is self explanatory, but for sake of clarity, it is explained in brief here.

- The output of ZCD is connected to port pin 1.1 of 8051 microcontroller.

- The counter action is controlled by bit states in the timer mode control resistor (TMOD) and the timer / counter control resistor (TCON). Bit level manipulation of TCON and TMOD is executed by AND and OR operations.
- The counters, i.e., two 8-bit resistors namely Timers High (TH0, TH1) and Timers Low (TL0, TL1) are cleared.
- To enable timer to count, TR0 is set to 1.
- Timer is halted by clearing TR0, when the output of ZCD falls from high to Low.
- The programme checks for the event when output of ZCD falls from high to low. At this moment an internal time is started. The timer continues to increment the counts for the entire period of ZCD output and following high period of ZCD. Once the output of ZCD falls from high to low, the counter is stopped. Therefore, for one cycle duration of power signal, the counter increments its value. Since clock input to counter is of known frequency, therefore, the count value can be converted into time period or the frequency of power signal. This obtained value is compared with the set limiting high and low values of frequency. When the frequency value is low, port pin 1.2 is set while port pin 1.3 is set when frequency is high.

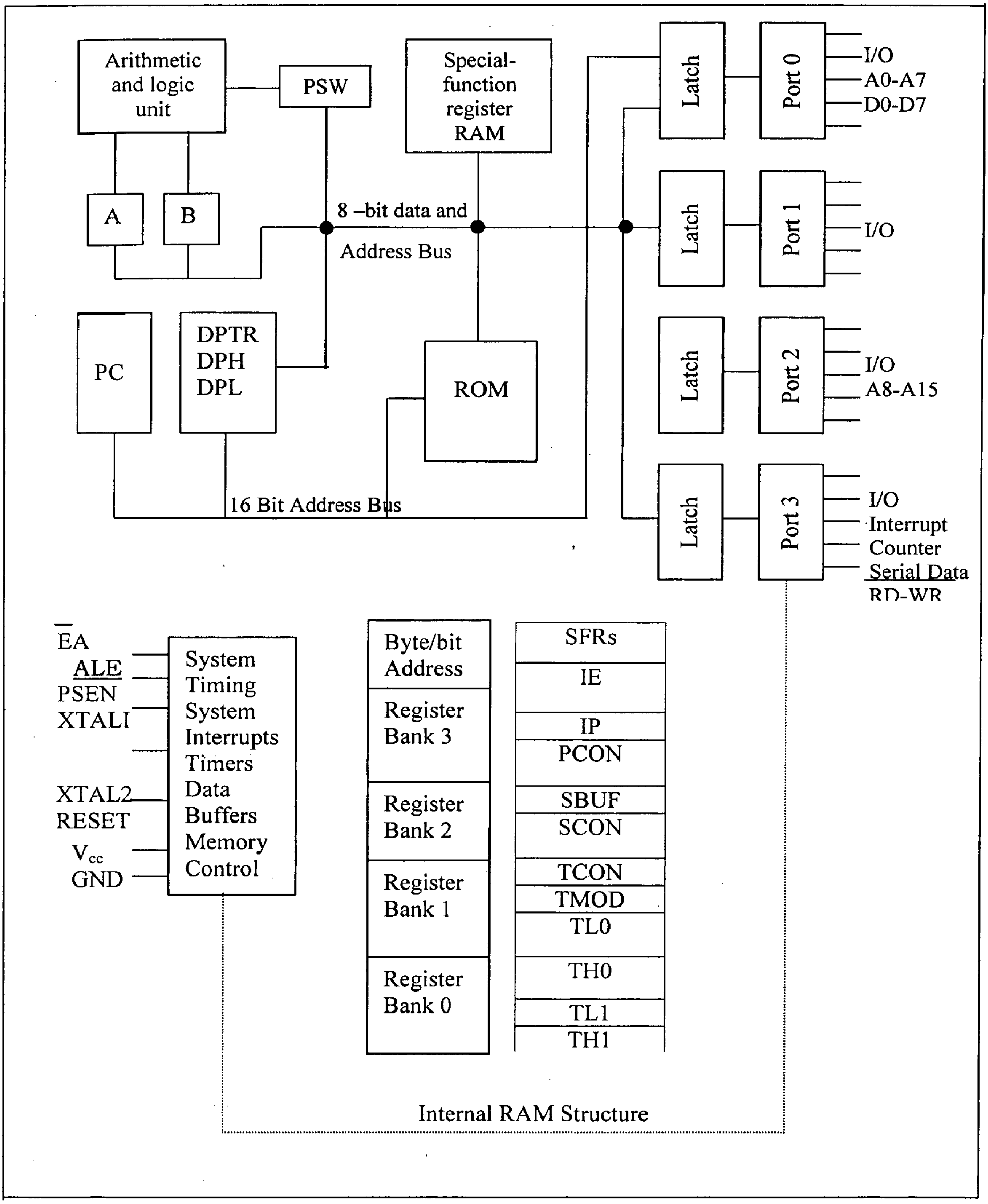



*FIG 7.* **MICRO CONTROLLER 8051 BLOCK DIAGRAM**

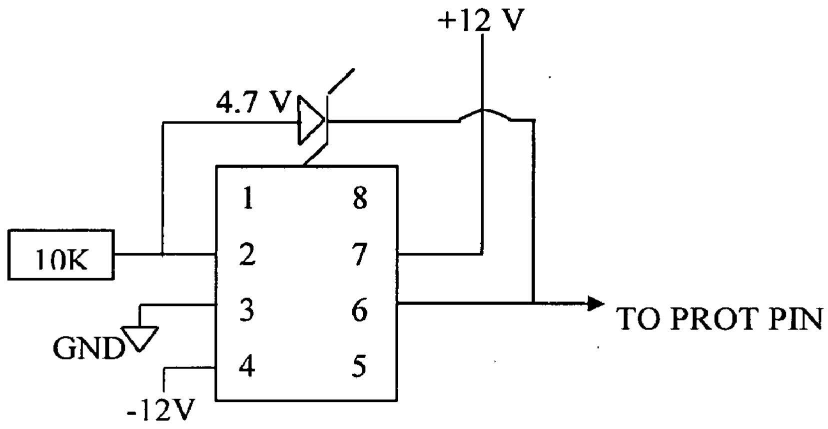



***FIG. 8*** **ZERO CROSSING DETECTOR**

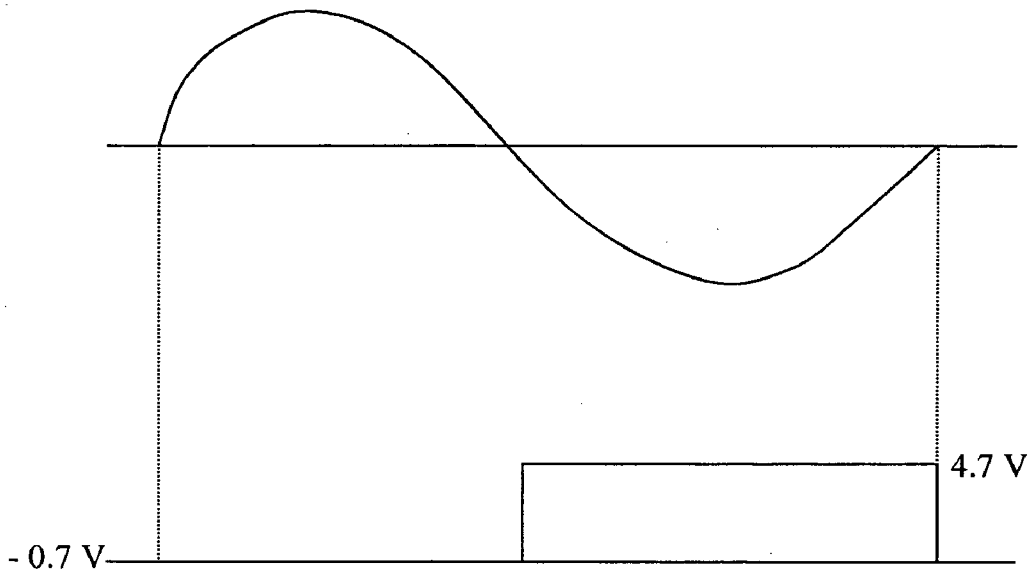
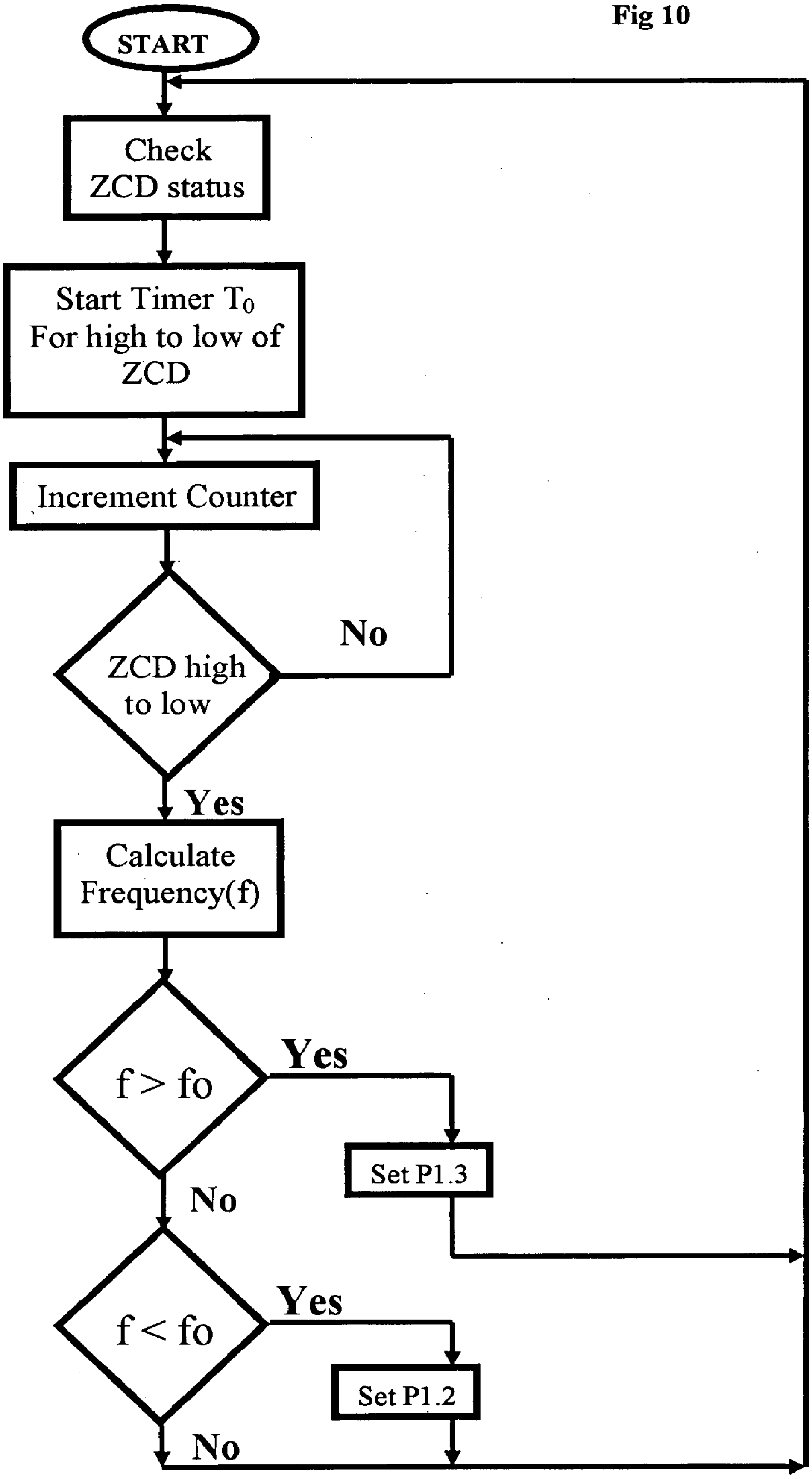



***FIG. 9*** **INPUT OUTPUT WAVE FORMS**

Fig 10


START
Check
ZCD status
Start Timer $T_0$
For high to low of
ZCD
Increment Counter
ZCD high
to low
No
Yes
Calculate
Frequency(f)
f > fo
Yes
Set P1.3
No
f < fo
Yes
Set P1.2
No

# CHAPTER 8

# CONCLUSION

Extreme system conditions can damage equipment in several ways, from melting aluminum conductors concessive currents to breaking turbine blades on generators (frequency excursions). The power system is designed to ensure that if conditions on the grid (excessive or inadequate voltage, apparent impedance or frequency) threaten the safe operation of the transmission lines, transformers, or the power plants, the threatened equipment automatically separates from the network to protect itself from Physical damage. Relays one the devices that effect this protection.

Relay systems are applied with redundancy in primary and backup modes. If one fails, another should defect the fault and trip appropriate circuit breakers. Some backup relays have significant "reach", such that non-faulted line over loads or stable swings may be seen as fault and cause the tripping of a line when it is not advantageous to do so. Proper coordination of the many relay devices in an interconnected system is a significant challenge, requiring continual review and revision. Some relays can prevent synchronizing, making restoration more difficult.

With the ever increase demand for electrical power, "grids" have been established at various level. However the need for grid synchronization and power system protection schemes have also gained currency. With the deregulation and because of various condole economic derivatives, there is a tendency to operate the grid at the brink of instability and hence accurate, faster measurement of system parameters have become more vital. To this end there has been many impetus to the field of power system protection and to that effect many research work has been done/under progress in the field of frequency based protection scheme.

# REFERENCES


1. Sachdev, MS, Baribeu, MA, *"A New Algorithm for Digital Impedance Relay"*, Trams. IEEE, PAS, Vol PAS 98, Nov/Dec. 1979. PP 2232-2240.

2. Sachdev, MS, Giray MM, *"A least error squares Technique for Determining power system Frequency"*. Trans. IEEE, PAS, Vol. PAS 104, No-2 Feb 1985. pp 437-441.

3. Sachdev, M.S.Giray M.M, *"Off-Nominal frequency Measurements in Electric power system."* Trans. IEEE. Power Delivery, Vol 4,No.3, July 1989 pp 1573-1578.

4. Nguyn C.T, Srinivasan K, *"A New Technique For Rapid Tracking of Frequency Deviations Based on level crossings"*. Trans. IEEE, PAS, Vol PAS-103, No 8, Aug 1984, pp 2230-2236.

5. Bergland G.D, *"A Guided Tour for The Fast Fourier Transform in Digital Signal Processing"*. IEEE. Press, 1972,pp 228-239.

6. Girgis AA., Fredric HM, *"A New FFT-Based Digital Frequency Relay for Load Shedding"*. Trans. IEEE. PAS, Vol PAS-101, no.2, Feb 1998, pp115-118.

7. Sidhu T.S, Sachdev M.S, *"An Iterative DSP Technique For Tracking Power System Frequency And Voltage Phasors"* Trans.IEEE, Power Delivery.Vol 13,Jan,1998,pp 115-118.

8. Houshang K, Masourd KG, Iravani M.R *"Estimation of Frequency and its Rate of change For Application in Power System"*. Trans. IEEE. Power Delivery Vol.19. No 2. Apr 2004.pp 472-480.

9. Chottopadhyay B, Ivanov C, Schdev M.S , *"A New Frequency Measurement Algorithm in the Presence of Harmonic Using Instantaneous phasor quantities"* , Electrical and computer engineering , 1996, Canadian conference on , Vol 1, 26,29 May 1996. pp 130-133.

10. Lee W.J, GU JC, *" A Microcomputer Based Intelligent Load Shedding Relay"*. Trans. IEEE. Power Delivery . Vol 4, No 4 Oct 1989. pp 2018-2004.

11. A. V Oppenheim, and R. W. Schafer, "*Discrete-Time signal processing*", Prentice-Hall, Englewood Cliffs, New Jersey, USA, 1989.

12. P. Denys, C. Counan, L. Hossenlopp, C. Holweck, "*Measurement of Voltage phase for the French future Defence plan Against Losses of Synchronism*", IEEE Trans. on Power Delivery, Vol 7, No 1, Jan,1992, pp. 62-69.

13. M.M. Begovic, P.M. Djuric, S. Dunlap, A. G. Phadke, "*Frequency Tracking in power Networks in the presence of Harmonics*", IEEE Trans. on Power Delivery, Vol 8, No 2, Apr, 1993, pp 480-486.

14. A.G Phadke, et al, "*Synchronized Sampling and Phasor Measurements for Relaying and Control*", IEEE Trans. on Power Delivery, Vol 9, No 1, January, 1994, pp 442-452.

**APPENDIX**

## TEST OF COMPLEX DEMODULATION

```
clc;
fs=input('ENTER THE SAMPLING FREQUENCY   ');
% fs=1000; % sampling frequecny
dt=1/fs;
Sigma=input('ENTER Sigma( Standard deviation for noise) ');
%Sigma=1e-2; % standard deviation for noise
f0=input('ENTER THE FUNDAMENTAL FREQUENCY   ');
% f0=50; %50Hz
windowtime=input('ENTER THE TIME FOR WHICH SAMPLES ARE TO BE TAKEN ');
V_rms=1.0;
V1=sqrt(2)*V_rms;
Phi1=0.00;
V2=1.0*V1; Phi2=Phi1-2*pi/3;
V3=1.0*V1; Phi3=Phi1+2*pi/3;
SNR=20*log10(V_rms/Sigma);
t=[0:dt:windowtime];
ns=windowtime*fs;
f1=(f0)*ones(size(t))+1*sin(2*pi*1*t)+0.5*sin(2*pi*6*t); % TRUE FREQUENCY
% ---------PLOT OF TRUE FREQUENCY-----------------------------------------------
% plot(t,f1,'g');
% hold on

w1=2*pi*f1;
 % Angle for fundamental phasor quantities-------------------
 Teta1=zeros(size(t));
 Teta2=zeros(size(t));
 Teta3=zeros(size(t));
 Teta1(1)=Phi1;
 Teta2(1)=Phi2;
 Teta3(1)=Phi3;
  for k=2:length(t)
     Teta1(k)=rem(Teta1(k-1)+w1(k)*dt,pi*2);
     Teta2(k)=rem(Teta2(k-1)+w1(k)*dt,pi*2);
     Teta3(k)=rem(Teta3(k-1)+w1(k)*dt,pi*2);
  end
 % ----Angle for 3rd harmonic-----
 Gamma1=zeros(size(t));
 Gamma2=zeros(size(t));
 Gamma3=zeros(size(t));
 Gamma1(1)=0;
 Gamma2(1)=-10*pi/180;
 Gamma3(1)=10*pi/180;
```

```
for k=2:length(t);
   Gamma1(k)=rem(Gamma1(k-1)+3*w1(k)*dt,2*pi);
   Gamma2(k)=rem(Gamma2(k-1)+3*w1(k)*dt,2*pi);
   Gamma3(k)=rem(Gamma3(k-1)+3*w1(k)*dt,2*pi);
end
%------angle for 5th harmonic----------------
Epsilon1=zeros(size(t));
Epsilon2=zeros(size(t));
Epsilon3=zeros(size(t));
Epsilon1(1)=0;
Epsilon2(1)=-125*pi/180;
Epsilon3(1)=115*pi/180;
 for k=2:length(t);
    Epsilon1(k)=rem(Epsilon1(k-1)-5*w1(k)*dt,2*pi);
    Epsilon2(k)=rem(Epsilon2(k-1)-5*w1(k)*dt,2*pi);
    Epsilon3(k)=rem(Epsilon3(k-1)-5*w1(k)*dt,2*pi);
 end
    % --------------------------------------
 Harm1=zeros(size(t));
 Harm2=zeros(size(t));
 Harm3=zeros(size(t));
 Harm1=0.5*V1*sin(Gamma1)+0.02*V1*sin(Epsilon1);
 Harm2=0.5*V2*sin(Gamma2)+0.02*V2*sin(Epsilon2);
 Harm3=0.5*V3*sin(Gamma3)+0.02*V3*sin(Epsilon3);
  %-------------ADDITION OF DC COMPONENT( DECAYING)------------------
  ADC=10.5;
  D1=0.5*exp(-ADC*t);
  D2=0.5*exp(-ADC*t)*sin(2*pi/3);
  D3=0.5*exp(-ADC*t)*sin(-2*pi/3);

      % phasors with noise
  v1=V1*sin(Teta1)+Sigma*randn(size(t))+Harm1+0*D1;
  v2=V2*sin(Teta2)+Sigma*randn(size(t))+Harm2+0*D2;
  v3=V3*sin(Teta3)+Sigma*randn(size(t))+Harm2+0*D3;
 tic;
  % alpha Beta components
  Alpha=sqrt(2/3)*(v1-0.5*v2-.5*v3);
  Beta=sqrt(1/2)*(v2-v3);

  % Complex input signal
  V=Alpha+ j*Beta;

  % modulation signal
  Z=cos(-2*pi*f0*t)+j*sin(-2*pi*f0*t);

  % demodulated signal
```

```
Y=V.*Z;
Im_Y=imag(Y); Re_Y=real(Y);
Amp_Y=sqrt(Re_Y.*Re_Y+Im_Y.*Im_Y);
Pha_Y=atan2(Im_Y,Re_Y);

% create the signal U
NN=length(Y);
Re_U=[0 Re_Y(2:NN).*Re_Y(1:NN-1) + Im_Y(2:NN).*Im_Y(1:NN-1)];
Im_U=[0 Im_Y(2:NN).*Re_Y(1:NN-1) - Re_Y(2:NN).*Im_Y(1:NN-1)];

Arg_U=atan2(Im_U,Re_U);
f_hat=f0+fs*Arg_U./(2*pi);  % estimated UNFILTERED frequency
time=toc
plot(t,f_hat,'g')
xlabel('time in s');
ylabel('frequency in Hz');
title('Actual Frequency Vs  Unfiltered Frequency estimate when noise is low');
figure
  %-------------filter estimate----------------
N=3; % filter order
fc=20; % (Hz) cutoff frequency
fn=fs/2;
 % specification in normalised frequency
 Wn=fc/fn;

 % design LP Butterwoth filter
 [B,A]=butter(N,Wn);
 % filter the estimate
 f_hat_filt_ex3=filter(B,A,f_hat-f0)+f0;   % ESTIMATED FILTERED FREQUENCY
USING CAUSAL FILTER
 f_hat_filt_ex4=filtfilt(B,A,f_hat-f0)+f0; % USING NON-CAUSAL

for i=1:ns
plot(t(i),f_hat_filt_ex3(i),'r')
hold on;
 plot(t,f1,'g');
xlabel('time in s');
ylabel('frequency in Hz');
legend('estimate','true');
title('Actual Frequency Vs  Filtered Frequency Estimate Using Causalfilter');
end
figure
for i=1:ns
plot(t(i),f_hat(i),'r');
hold on;
end
```

```
plot(t,f1,'g');
 xlabel('time in s');
 ylabel('frequency in Hz');
 legend('Estimated');
 title('Unfilterd Response')
ferr=f1-f_hat_filt_ex3(i);

 figure
 plot(t,ferr,'b');
  xlabel('time in s');
 ylabel('frequency in Hz');
 title('Estimation Error');
 figure
 plot(t,f_hat_filt_ex4,'r');
 hold on
 plot(t,f1,'g');
  xlabel('time in s');
 ylabel('frequency in Hz');
 legend('estimate','true');
 title('Actual Frequency Vs  Filtered Frequency Estimate Using Noncausalfilter');
     end
```