International Journal of Multidisciplinary Research and Development

IJMRD 2015; 2(3): 213-220
www.allsubjectjournal.com
Impact factor: 3.672
Received: 22-02-2015
Accepted: 14-03-2015
E-ISSN: 2349-4182
P-ISSN: 2349-5979

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Correction of transient response of sensor by adaptive technique

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Abstract
This paper presents a method of dynamic compensation of piezoelectric sensor using adaptive control technique. The case can be illustrated by showing how the response of a piezoelectric sensor can be improved. The dynamic response of the piezoelectric sensor can be improved to obtain a steady state response using adaptive control techniques. This presents a method of sensors dynamic characteristics correction, a method of transient response signal forcing in the measuring circuit of the sensors. So compensation technique is used which helps in tracking the variation of the measuring values. Firstly, a model of the sensor is made. The next step deals with the modeling of filter technique in order to minimize the oscillations in the output. Then the output is discretized using proper sampler. The rectification of the sampler output is done using an adaptive control technique. It deals with the nonlinearity in the output response of piezoelectric sensors and the adaptive control approach to eliminate this nonlinearity in the output. The trained signal is a delayed version of the input signal which is actually the sensor output. Implementation of RLS, LRLS, LMS and NLMS algorithms are used.

Keywords: Dynamic Compensation, Piezoelectric sensors, Adaptive control, Transducer, FIR, IIR, Temperature Sensor, Thermistor, Thermometer, Bimetal, Thermocouple, LMS, RLS.

1. Introduction
1.1 Motivation
There are some sensors which have oscillatory response that needs time to settle down. In this, the dynamic nonlinearity of the sensor is corrected to give a steady state response. Here, the adaptive filter parameters are varied in order to obtain desired sensor output. In this work, the maximum peak overshoot is minimized or, rather the damping is reduced. The filter parameters change as per LMS algorithm.
Piezoelectric sensors find its widespread use in the fields of quality assurance, process control and R&D in different industries. In automotive industries, piezoelectric elements are used in monitoring combustion during the development of internal combustion engines. Besides, these sensors are used in various applications like medical, aerospace, nuclear instrumentation, etc.

1.2 Components
1.2.1 Sensor
Sensor is a converter which measures a physical quantity and converts it into a signal that can be detected by an observer or by an instrument. A sensor can also be defined as a device which receives and responds to a signal when touched. A sensor is used to detect a parameter in one form and convert it to another form of energy usually an electrical signal.

1.2.2 Transducer
A transducer is a device that converts a signal in one form of energy to another form. Basically, it converts a non-electrical signal to an electrical form. Any device which converts energy can be treated as transducer.

1.2.2.1 Active Transducer
These transducers do not need any external source of power for their operation. Therefore they are also called as self-generating type transducers.

1.2.2.2 Passive Transducer
These transducers need external source of power for their operation. So they are not self-generating type transducers.

1.2.3 Piezoelectric Sensor
This is a device that uses piezoelectric effect to measure pressure, acceleration, and force by converting them to an electrical charge. The various operating modes in such type of sensor are: Transverse, Longitudinal and Shear. The figure below provides mechanical model of piezoelectric sensor.

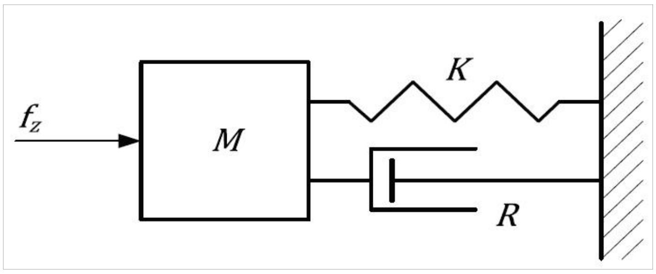

Fig 1: Mechanical model of piezoelectric sensor

1.2.4 Rise Time
The rise time may be defined as the time required by the response to reach its final value from the initial value. For under damped second order systems, 0% to 100% rise time is usually used. For over damped systems, 10% to 90% rise time is usually used.

1.2.5 Peak Time
The peak time may be defined as the time required by the response to reach the first peak of the overshoot.

2. Theoretical Review
2.1 Implementation of various adaptive Techniques
2.1.1 Structure of Filter
A finite impulse response (FIR) filter is a kind of a digital filter in which the impulse response is finite because it settles to zero in a finite number of sample intervals. This is in contrast to infinite impulse response (IIR) filters, which have internal feedback and may continue to respond indefinitely. The impulse response of an Nth-order FIR filter lasts for N+ 1 sample, and then dies to zero.
The difference equation is given which defines how the input signal is related to the output signal:
where: X[n] is the input signal,
Y[n] is the output signal, and b_i is the filter coefficients.
N is known as the filter order; an Nth-order filter has (N + 1) terms on the right-hand side; these are commonly referred to as taps.
The previous equation can also be expressed as a convolution of filter coefficient and the input signal.

$$y[n] = \sum_{i=0}^{N} b_i x[n-i].$$

2.1.2 FIR filters
In the context of signal processing, finite impulse response (FIR) filter is a filter with impulse response of finite duration. The impulse response of an Nth-order discrete-time FIR filter lasts for N + 1 samples, and then settles to zero. The structure of FIR filter is given below.

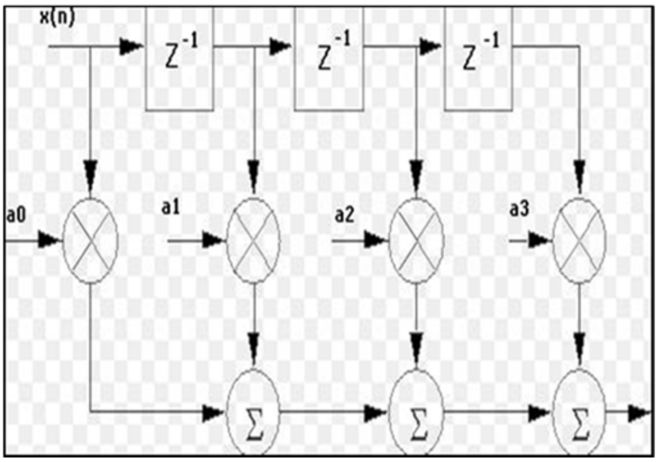

Fig 2: Structure of an FIR filter

2.1.3 Adaptive filter
Adaptive filter may be defined as a self-adjusting filter which is capable of adjusting its transfer function as per the optimization algorithm that is driven by an error signal. The adaptive filter makes use of feedback in the form of an error signal to improve its transfer function in order to suit the changing parameters.
The block diagram of adaptive algorithm is shown below.

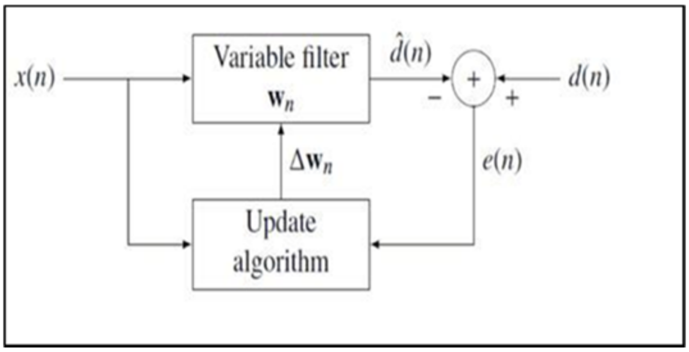

Fig 3: Block diagram of an adaptive filter

3. Material and Methods
3.1 Sensors
There are various types of sensors among which the significant types are listed below:

3.1.1 Temperature sensor
The various types of temperature sensors include thermistor, thermocouple, resistance thermometer, thermometer, bimetal, etc.

3.1.2 Thermistor
A thermistor is a type of resistor whose resistance varies significantly with temperature.

$$y[n] = b_0 x[n] + b_1 x[n-1] + \cdots + b_N x[n-N]$$

3.1.3 Thermometer
A thermometer is a device that can measure temperature or temperature gradient with the help of different principles. A thermometer has two basic components: the temperature sensor on which the physical change take place and some means of converting this physical change into a numerical value.

3.1.4 Bimetal
An object that is composed of two separate metals joined together is termed as bimetal. Instead of being a mixture of two or more metals, like alloys, bimetallic objects comprise of layers of different metals.

3.2 Non Linearity in Sensor output
Sensors do not show linear output characteristics. The deviation from perfect linear characteristics exhibited in the output characteristics of the sensors produces fallacious data in the output. In order to eliminate this discrepancy, various measures can be taken. Among these, one method is using adaptive filter which is controlled by various adaptive algorithms. Load cell is also a kind of sensor which exhibits similar type of non-linearity in its output characteristics.
There are three types of methods used to compare and investigate for calculating weight of any types of item.

3.2.1 Average Method
This type of load cell modeling method is based on the assumption that the oscillatory response has enough settle time. In stable time the weight can be found from the value of the graph.

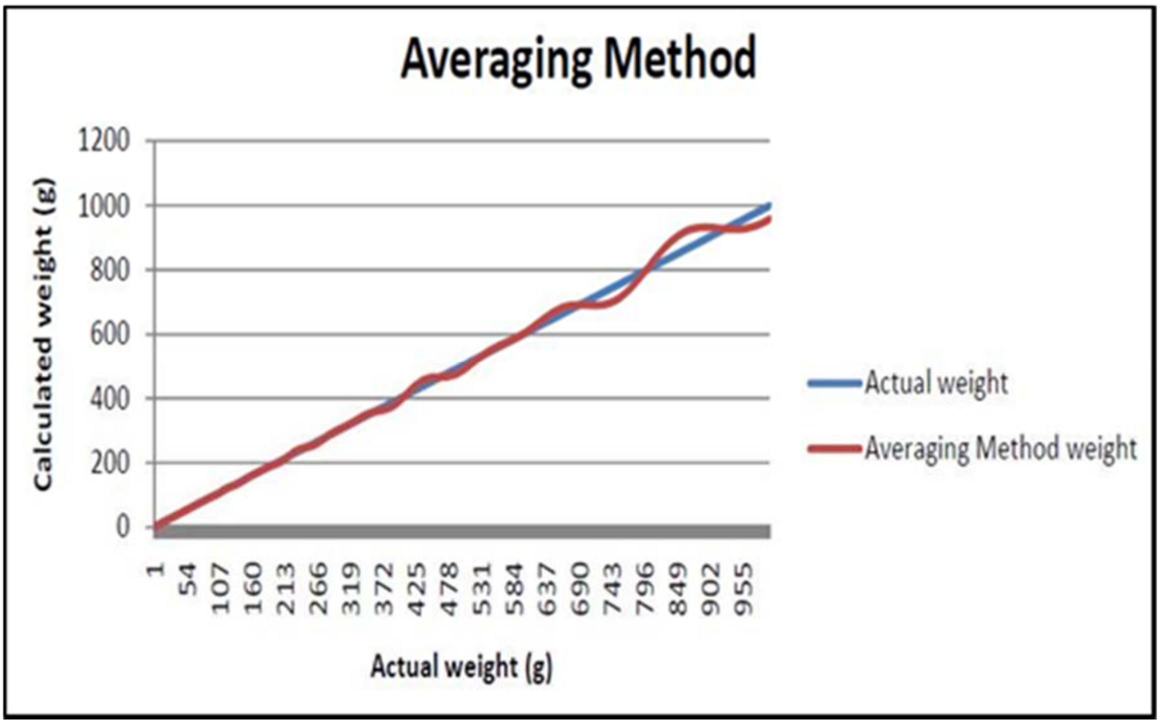

Fig 4: Averaging Method graph.

3.2.2 Frequency Method
The method describe in this section has been named the frequency method.

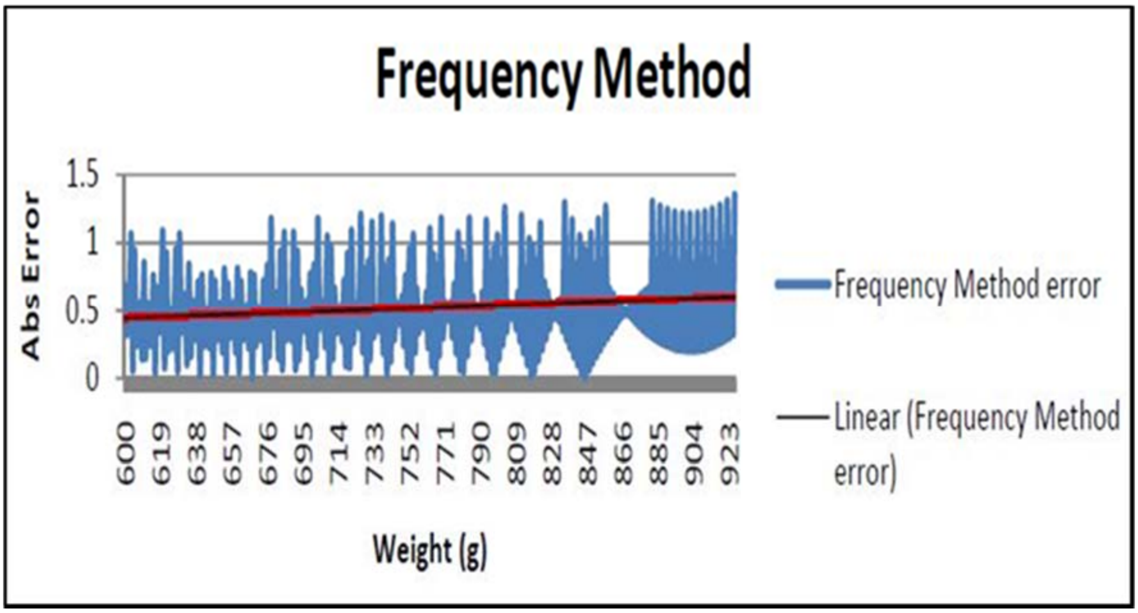

Fig 5: Frequency method

3.2.3 Damping Method
Considering the model of the load cell system presented in above figure. Another method can be described.

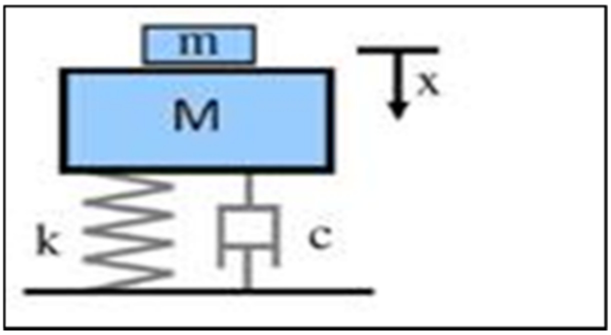

Fig 6: Mass spring damping system

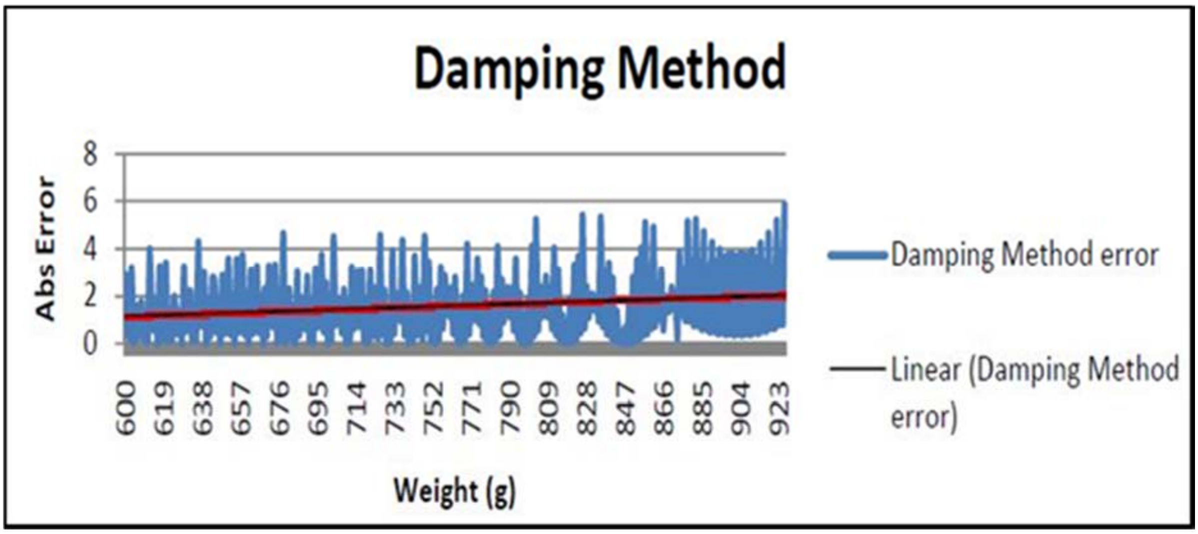

Fig 7: Damping Method

4. Results and Discussion
4.1 Thermocouple
A thermocouple is a device consisting of two dissimilar conductors in contact, which produces a voltage on heating. The size of the voltage is dependent on the difference of temperature of the junction to other parts of the circuit. Thermocouple is a type of temperature sensor used for measurement and control. It can also be used for converting a temperature gradient into electricity. A thermocouple works on the principle of See beck effect which states that when a conductor is subjected to a thermal grad

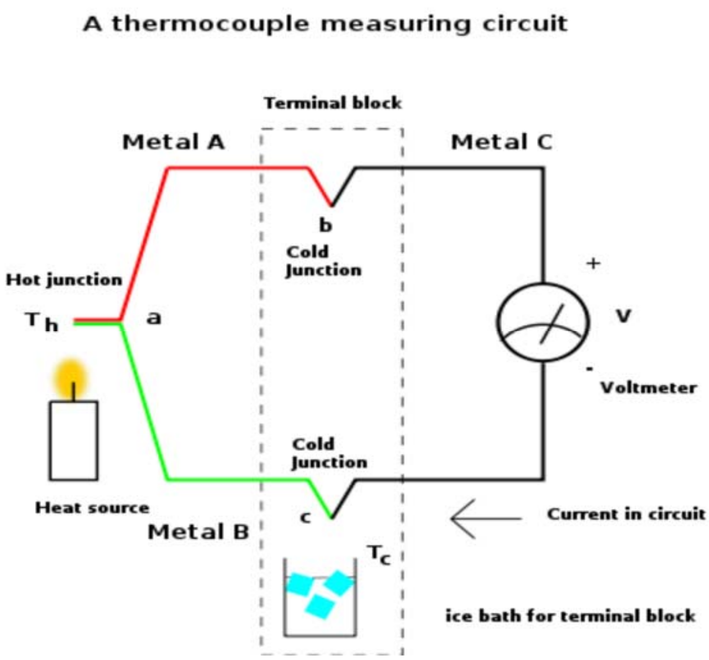

Fig 8: Model illustrating a thermocouple

4.2 Output Voltage Characteristics of Thermocouple
Output Voltage characteristics of Thermocouple as given below.

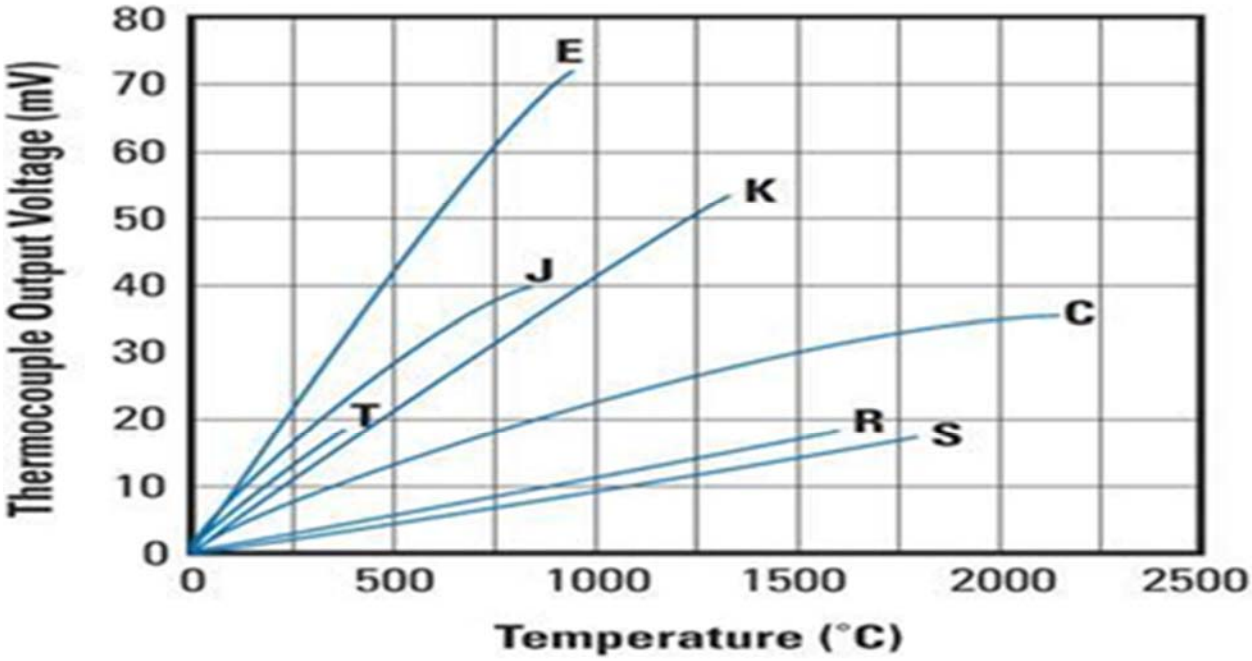

Fig 9: Output voltage characteristics of thermocouple

4.3 Output Response of a sensor
Output response of sensors graph as shown below.

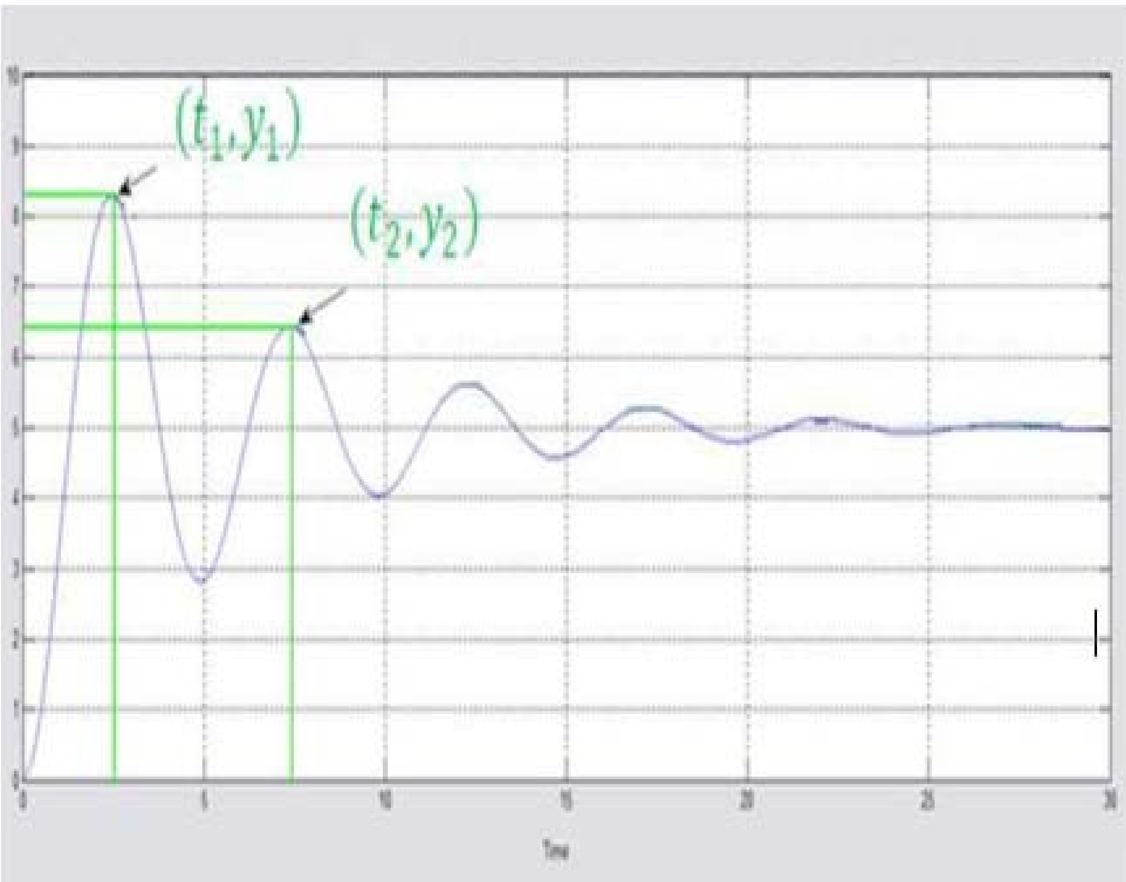

Fig 10: Graph of Output response of sensor

4.4 Output Voltage Characteristics of Piezoelectric Sensor
Output Voltage characteristics of Piezoelectric Sensor as given below.

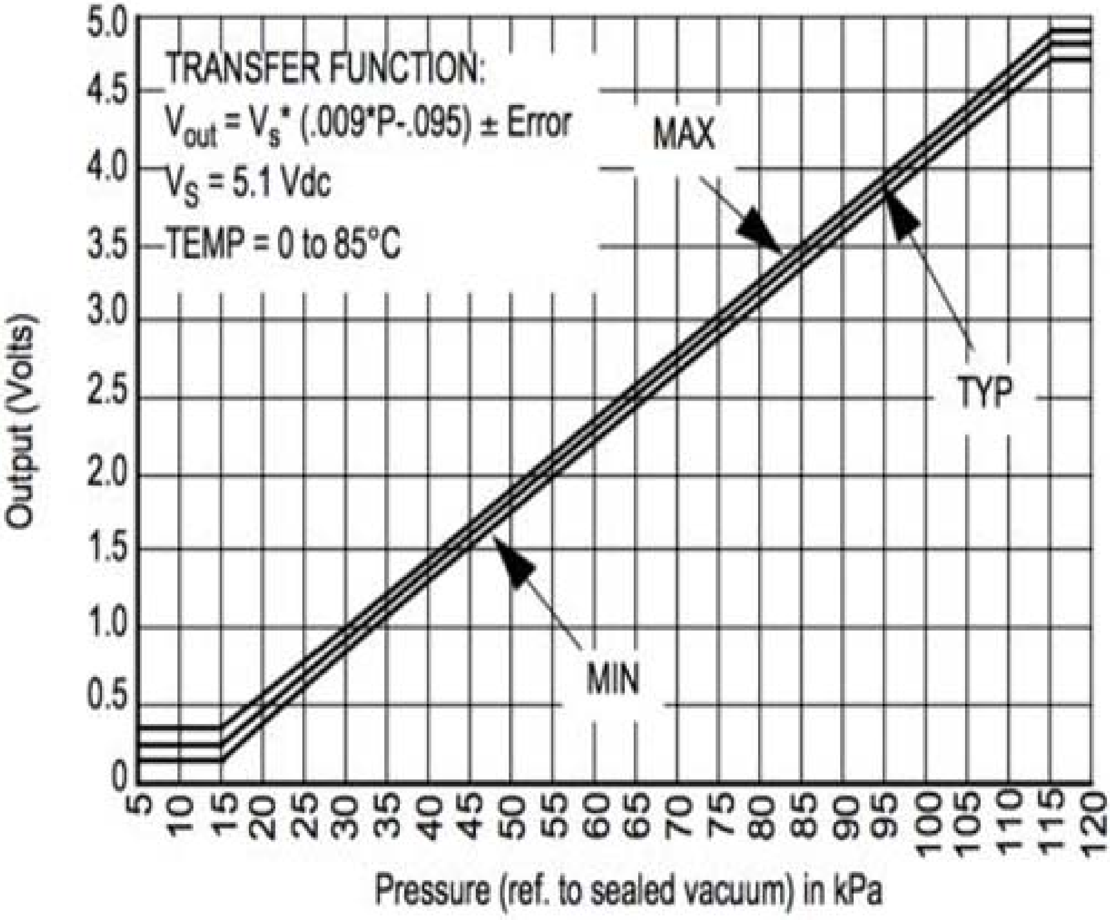

Fig 11: Output voltage characteristics of a piezoelectric sensor

A thermocouple is a device consisting of two dissimilar conductors in contact, which

4.5 Implementation of various Adaptive Algorithms
4.5.1 RLS Algorithm: The Recursive Least Squares adaptive filter is an algorithm that is capable of finding the filter coefficients recursively which in turn minimizes a weighted linear least squares cost function related to the input signals.

4.5.2 Lattice recursive least squares filter (LRLS)
The Lattice Recursive Least Squares adaptive filter is similar to the standard RLS but it requires lesser arithmetic operations. It is advantageous over conventional LMS algorithms as it has faster convergence rates, modular structure, and due to its insensitivity to variations in Eigen value spread of the input correlation matrix.

4.5.3 Normalized lattice recursive least squares filter (NLRLS)
The normalized lattice recursive least squares filter contains lesser number of recursions and variables. This can be calculated by applying normalization to the internal variables of the algorithm. This is not practically used in real-time applications due to the number of division and square-root operations.

4.5.4 Least Mean Square Algorithm:
In 1959, Widrow and Hoff through their studies of pattern recognition first developed the Least Mean Square (LMS) algorithm. The LMS algorithm is a kind of adaptive filter called stochastic gradient-based algorithm. This is because; it uses the gradient vector of the filter tap weights to converge on the optimal wiener solution. This algorithm is very popular for its computational simplicity.

4.5.5 Trained Adaptive Linear Filter algorithm Flow Chart
Flow chart as shown below.

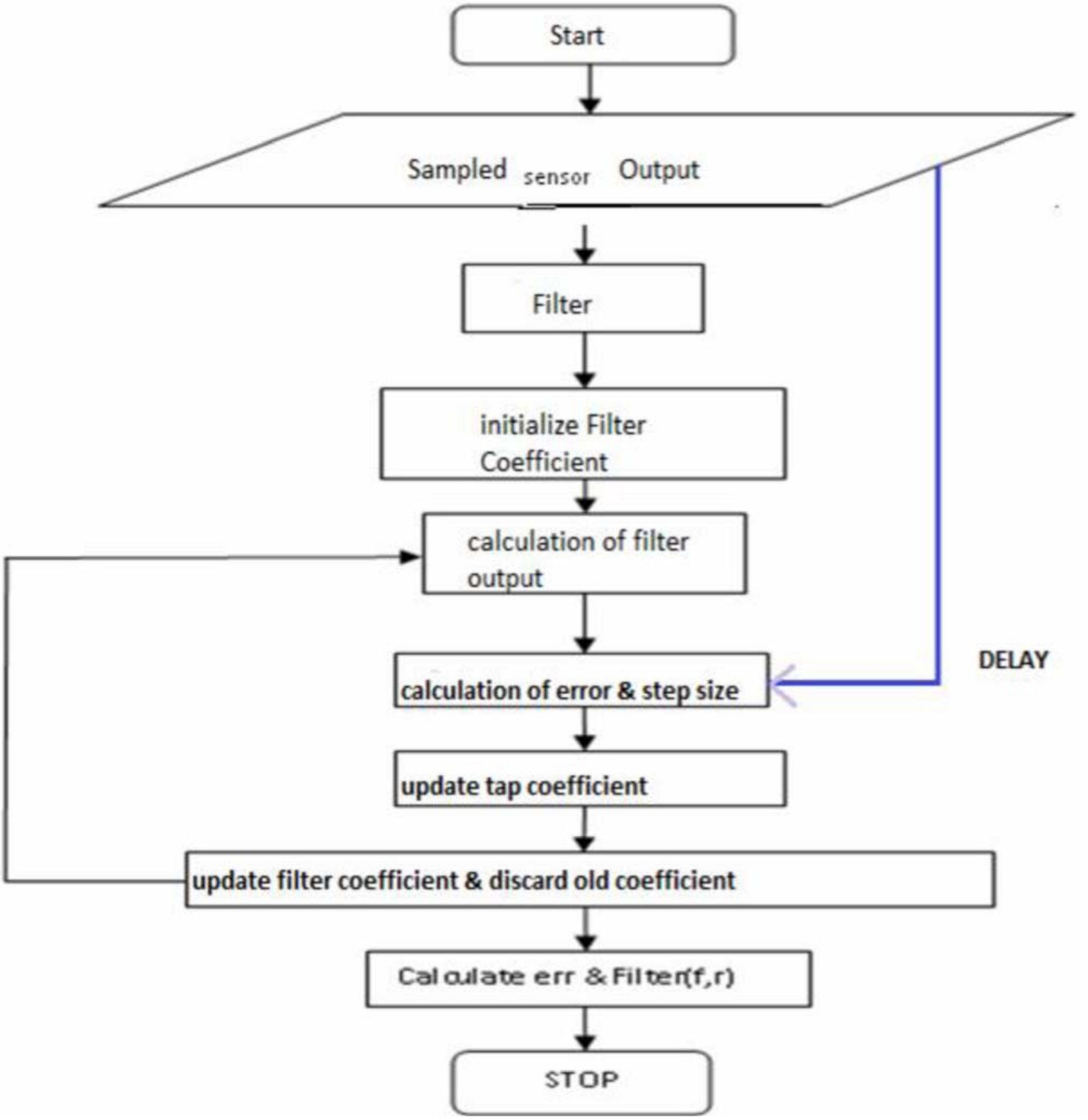

Fig 12: Trained Adaptive Linear Filter algorithm flow chart.

4.5.6 A Trained Approach to the Adaptive Filter
The block oriented design of the previous section requires substantial computation even when the system delay is known since it requires calculating the inverse of an (n+1) X (n+1) matrix, when n is the largest delay in the FIR linear filter. This section considers using an adaptive element to minimize the average of the squared error.

JLMS = ½ avg {e2 [k]}

Observe that JLMS is a function of all the filter coefficients fi since

$$e[k] = s[k-\delta] - y[k] = s[k-\delta] - \sum_{j=0}^{n} f_j r[k-j]$$

where r[k] is the received signal baseband after sampling. An algorithm for the minimization of JLMS with respect to the ith filter coefficient fi is

$$f_i[k+1] = f_i[k] - \mu \left. \frac{dJ_{LMS}}{df_i} \right|_{f_i = f_i[k]}$$

The following Matlab code implements an adaptive filter design. The opening and the closing of the program are similar to the previous program. The heart of the recursion lies in the for loop. For each new data point, a vector is built containing the new value and the past n values of the received signal. This is multiplied to f to make a prediction of the next source symbol, and the error is the difference between the prediction and the reality (this is the calculation of e[k]). The filter coefficients f is,

```
b= [0.5 1 -0.6];              % define channel
m=1000;
s=sign (randn (1, m));        % binary source of length m
r=filter (b, 1, s);           % output
n=4; f=zeros (n, 1);          % step size and delay delta
mu=.1; delta=2; for
i=n+1: m                      % iterate
rr=r(i:-1:i- n+1)‟;           % vector of the received signal
e=s(i-delta)- f‟*rr;          % calculate
error
f=f+mu*e*rr;                  % update filter coefficients
end
y=filter (f, 1, r);           % equalizer is a filter
dec= sign(y);                 for %quantization
sh=0:n                        % error at different delays
err (sh+1) =0.5*sum(abs(dec(sh+1:end)-s(1:end-sh)));
end
% output  rr,e,f
mu=       0.1000
rr=       0.9000
          2.1000
          0.1000
         -0.9000
e=       -0.0953
f=       -0.2621
          0.6604
          0.2564
          0.1543
```

4.5.7 Matlab coding implementating LMS algorithm

Starting from the basics, the following program in MATLAB takes an input vector u and a reference signal d, both of length N, and calculates the error e for all time instants.

```
Function [e, w] =lms (mu, M, u, d);

%    Call:
%    [e, w] =lms (mu, M, u, d);
%    Input arguments:
%    mu = step size, dim 1x1
%    M = filter length, dim 1x1
%    u = input signal, dim Nx1
%    d = desired signal, dim Nx1
%    Output arguments:
%    e = estimation error, dim Nx1
%    w = final filter coefficients, dim Mx1
%    initial values: 0
w=zeros (M, 1);
%number of samples of the input signal
N=length (u);
%Make sure that u and d are column vectors
u=u (:);
d=d (:);
%LMS
for n=M: N
uvec=u (n:-1: n-M+1);
e (n) =d (n)-w'*uvec;
 w= w+mu*uvec*conj (e (n));
end
e=e (:);
```

4.5.8 Simulation and Results

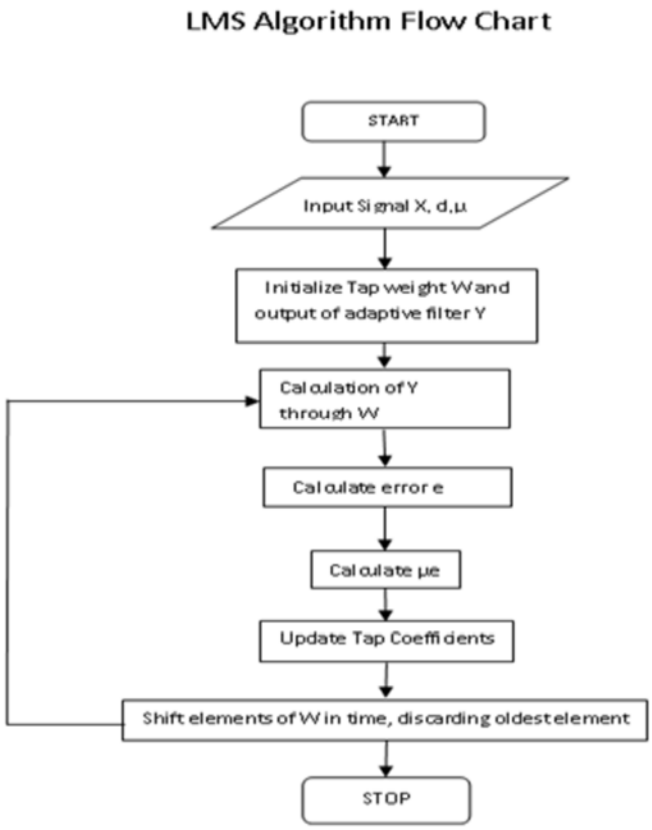

Fig 13: Flow chart showing LMS algorithm

5. Conclusion

In this paper we are able to perform effective response compensation of sensor output using digital adaptive technique or rather we can say discrete adaptive technique. The LMS algorithm works well in the process of elimination of transient response of the sensor. Non linearity in the sensor can be compensated by adaptive technique. Simulation and results verifies the feasibility of the proposed technique.

In this, it is shown how the transient response in the output of a load cell can be eliminated. The result is feasible for various other types of sensors.

6. References

1. Adaptive Filter Concept:- Wiener in 1949 and Kolmogorov in 1942.
2. Widrow & Hoff (1960): least-mean square algorithm (LMS) = delta rule.
3. Adaptive control by karl johan Astrom Bjorn.