MATRIX PENCIL METHOD AND FFT BASED EMBEDDED CONDITION MONITORING OF ROTATING MACHINE

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Abstract

Rotating machines are an integral part of large electrical power machinery in most of the industries. Any degradation or outages in the rotating electric machinery can result in significant losses in productivity. It is critical to monitor the equipment for any degradation's so that it can serve as an early warning for adequate maintenance activities and repair. Prior research and field studies have indicated that the rotating machines have a particular type of signal structure during the initial start-up transient. This signal structure can be modelled by an exponentially damped sinusoidal signal during initial start-up. The major focus of this project is to take advantage of this a prior knowledge of signal structure to study the effect of degradation in two main signal parameters: frequency components of transient signal and the associated exponential damping factor.

In this work, a data acquisition system is first used to sample the transient signal. This signal is analysed on an embedded TMS320 based DSP core platform. Both FFT and MPM estimators have been implemented to study various signals captured by the data-acquisition system. For measuring the frequency of operation, both FFT and MPM were employed whereas the damping coefficient is calculated using MPM method. Initial implementation and validation was conducted in MATLAB and Simulink. Once validated, the code was ported on to TMS320 platform. The results show good agreement with expected results for both simulated and real-time data. The real-time data from AC water pumps which have rotating motors built-in were collected and analysed.

It is found that the frequency estimation performance of the MPM matches that of the FFT for synthetic and real time data. The evaluation of the condition of machines based on the damping coefficient using MPM matched with the actual condition of the machine.

Keywords: FFT, Matrix Pencil Method (MPM), Rotating machine, Frequency, Damping, TMS320DM624

Nomenclature

Cycle	Cycle, s
Data	bit, b
Date rate	bits per second, bps
Kbits	Kilo bits, Kb=2^{20}=1024b
Mbits	Mega bits, Mb= 2^{20} = 1,048,576b
f	Frequency, Hz

Abbreviations

ARM	Advanced RISC Machine
AGWN	Additive Gaussian White Noise
BER	Bit Error Rate
CC	Code Coverage
DFT	Discrete Fourier Transform
DOA	Direction of Arrival
DRAM	Dynamic Random Access Memory
FT	Fourier Transform
FFT	Fast Fourier Transform
IC	Instruction Count
IPS	Instructions Per Second
ISI	Inter Symbol Interference
IFFT	Inverse Fast Fourier Transform
JTAG	Joint Test Action Group
LCD	Liquid Crystal Display
MIPS	Million Instructions Per Second
MSE	Mean Square Error
RISC	Reduced Instruction Set Computer
SNR	Signal to Noise Ratio
SVD	Singular Value Decomposition
ZP	Zero Padding

1. INTRODUCTION

Rotating machines cover a wide range of critical facilities and are the backbone of numerous industries, from gas turbines used in the production of electricity to turbo-machinery utilized to generate power in the aerospace industry. It is vital that these machines run safely over time and under different operational conditions, to ensure continuous productivity and prevent any catastrophic failure, which would lead to extremely expensive repairs and may also endanger the lives of the operating personnel. Condition monitoring is the process of monitoring a parameter of condition in machinery, such that a major change is indicative of developing failure. It is a main component of predictive maintenance. The use of conditional monitoring allows maintenance to be scheduled, or other actions to be taken to avoid the consequences of failure, before the failure occurs.

The basis of any condition monitoring depends on understanding the electric, magnetic and mechanical behavior of a rotating machine in both the healthy and faulty state. Several parameters may be used for condition monitoring, including corrupted signals such as vibration, electrical signals, and noise [1].

Whenever a machine is turned ON due to vibration transient signals in the machine may occur. Vibration measurement provides a very efficient way of monitoring the dynamic conditions of a machine such as unbalance, misalignment, mechanical looseness, structural resonance, soft foundation and shaft bow.

Noise, the term for unwanted sound, is closely related to the occurrence of structural vibrations. The demand for low noise machinery has become an increasingly important issue in our society [2].

Whenever a rotating machine is turned on a direct step signal is not achieved, a transient signals are obtained. And these signals in many areas can be modeled as a super-position of several damped sinusoids. Hence rotating machines generates a signals composed of sinusoids equi-spaced in frequency. Some examples are power line surges when appliances/loads are turned ON or OFF, DOA- RF/Antennas and mechanical vibration transients during start up (motors, generators etc.). The spectrum of such signals is useful in detecting faults. Traditional Fourier based spectrum estimation methods are not sufficient due to the non-stationary of measured signals. Hence Matrix pencil method is used for estimating exponential damped sinusoid.

Spectrum analysis is the process of determining the frequency domain representation of a time domain signal and most commonly employs the Fourier transform. A spectrum analyzer can measure the noise, frequency of response, signal-to-noise ratio, and distortion inherent. Spectrum analyzers can be used in almost any modern signaling process. Spectrum analyzers are often used in various industries to measure sound levels [3].

Traditional signal analysis relies extensively on spectral analysis. Many things oscillate in our universe. For example, speech is a result of vibration of the human vocal cords; stars and planets change their brightness as they rotate on their axes and revolve around each other; ship's propellers generate periodic displacement of the water, and so on. The shape of the time domain waveform is not important in these signals; the key information is in the frequency, phase and amplitude of the component sinusoids, therefore spectral analysis using Fourier transforms is used to extract information related to health condition of machines.

Fast Fourier Transform (FFT) which is a special case of the generalized Discrete Fourier Transform and converts the signal from its time domain representation to its equivalent frequency domain representation. However, frequency analysis (sometimes called Spectral Analysis) is only one aspect of interpreting the information contained in a vibration signal. Frequency analysis tends to be most useful on machines that employ rolling element bearings and whose main failure modes tend to be the degradation of those bearings, which typically exhibit an increase in characteristic frequencies associated with the bearing geometries and constructions.

The Matrix Pencil Method allows a fast, accurate detection and estimation of transient signals caused by rotating machine. Its efficiency in many areas such as speech pattern recognition, aircraft, signal processing etc. Moreover, transient signals it shows better performance than many improved versions of the Prony's method such us Tufts and Kumaresan polynomial one.

2. MATHEMATICAL ANALYSIS OF MATRIX PENCIL METHOD

$$y(t) = x(t) + n(t) \approx \sum_{i=1}^{M} R_i e^{(s_i t)} + n(t); 0 \le t \le T \quad \ldots\ldots\ldots\ldots (1)$$

Where

y (t) = observed time response

n (t) = noise in the system

x (t) = signal

R_i= residues or complex amplitudes

$S_i = -\alpha_i + j\omega_i$

ω_i= angular frequencies where $\omega_i = 2\pi f_i$

After sampling, the time variable t, is replaced by kT_s, where T_s is the sampling period. The sequence can be rewritten as,

$$y(kT_s) = x(kT_s) + n(kT_s) \approx \sum_{i=1}^{M} R_i z_i^{\,k} + n(kT_s); \quad for\ k = 0, \ldots\ldots, N-1 \quad \ldots (2)$$

And

$$z_i = e^{s_i T_s} = e^{(-\alpha_i + j\omega_i)T_s} \quad \text{For i=1, 2, M} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(3)$$

In general, simultaneous estimation of M, R_is, and z_is is a nonlinear problem. In many cases, linear problem is equivalent to solving the nonlinear problem. In addition, the solution to the linear problem can be used as an initial step to the non-linear-optimization problems [4].

Two of the popular linear methods are the “polynomial” method and the “matrix pencil” method. The basic difference between the two is that the “polynomial” method is a two-step process in finding the poles, z_i. One needs to solve a matrix equation for the co- efficient of a polynomial, whose roots provide z_i.

On the other hand, the “matrix pencil” approach is a one-step process. The poles z_i are found as the solution of a generalized Eigen value problem. Hence, there is no practical limitation on the number of poles, M, that can be obtained by this method. In contrast, for a polynomial method it is difficult to find roots of a polynomial for example when M is greater than 50. The Matrix Pencil approach is not only more computationally efficient, but it also has better statistical properties for the estimates of $z_i$than the “polynomial” method

$$f(t,\lambda) = g(t) + \lambda\, h(t) \qquad \text{.............. (4)}$$

F (t, λ) is called a pencil of functions g (t) and h (t), parameterized by λ. To avoid insignificance, g (t) is not permitted to be a scalar multiple of h (t). The pencil of functions contains very important features about extracting information about z_i, given y (t), when g (t), h (t), and λ are approximately selected.

Case (1): the noiseless case,

For noiseless data, we can define two (N - L) x L matrices, Y_1 and Y_2, defined by

$$[Y_2] = \begin{bmatrix} x(1) & x(2) & \cdots & x(L) \\ x(2) & x(3) & \cdots & x(L+1) \\ \vdots & \vdots & \vdots & \vdots \\ x(N-L-1) & x(N-L) & \cdots & x(N-1) \end{bmatrix}_{(N-L)XL} \qquad \text{.. (5)}$$

$$[Y_1] = \begin{bmatrix} x(0) & x(1) & \cdots & x(L-1) \\ x(1) & x(2) & \cdots & x(L) \\ \vdots & \vdots & \vdots & \vdots \\ x(N-L-1) & x(N-L) & \cdots & x(N-2) \end{bmatrix}_{(N-L)XL} \qquad \text{.. (6)}$$

Where, L is pencil parameter. The pencil parameter, L, is very useful in eliminating some effects of noise in the data.

We can write Y_1 and Y_2 as,

$$[Y_2] = [Z_1][R][Z_0][Z_2],$$

$$[Y_1] = [Z_1][R][Z_2], \qquad \text{..................... (7)}$$

Where,

$$[Z_1] = \begin{bmatrix} 1 & 1 & \cdots & 1 \\ Z_1 & Z_2 & \cdots & Z_M \\ \vdots & \vdots & \vdots & \vdots \\ Z_1^{(N-L-1)} & Z_2^{(N-L-1)} & \cdots & Z_M^{(N-L-1)} \end{bmatrix}_{(N-L)XM} \qquad \text{.. (8)}$$

$$[Z_2] = \begin{bmatrix} 1 & Z_1 & \cdots & Z_1^{(L-1)} \\ 1 & Z_2 & \cdots & Z_2^{(L-1)} \\ \vdots & \vdots & \vdots & \vdots \\ \vdots & \vdots & \cdots & \vdots \\ 1 & Z_M & \cdots & Z_M^{(L-1)} \end{bmatrix}_{MXL} \qquad \text{................. (9)}$$

$$[Z_0] = diag\,[Z_1, Z_2, \dots \dots \dots Z_M] \qquad \text{................ (10)}$$

$$[R] = diag\,[R_1, R_2, \dots \dots \dots R_M] \qquad \text{................ (11)}$$

Where diag [•] represents an M x M diagonal matrix.

Now consider the matrix pencil,

$$[Y_2] - \lambda[Y_1] = [Z_1][R]\{[Z_0] - \lambda[I]\}[Z_2] \qquad \text{,... (12)}$$

Where [I] is the M x M identity matrix, in general, the order of $\{[Y_2] - \lambda[Y_1]\}$ will be M, provided that M ≤L≤ N-M.

In case, if λ $= Z_i$, i=1, 2...... M, the i^th row of $\{[Z_0] - \lambda[I]\}$ is zero, and the order of this matrix is M -1. Hence, the parameters z may be found as the generalized Eigen values of the matrix pair {[Y_2]; [Y_1]}. Similarly, the problem of solving for Z_i, can be ordinary Eigen value problem,

$$\{[Y_1]^{+}[Y_2]\text{-}\lambda\,[I]\} \qquad \text{.............................. (13)}$$

Where $[Y]^{+}$ is the Moore-Penrose pseudoinverse of [Y_1]. This, in turn, is defined as

$$\{[Y_1]^{+} = \{[Y_1]^{H}[Y_1]\}^{-1}[Y_1]^{H} \qquad \text{......... (14)}$$

Where the superscript “H” denotes the conjugate transpose.

Case (2): the noise case

In the presence of noise, pre-filtering required. To remove noise, here one form of the data matrix [Y] from the noise-contaminated data y (t) by combining [Y_1] and [Y_2] as

$$[Y] = \begin{bmatrix} y(0) & y(1) & \cdots & y(L) \\ y(1) & y(2) & \cdots & y(L+1) \\ \vdots & \vdots & \vdots & \vdots \\ y(N-L-1) & y(N-L) & \cdots & y(N-1) \end{bmatrix}_{(N-L)X(L+1)}$$

$$\text{... (15)}$$

[Y_1] is obtained from [Y] by deleting the last column, and [Y_2] is obtained from [Y] by deleting the first column. So, in Equations (5)-(6), the x (k) is replaced by y (k) to obtain [Y_1] and [Y_2]. For efficient noise filtering, the parameter L is chosen between N/3 to N/2. For these values of L, the variance in the parameters Z_i, due to noise, has been found to be minimum. All the N data samples are utilized, even though L may be considerably less than N.

Singular-value decomposition (SVD) of the matrix [Y] is carried out as

$$[Y] = [U]\,[\Sigma]\,[V]^{H} \qquad \text{.......................... (16)}$$

Here, [U] and [V] are unit matrices, composed of the Eigen vector of $[Y][Y]^{H}$ and $[Y]^{H}[Y]$, respectively, and matrix containing the singular values of [Y], i.e.

$$[U]^{H}[Y][V] = [\Sigma] \qquad \text{....................... (17)}$$

Choosing the parameter of M is done at this stage. Typically, the singular values beyond M are set equal to zero. The way M is chosen is as follows. Consider the singular value σ_c such that

$$\frac{\sigma_c}{\sigma_{max}} \approx 10^{-p} \qquad \text{............................. (18)}$$

Where, p is number of significant decimal digit of the data. For example, if the data is accurate up 1 to 3 significant digits, then the singular values for which the ratio is below 10^{-3} is essentially noise singular values, and they should not be used in the reconstruction of the data. Where $[V_1']$ is obtained from $[V']$ with the last row of $[V']$ deleted; $[V_2']$ is obtained by removing the first row of $[V_1']$ and $[\Sigma']$ is obtained from the M columns of $[\Sigma]$ corresponding to the M dominant singular values.

The "filtered" matrix, [V'], constructed so that it contains only M dominant right-singular vectors of [V]:

$[V'] = [V_1, V_2, \dots \dots \dots V_M]$

The right-singular vectors from M +1 to L, corresponding to the small singular values, are discarded. Therefore,

$$[Y_1] = [U]\,[\Sigma'][V_1']^H \quad \dots\dots\dots\dots\dots (19)$$

$$[Y_2] = [U]\,[\Sigma'][V_2']^H \quad \dots\dots\dots\dots\dots (20)$$

It can be shown that, for the noiseless case, the Eigen values of the following matrix

$$\{[Y_2] - \lambda[Y_1]\}_{LXM} = \{[Y_1]^+[Y_2] - \lambda[I]\}_{MXM} \quad \dots\dots\dots\dots\dots (21)$$

Are similar to the Eigen values of the following matrix

$$\{[V_2']^H - \lambda[V_1']^H\} = \{[V_1']^H\}^+\{[V_2']^H\}^+ - \lambda[I] \quad \dots\dots\dots\dots\dots (22)$$

This methodology of solving for Z_i provides minimum variance in the estimate of Z_i in the presence of noise. Typically, up to 20-25 dB of signal-to-noise ratio (SNR) can be handled effectively by this technique. Also, the poles Z_i are much more simply than in a Prony type method [4, 5].

3. MATHEMATICAL ANALYSIS OF FFT

Discrete Fourier Transform: A DFT is a Fourier that converts discrete data from a time wave into a frequency spectrum. However, calculating a DFT is sometimes too slow, because of the number of multiplies required.

$$F(n) = \sum_{k=0}^{N-1} x(k)e^{-jk2\pi\frac{n}{N}} \,(or)\, \sum_{k=0}^{N-1} x(k)W_N^{kn} \quad \dots\dots (1)$$

n=0................N-1 .... (23)

$$W_N^{kn} = e^{-j\frac{2\pi}{N}}$$

F (n) is the amplitude at the frequency n and N is the number of discrete samples.

Fast Fourier Transform: The Fast Fourier Transform does not refer to a new or different type of Fourier transform. It refers to a very efficient algorithm for computing the DFT. An FFT is an algorithm that speeds up the calculation of a DFT. The time taken to evaluate a DFT on a computer depends mainly on the number of multiplications involved. DFT needs N^2 multiplications. FFT only needs Nlog2 (N).The entire purpose of an FFT is to speed up the calculations [6].

Advantages of Fourier Transforms:

- Fourier Transform is the one of the most significant transformation since it ties together two of the most used phenomena's i.e. Time and frequency
- If a signal or waveform is in time domain and by using Fourier transform the frequencies of the signal can be determined. And observing the signal in the frequency domain, lot of manipulation can be performed on the signal including filtering, sampling, modulation etc.
- The Fourier transform is best when dealing with boundary-value problems
- Fourier transform describes how a system, responds to pure sinusoidal signals

4. DESIGN AND IMPLEMENTATION

Literature review on various DSP algorithms such as FFT, Matrix Pencil Method for monitoring rotating machine has been carried out using books, journal papers, conference papers and other manuals. The principles and techniques of FFT and Matrix Pencil approach have been studied. Design specifications of the FFT and Matrix Pencil approach has arrived based on the literature review and system requirements.

Development of MATLAB code for DSP algorithms based on FFT and Matrix Pencil Method has carried out for condition monitoring of rotating machine. Analysis of exponentially decaying sinusoids using FFT and Matrix Pencil Method has been performed. Development of C code for DSP algorithms based on FFT and Matrix Pencil Methods on a DSP platform was carried out. Verification of the algorithms through real field data captured by data acquisition systems for monitoring the health of rotating machines has been performed. Performance of each of the techniques, in terms bias and variance, in the presence of noise was analyzed and compared. Study of the performance of the developed techniques under AWGN with various SNR's was carried out [7].

To have a better implementation of the Matrix Pencil Method (MPM) and FFT a DSP processor is more suitable then the conventional processor. As in this embedded system an extensive MPM and FFT need to be computed based on the rotating speed of the electric machine. In this system DM642 EVM is a low-cost standalone development platform that enables users to evaluate and develop applications for the TI TMS320DM64x DSP family.

The system is simulated in MATLAB and implemented on the DM642.The heart of the DM642 is the C64x CPU which includes special instructions to accelerate the performance. Also, the RISC-like instruction set and extensive use of pipelining in C64x, allow many instructions to be scheduled and executed in parallel. Parallelism is the key to extremely high performance needed for video and imaging applications. A high performance two-level cache design allows the CPU to operate at the maximum rate. The two-level cache lowers development time by automating off-chip to on-chip data transfers. A high performance EDMA controller feeds the CPU through flexible high-bandwidth bus architecture. TMS320DM642 DSP is operating at the 225MHz with a 16Mbytes of synchronous DRAM and 512 Kbytes of non-volatile flash memory.

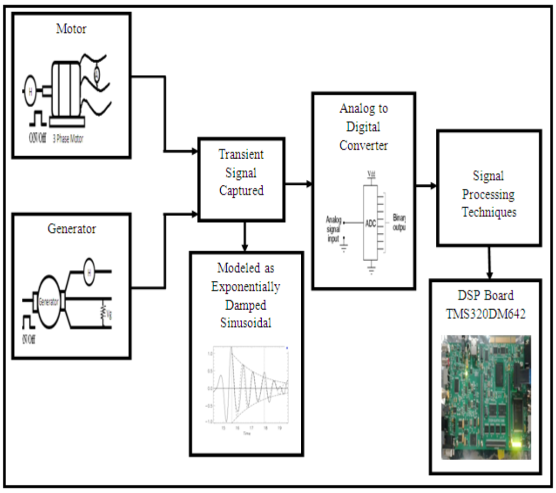

Fig. 1 Block Diagram for the Condition Monitoring of the Rotating Machine

Figure 2 illustrates the completed flow of matrix pencil method. Initially the signals captured from the machine and this signals are taken as an input and are added, signals are sampled and Additive White Gaussian Noise (AWGN) is added and then the values of K, L, N and M are set, K is number of real damped sine components, N is number of samples in the added signal and L is linear prediction order the value of L is chosen between N/3 to N/2. Once when the values are set the matrix [Y] is resized to (N-L) x (L+1) from this matrix [Y1] and [Y2] matrices are created. [Y1] matrix is created by deleting the last column of [Y] matrix whereas, [Y2] is created by deleting first column of [Y] and theses matrix are resized to (N-L) x L and then singular value decomposition (SVD) of [Y2] matrix is taken than only the diagonal elements of S is extracted. U, S and V are resized and are multiplied with [Y1] and the Eigen values is determined and natural logarithm is applied to the Eigen values and the values are rearranged in descending order, real and imaginary terms are separated. Real term is multiplied with sampling frequency this gives damping of the signals, whereas the imaginary terms term is also multiplied with sampling frequency this gives the frequency of the signal.

In analog or digital communication, **signal to noise ratio (SNR)** is defined as is a measure of signal strength relative to background noise; it is measured in decibels (dB).The **mean square error**(**MSE**) is defined as the average of the square of the difference between the desired response and the actual system output (the error).

If $\bar{y}$ is a vector of n predictions, and y is the vector of the true values, then the MSE of the predictor is:

$$MSE=\frac{1}{n}\sum_{i=1}^{n}(\bar{y}-y)^2 \text{ .... (24)}$$

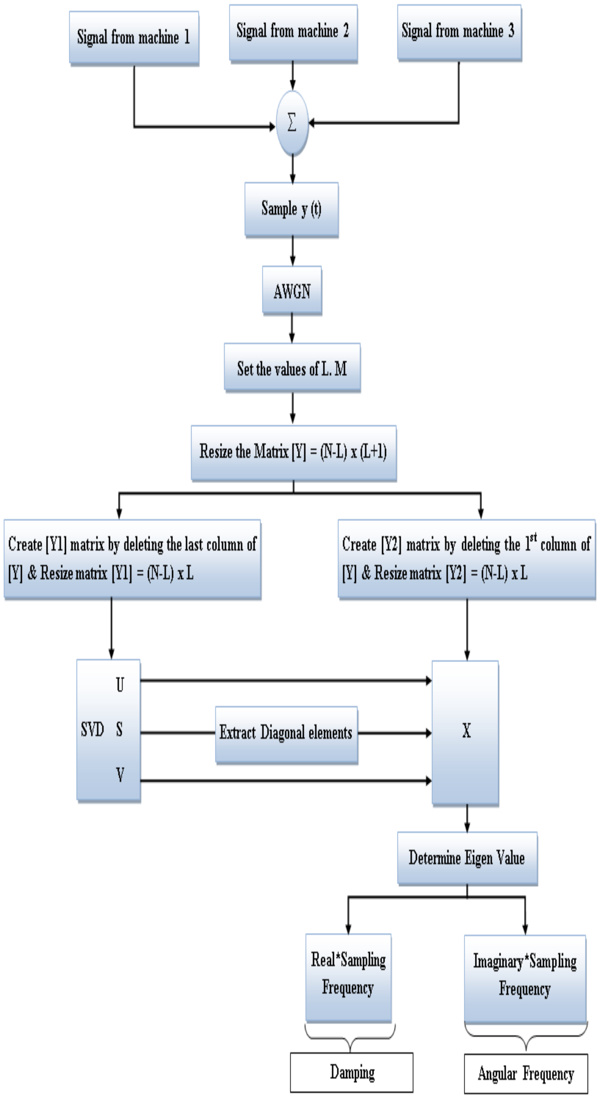

Fig. 2 Flow of Implementation of Matrix Pencil method

For various values of SNR frequencies and damping is calculated. From the definition of MSE, the original frequency is subtracted by the obtained frequency and is squared and then average is taken for each values frequency and damping, and a graph is plotted against MSE vs. SNR.

Digital signal processing (DSP) involves the manipulation of digital signals in order to extract useful information from them. Although an increasing amount of signal processing is being done in digital domain, there remains the need for interfacing to the analog world. The TMS320DM642 DSP processor family has been introduced by Texas Instruments to meet high performance demands in analog signal processing applications.

The Developed MPM is ported on the target board i.e. TMS320DM642. Figure 3 shows TMS320DM642 hardware block diagram. Here JTAG emulator is used to convert high level language of code compressor studio to machine language and then the code is dumped on the TMS320 board and LCD display is connected to the board through which the output of MPM is observed.

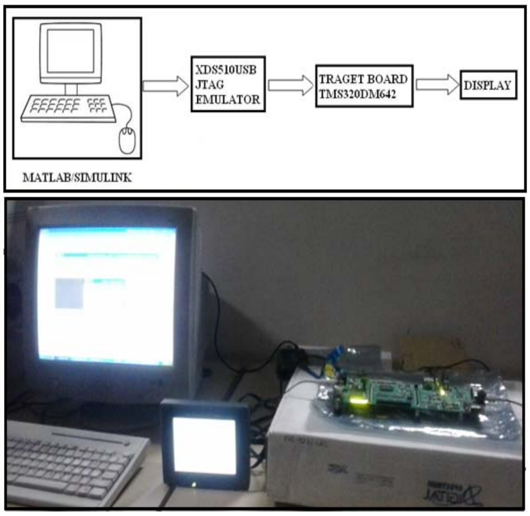

Fig. 3 Hardware Prototype of the MPM and FFT Implementation for the Condition Monitoring of the Rotating Machine

5. RESULTS AND DISCUSSION

In this section the various results of signals obtained from different machine had been analysed. The stage-wise signals are obtained from different machine had been discussed. Figure 4 shows the different signals which were measured from four different machines (three healthy water pump motors and one which is degraded water pump) but all the four machines had same specification rating and these signals are used to determine the machine conditions. (Using MPM and FFT). FFT which is developed in MATLAB is simulated with synthetic data i.e. with three sinusoidal signals. The synthetic data is developed in order to verify the developed FFT algorithm.

Figure 5 and Table 1 show output of FFT obtained in MATLAB for synthetic data. The synthetic data of FFT contains both frequency and damping of the signals. But from FFT only frequency of the signals is estimated through FFT.MPM which is developed in MATLAB is simulated with synthetic data i.e. with three sinusoidal signals. The reason for developing the synthetic data was to verify the developed MPM algorithm.

Figure 6 and Table 2 show output of MPM obtained in MATLAB for synthetic data. The synthetic data of MPM contains both frequency and damping of the signals. MPM determines both frequency and damping of the given signals. FFT which is developed in MATLAB is simulated with real field data, which is captured from different machines. Figure 7 and Table 3 shows an FFT output which is obtained for real time data.

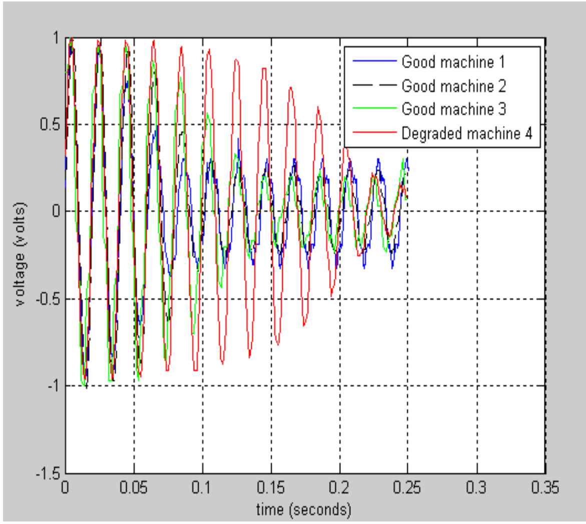

Fig. 4 Signal Obtained from Different Machine

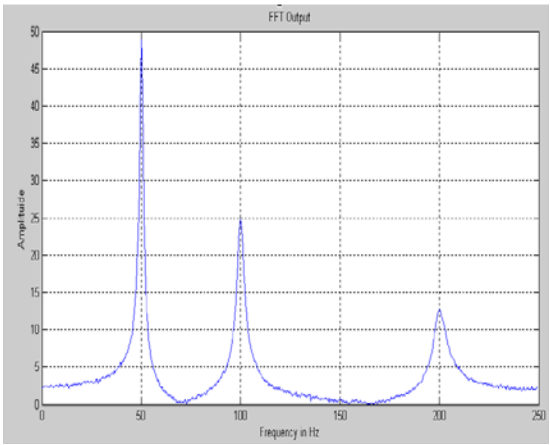

Fig. 5 Output of FFT Obtained in MATLAB for Synthetic Data

Table 1. Frequency Obtained for Synthetic Data

Signals	Frequency (Hz)
Signal 1	50
Signal 2	100
Signal 3	200

From the above plot it is clear that the frequency of all the four machines is of 50Hz but from FFT one cannot decide whether the machine is degraded or not, So MPM is used in order to estimate the machine condition.

Since the damping of the degraded machine is low, it has more amplitude for a longer duration of time and the transient period is also more because of this reason the degraded machine has high magnitude when compare to good machine. The peak magnitude from the spectrum can be related to the damping co-efficient of the machine but this is not an adequate condition to confirm the condition of the machine henceforth MPM is used.

MPM which is developed in MATLAB is simulated with real field data, the data is captured from different machines.

```
frequency_Signal_1 =   200.0003

frequency_Signal_2 =   100.0046

frequency_Signal_3 =   49.9996

Damping_Signal_1 =  19.8760

Damping_Signal_2 =  10.0742

Damping_Signal_3 =   4.9949
```

Fig. 6 Output of MPM obtained in MATLAB for Synthetic Data

Table 2. Frequency and Damping Obtained from Synthetic Data

Signals	Frequency(Hz)	Damping
Signal 1	200	20
Signal 2	100	10
Signal 3	50	5

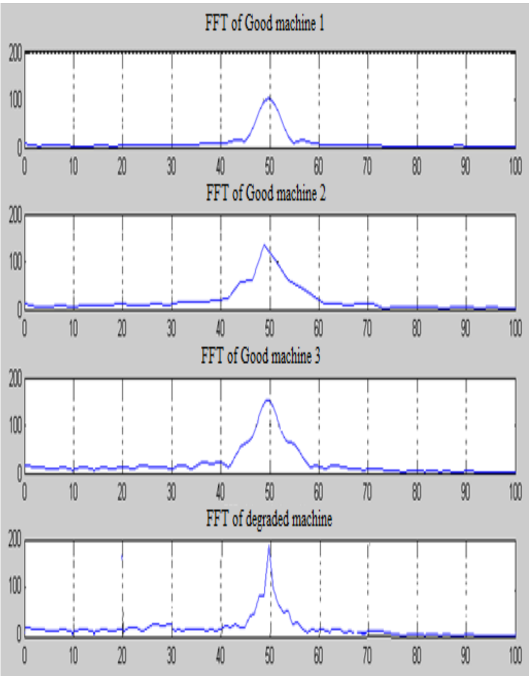

Fig. 7 Output of FFT Obtained in MATLAB for Real Data

Table 3. Frequency Obtained from Real Data

Type Of Machine	Frequency (Hz)
Good Machine 1	49.6646
Good Machine 2	48.9621
Good Machine 3	49.6969
Degraded Machine	49.786

Figure 8 and Table 4 shows an MPM output which is obtained for real time data. Using Matrix Pencil Method one can determine the signal frequency as well as damping by which the machine condition can be estimated. Figure 9 shows an MSE vs. SNR curve for frequency. The graph is plotted for the synthetic data. From the above MSE vs. SNR graph it is clear that when the number of input signals increases the MSE value also increases. For one signal input the MSE value is less when compare to the three signals.

```
freq_machine1 =   49.6646

damp_machine1 =   10.0475

freq_machine2 =  48.9621

damp_machine2 =   9.2181

freq_machine3 =   49.6969

damp_machine3 =   8.0515

freq_degraded_machin =  49.7895

damp_degraded_machin =   3.5188
```

Fig. 8 MPM Result Obtained in MATLAB

Table 4. Frequency and Damping Obtained from Real Data

Type of Machine	Frequency Hz)	Damping Coefficient
Good Machine 1	49.6646	10.0475
Good Machine 2	48.9621	9.2181
Good Machine 3	49.6969	8.0515
Degraded Machine	49.786	3.5188

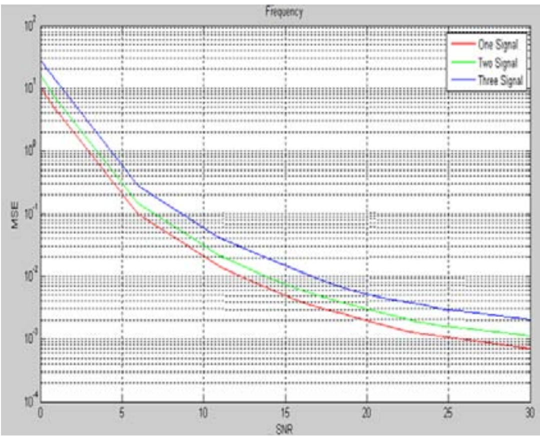

Fig. 9 MSE vs. SNR Curve for Frequency

Figure 10 shows an MSE vs. SNR curve for frequency and damping. The graph is plotted for the synthetic data. From the above MSE vs. SNR graph it is clear that when the number of input signals increases the MSE value also increases. For one signal input the MSE value is less when compare to the three signals input.

Figure 11 shows an MPM output which is obtained for real time data on the target board i.e. TMS320DM642. The output is obtained by simulating all the four different signals which were captured from different machines. As shown in the above figure as the machine get deteriorated the damping of the signal reduces up.

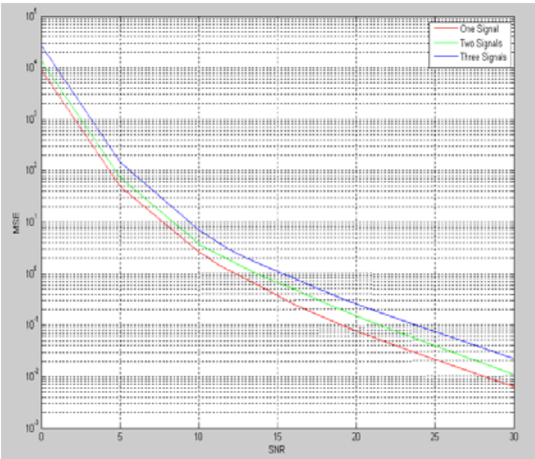

Fig. 10 MSE vs. SNR Curve for Damping

Fig. 11 Output Obtained for the Implemented System Using TMS320DM642

6. CONCLUSIONS

Embedded condition monitoring of rotating machines is carried out using signal processing techniques (MPM and FFT) and a MATLAB code is developed for MPM and FFT. Verification of the algorithms through real field data captured by data acquisition systems for monitoring the health of rotating machines has been performed. The performance of the MPM under AWGN with various SNR's is carried out. And MSE vs. SNR curves are developed for both frequency and damping. The tested MPM and FFT algorithm is ported on DSP platform TMS320DM642. Matrix Pencil Method can be employed for real time condition monitoring of rotating machines based on the transient signal analysis.

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