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Development of Intelligent Time/Frequency Based Signal Detection Algorithm for Intrusion Detection System

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***Abstract*—**For the past couple of decades Weak signal detection is of crucial importance in various engineering and scientific applications. It finds its application in areas like Wireless communication, Radars, Aerospace engineering, Control systems and many of those. Usually weak signal detection requires phase sensitive detector and demodulation module to detect and analyze the signal. This article gives you a preamble to intrusion detection system which can effectively detect a weak signal from a multiplexed signal. By carefully inspecting and analyzing the respective signal, this system can successfully indicate any peripheral intrusion. Intrusion detection system (IDS) is a comprehensive and easy approach towards detecting and analyzing any signal that is weakened and garbled due to low signal to noise ratio (SNR). This approach finds significant importance in applications like peripheral security systems.

***Keywords*—**Data Acquisition, fast frequency transforms, Lab VIEW software, weak signal detection.

I. Introduction

SECURITY is one of the major concern these days globally. An intensive work is going on in research field to develop such intelligent systems that can effectively cope with any ambush. Considering existing law and order situation particularly in Pakistan and worldwide in general there is a need to develop an intrusion detection system for Armed Forces and law enforcement agencies. This article gives an overview of such a system that detects intrusion based motion detection scheme. Detection of enfeebled signal from garbled one is of primary interest. Weak signal detection techniques find its major use in Radar, Electrical machines, Satellite communication etc.

Weak signal detection is a method to extract a signal of our interest from a noisy channel. It's necessary because it carries useful information based on what certain desired actions are performed. Weak signal detection technique is a sophisticated technology and is highly appreciated. It has grown rapidly in the last couple of years for negating the concept that some weak signals cannot be measured and analyzed. The purpose of the technology is to study different factors that cause noise during transmission and to effectively recover the original signal buried in noise. This technique is used to acquire signal from noise or could be used to enhance signal detection capabilities of the system by improving signal to noise ratio (SNR) by implementing some advance techniques and methods [1]. There are several methods of detection including periodic signal self-correlation detection method, periodic signal sampling point's integral method and signal frequency signal lock detection method etc. All these methods works on principle of differentiating signal and noise in frequency and time domain and then extracting the required signal for further use [2].

The developed system discussed in this paper detects the required signal in frequency domain and then extracting the useful information using some filtering techniques. This system is designed in LabView Software which is developed by National Instruments. LabView is widely used in virtual instrument design and development. The detecting software developed under the Lab Windows (an integrated development environment of NI Company's) acquires data from testing equipment which in our case is piezo electric vibration sensors and manipulates the signal in software by applying some digital signal processing techniques.

A. Literature Review

Detection of weak signal finds its application in variety of fields especially in signal processing domain. There are many classical methods of detecting these signals, one of the novel technique is wavelet based. This technique is most common in applications where we need to extract signal from noise, the process is known as denoising. Extraction of signal from noisy environment is accomplished by thresholding wavelet coefficients. Substantial work has been carried out in this direction, however this method fails when working with signals having low signal to noise ration (SNR). In case of biosignals, electrical signals that are acquired from human body, that have below zero SNR this technique fail [3].

Wall mounted vibration sensors are used whose sensitivity is adjustable according to our need. These sensors act like a transducer and generate an analog voltage signal whenever they sense an intrusion across them. The various characteristics of the signal generated like frequency or amplitude depends upon the magnitude of force applied. Vibration sensors find its usage in variety of applications. There are different kinds of vibration sensors and the area of application is dependent on functionality and manufacturing of them. Some find their usage in monitoring machine health, aerospace transportation, damage detection for civil infrastructures and mechanical processes like fiber optical vibration sensors [4]. For detecting any anomaly in machine engines the output of these sensors may vary from several KHz to MHz. Multilongitudinal mode fiber sensor [5], Fabry Perot Interferometers have been proposed for frequency measurement in the range of several KHz.

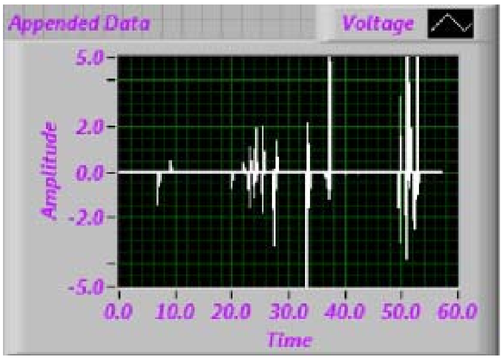

Fig. 1 Sensor Output

Testing any system by vibration analysis is to compare the vibration pattern of the resultant output of that system with a reference or standardized vibration signature during real time monitoring. The system is considered to be running in a perfect condition if two signatures match, otherwise there is a fault. The standard vibration signature is the data acquired when the system is running in perfect condition [6].

B. System Architecture

Wall mounted piezo electric vibration sensors were used for this system. Their output frequency response is directly proportional to their mechanical vibration. These sensors emit signals within milivolts range, so were amplified for long distance commutation. The sensitivity of the sensor can be adjusted by adding small masses to the tip of the sensor. The results after adding mass are shown in following table. The real time output of this signal is shown in Fig. 1.

For this system two techniques were used to monitor signals i.e fast fourier transforms and Joint time frequency analysis using short time fourier transform and their results were compared. This system is expected to work effectively through all year long facing all weather conditions. Joint time frequency analysis gives a better idea of signal processing both in time-frequency domains. This method is crucial for analyzing non-stationary signals whose spectral components continue to vary over all period of time. The system works on detecting the frequency components of the real time signals continuously. Initially we fed some testing data to our system so as it becomes intelligent enough to make its own decisions. The methodology proposed for this system can be well understood by the help of Fig. 2.

TABLE I
LDT0 AS VIBRATION SENSOR

Added Mass	Baseline Sensitivity	Sensitivity at Resonance	Resonant Frequency
0	50 mV/g	1.4 V/g	180 Hz
1	200 mV/g	4 V/g	90 Hz
2	400 mV/g	8 V/g	60 Hz
3	800 mV/g	16 V/g	40 Hz

C. Implementation of Intrusion Detection System

The basic functionality of the system can be perceived from the block diagram as shown in fig. 3. LabVIEW is used as a programming software as it's a potent tool used in industry for variety of applications. Wall mounted sensors will generate analog electrical signals upon detecting any vibration. These sensors communicate with DAQ card through RS-232 protocol. These real time analog waveforms are our point of interest and we need to analyze them in our simulating software i.e. LabVIEW. The block diagram representing the flow of the system is shown in Fig. 4.

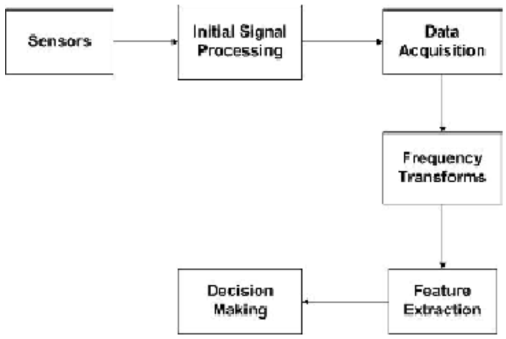

Fig. 2 Block Diagram

1) Astable Multivibrator Circuit:

Analyzing Fig. 1, we cannot differentiate the information embedded in it. As the sensors output continues to vary so does its spectral com- ponent. So it's almost impossible to judge which frequency components exist after applying fast frequency transforms. To cure this we used Astable 555 oscillator circuit to generate stable square wave output waveform with fixed frequency of upto 500 KHz of varying duty cycles from 50 to 100 percent. In order to get the 555 Oscillator to operate as an astable multivibrator, it is necessary to continuously re-trigger the 555 IC after each and every timing cycle. This triggering is achieved by connecting input trigger pin 2 with threshold input pin 6, to allow device to act as astable oscillator. This circuit needs a power supply of 5-10 volts, which is sufficiently generated by using amplifier circuit. For sensor S_1 , values of R_1 , R_2 were 555Ω and for C_1 , C_2 were 22μF. For second sensor S_2 , values of R_1 , R_2 were 480Ω and for C_1 , C_2 were 10μF respectively. In each cycle, capacitor 'C' charges itself through two timing resistors R_1 and R_2. The capacitor discharges only through R_2 as it's connected to discharge terminal to one side. The charging and discharging time of the capacitor is determined by following formulas as referred by eq. 1 and 2.

$$t_1 = 0.693(R_1 + R_2)^{*}C \quad (1)$$
$$t_2 = 0.693^{*}R^{*}C \quad (2)$$

The output of 555 oscillator will continue to charge and discharge between 2/3 Vcc and 1/3 Vcc until the power supply is removed. The frequency of oscillation is inversely proportional to time period 'T' which is given by equation 3.

$$f = \frac{1}{T} = \frac{1.44}{(R_1+2R_2)C} \quad (3)$$

Real time signal that was acquired through DAQ card through oscillator circuit is shown in Fig. 5.

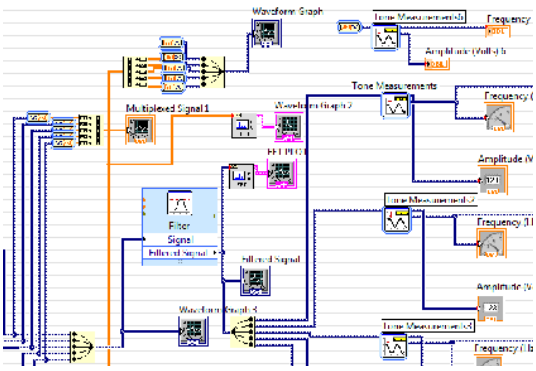

Fig. 3 Graphical Code

2) LM 741 Operational Amplifier

LM-741 was used to make this circuit work as an inverting amplifier. This circuit gives a negative (inverted) amplified feedback through a feedback resistor R_f. For DC voltage only amplification takes place as there is no phase on DC. The amplification occurs according to following formula in eq. 4.

$$Vout = -\left(\frac{Rf}{Rin}\right) * Vin \quad (4)$$

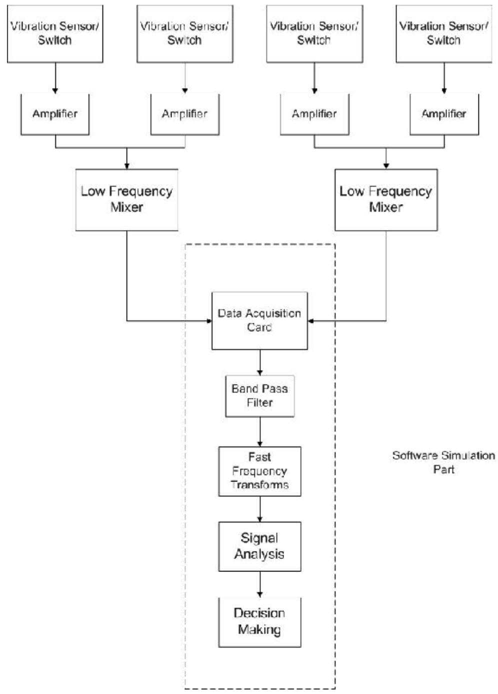

Fig. 4 System Architecture

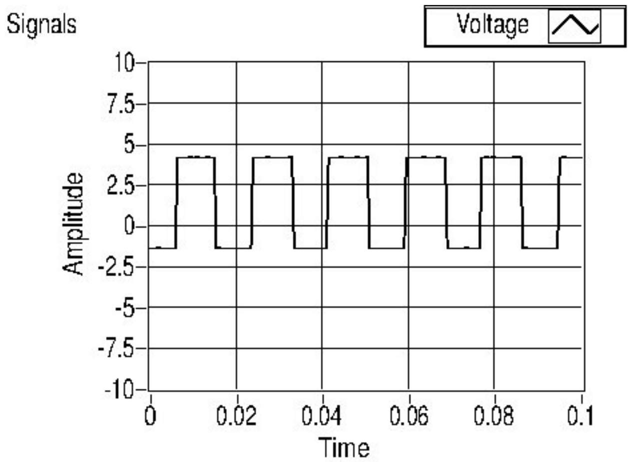

Fig. 5 Real Time Signal

3) Low Frequency Mixer

Low frequency mixer is a circuit that is used where two input frequencies (f_1 and f_2) produce a sum or difference frequencies. This circuit has a gain of 10 and contains a low pass mono-pole filter made by using 1MΩ and 1pF with a corner frequency of 1 KHz. The circuit can be used for relatively high frequencies as long as the desired difference of two input frequencies is within the bandwidth of amplifier and RC low pass filter. For schematic circuit diagram please refer to fig. 6. The diode at the positive (+) input can be used for non linear signal processing. The signal with larger amplitude serves as a local oscillator V_1 and the signal with smaller amplitude us given at the second input terminal V_2. The sum or difference in frequencies is filtered out and given at output terminal.

4) Data Acquisition

Real time signals can be analyzed in LabVIEW by using National Instruments (NI) USB-6009 14-bit data acquisition card. It has a maximum sampling frequency of 48 Ks/sec. We garnered our training data at sampling rate of 1 KHz with continuous mode. Data acquisition (DAQ) system uses data acquisition board which passes electrical signals generated by sensors to LabView for software analysis and data logging. We can use DAQ card that uses a PXI bus, a compact PCI bus or a computer USB port to create a portable, versatile and rugged measurement system. We used NI-DAQ Assistant in LabVIEW to communicate with data acquisition board.

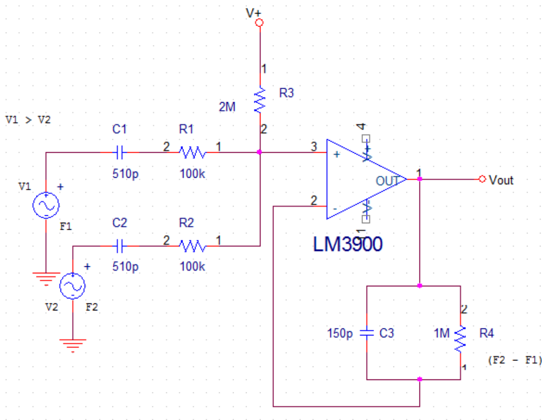

Fig. 6 LM-3900 Schematic Diagram

5) Filtration Technique

Initially garbled signal is received due to noisy environment over the transmission channel. There is no evident information in this time domain signal to be analyzed. After converting it to frequency domain by applying FFT, we removed noise by applying filter. There are several techniques used for filtering signals like adaptive filtering, wavelet thresholding [7], non linear filtering and independent component analysis etc [8]. It's a general practice to use a Band pass filter. For this purpose there is a dedicated block for filtering signals. Using a low pass filter is not a good approach because the signal bands are close enough to noise, which make it difficult for the system to eliminate the original waveform. Here we can set the frequency band of our desired range and can eliminate all unnecessary frequency components from the waveform.

D.Experimental Results

A total of five wall mounted vibration sensors were used. From the tests it was verified that sensors exhibits an output signal with a frequency ranging from 10 Hz to 180 Hz upon human interference. In real time, many signals with different frequency components are generated because the sensor comes into contact with different objects apart from human beings. So by carefully measuring the Frequency components of the waveform after filtering it and comparing it with pre-defined frequency range, a vigilant and dynamic intrusion monitoring system can be developed. The simulation results have verified the efficient behavior of the model that continuously monitors the real time data. The front panel designed for the system can be viewed in the Fig. 7.

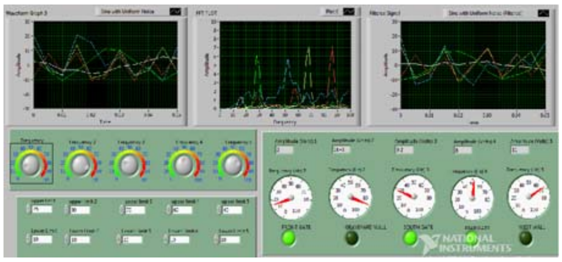

Fig. 7 Front Panel

The experimental results show the power of LabVIEW programming as its very convenient to use. With a careful simulation in LabVIEW we can develop an effective time-frequency based joint detection algorithm which avoids complicated hardware connection. The programming is simple and all the parameters of the code can be altered by a mere click. It not only reduces the design complications but also effectively detect a weak signal in frequency domain. Performance of the system can be enhanced by increasing sampling frequency.

1) Frequency Domain Analysis

Fourier transforms are one of the most powerful and efficient techniques for solving a wide variety of problems regarding physical, biological and engineering sciences. The cardinal idea behind fourier transform is to represent functions and their derivatives as sums of sines and cosines having an order O(N LOG N) operation represented in equation 5. Fourier transform decom- poses a periodic signal into its sinusoids at integer multiples of the fundamental frequency. Mathematically its represented by equation 6.

$$f(x) = \frac{a_0}{2} + \sum_{n=1}^{\infty}(a_n \cos \mathrm{nx} + b_n \sin \mathrm{nx}) \tag{5}$$

$$X(f) = \int_{-\infty}^{\infty} x(t).e^{-j2\pi ft}dt \tag{6}$$

The Inverse Fourier transform is shown by eq. 7

$$x(t) = \int_{-\infty}^{\infty} X(f).e^{j2\pi ft}df \tag{7}$$

In above equations, 'f' represents frequency, 't' shows time and 'x' denotes original signal. Its worth mentioning here that 'x' denoted the signal in time domain and 'X' denoted original signal in frequency domain. This convention is used to distinguish between two formats of signal. The first equations shows the Fourier transform of x(t) and second one shows its inverse fourier transform in terms of X(f). Let us explain these equations briefly. The original signal x(t) is multiplied with an exponential term having certain frequency 'f' and then integrated over all times. The exponential term mentioned in equations 6 and 7 can also be written as 8

$$\cos(2\pi f t) + j\sin(2\pi f t) \tag{8}$$

The real time signal does not provide sufficient information, because the signals spectra overlap with noise. In order to observe an intrusion at any particular instant of time we need to study the frequency components of signal. For this we take a frequency transform of the signal. LabVIEW has a block named "FFT Spectrum (Magnitude-Phase)" which effectively transforms signal into frequency domain and returns magnitude and phase of frequency components of the signal by creating a graph. The following fig. 8 shows the amplitude and corresponding frequency of all three respected signals. Magnitude shows the magnitude of the FFT spectrum in Hertz (Hz), 'f0' returns start frequency and 'df' returns frequency resolution in terms of Hz. If input waveform is in volts then magnitude has units volts-rms (Vrms), otherwise if other units are specified then magnitude units are dependent on input signal unit-rms.

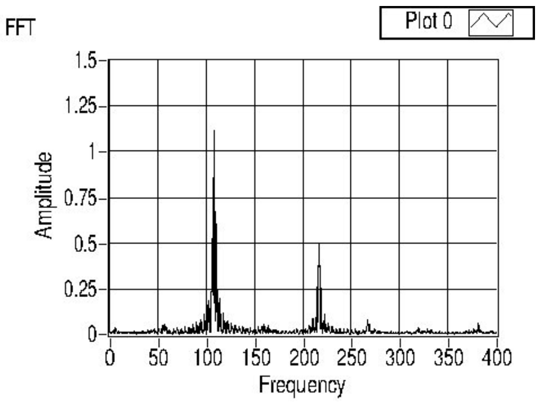

Fig. 8 Spectral Domain

If we analyze the graph, we see a clear picture of all the frequency components that were present in the real time signal. This makes it very easy for the system to compare with the pre-testified frequency range. Now the system can intelligently judge the intrusion based on comparison of real time data with the standard one.

2) Short Time Fourier Transform

It is a fourier transform used to determine the sinusoidal frequency of all components of a signal for all time periods. Short time fourier transform (STFT) or Gabor transform is a powerful tool for exploring audio signals for e.g. speech signal [9]. It defines a class of time frequency distribution which specifies complex amplitude time vs frequency for any given waveform. Short-time Fourier transform and fractional Fourier transform is used to process the multi-frequency signals. The mathematical definition of STFT is given as follows in equation 9.

$$Xm(w) = \sum_{n=-\infty}^{\infty} x(n)w(n-mR).e^{-jwn} \tag{9}$$

where
x(n) = input signal at time n
w(m) = length M window function (e.g Hamming or Hanning)
R = hop size in samples between successive DTFT's
$\boldsymbol{X^m}$ = DTFT of windowed data centered about time mR

There is a slight difference between fourier transform and short time fourier transform, in STFT the signal is divided into small enough segments where as these segments can be assumed stationary. But in reality the signal's frequency components are continuously changing in a non periodic fashion. Here we assume that some portion of the signal as stationary. Short time fourier transform gives a fixed resolution at time instants. Let's consider an example of a continuous signal whose spectral components continue to change after some interval of time as referred in fig. 9. This is a non stationary signal as its frequency changes for all time instants. Region where signal to be stationary is too small then we consider it a stationary signal. We will look at it with a very small window (w). This window function is located at the beginning of the signal i.e. at t=0. At this time instant (t=0), the window function will overlap with first T/2 seconds, where 'T' represents the time period of the waveform. The window function and the signal are then multiplied. This way only the first T/2 seconds of the waveform or audio signal are chosen with the appropriate weighing of the window. This product is then used to be another waveform.

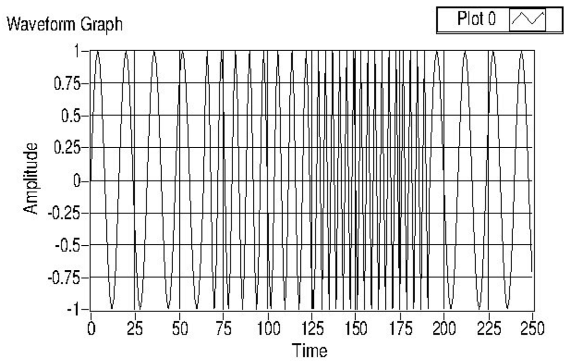

Fig. 9 Non Stationary Signal

After this frequency transform will be applied to this product. In practical applications the size of the window must be equal or smaller then the length of signal. Let's take an example of radar imaging which uses fourier transforms to retrieve Doppler information. Due to continuous motion of target, the Doppler frequency shifts are time varying. Hence frequency transforms are not suitable here because the image is blurred. Without resorting to sophisticated motion compensation algorithms, the image blurring problem can be resolved with the joint time-frequency transform [10]. Short-time Fourier Transform (STFT) can avoid the cross terms, however, it does not work well under the low signal-noise ratio [11]. The STFT guarantees positivity and is computationally efficient and very robust against noise. However, it suffers poor time frequency resolution [12].

This results in the FT of the first T/2 seconds of the signal. If this portion of the signal is stationary (assumed) then it will be true representation of the first T/2 seconds. The next step includes shifting of this window to a new location, multiplying with the signal and performing fourier transforms on that signal portion. This process is followed until the end of the signal by shifting the signal with some regular intervals. This approach is a revised addition of Fourier transform (FT). The following equation 10 satisfies the above conversation.

$$STFT_X^{(w)}(t,f) = \int [x(t).w * (t-t')].e^{-j2\pi ft} \tag{10}$$

Let's take a look at the equation where x(t) is the original signal, w(t) is the window function, and * is the complex conjugate. As we can infer from the equation that the STFT of the signal is nothing but the Fourier Transform of the signal multiplied by a window 'w' function. This way we can depict a true time-frequency representation (TFR) of the signal.

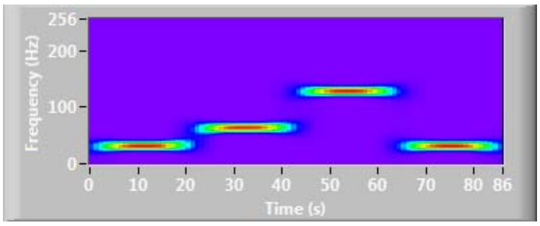

Fig. 10 Gabor Transform

The results of Gabor transform applied to our offline data are shown in fig. 9. The plot shows joint time-frequency analysis of our waveform from different sensors with spectral components around 30, 50 and 110 Hz respectively. The figure gives a good time localization representation of every spectral component. However while applying Gabor transforms there is always a tradeoff between time localization and frequency resolution.

II. Conclusion

Results show that, development of intelligent time /frequency based joint detection method was very feasible and convenient using LabVIEW programming. Its front panel is user friendly and allows the user to make desired actions upon detecting any anomaly in the system. Although Fourier analysis is sufficient to analyze spectral components of sensors output waveform. However if we are concerned with time localization as well then STFT or Gabor transform works well.

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