CLS Building Vibration Data Acquisition System

CLS Manual – 8.2.79.1 Rev. 1

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REVISION HISTORY

Revision	*Date*	*Description*	*Author*
A	2001-02-07	Conceptual Design	J.M. Vogt and E. Matias
0	2001-02-13	Conceptual Design	J.M. Vogt and E. Matias
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1. Introduction

The CLS is concerned with ambient and technical displacement vibrations between .1 Hz to 150 Hz in the order of a micrometer. An analysis of the on site vibrations at the CLS will yield the perturbations the beam lines will experience. The Vibration Monitoring System commissioned at the CLS is capable of accurately measuring the vibrations of concern.

2. Definitions and Abbreviations

DAQ: Data Acquisition

NI: National Instruments

VMS: Vibration Monitoring System

GUI: Graphical User Interface

DFT: Discrete Fourier Transform

FFT: Fast Fourier Transform

PGIA: Programmable Gain Instrument Amplifier

DAQ: Data Acquisition (Module)

SCXI: Signal Conditioning (Module)

ICP: Integrated Circuit Piezoelectric

3. References

Press, W. H. S. Teukolsky, W. Vetterling, B. Flannery 1992. *Numerical Recipes in C, The Art of Scientific Computing Second Edition.* Cambridge University Press.

4. Technical Specification

This unit shall:

a) Be able to monitor up to three accelerometers at each of two stations (with the option of increasing the number of stations and instruments per station).

Rationale: It is sometimes necessary to draw correlations between different units at geographically distributed locations.

b) The physical packaging must be sufficiently compact to permit the unit to be easily moved around the facility

Rationale: The measurements are taken on as needed basis with the equipment temporarily installed in several locations within the facility.

c) The system must support the PCB Piezotronics Seismic ICP accelerometers (Model 393B31).

5. Design

5.1 Hardware

The sensors used in the CLS Building Vibration Monitoring System are Model 393B31 Seismic ICP Accelerometers made by PCB Piezotronics.

Fig. 1 shows the data acquisition electronics. Each station consists of a National Instruments PXI-1010 mainframe with 8 PXI slots and 4 SCXI slots. The PXI-1010 supports both multiplexed and parallel operating modes for SCXI. The mainframe houses a National Instruments SCXI-1531 8 channel accelerometer input module. The accelerometers are directly connected to this module with "short" cables, the length of which depends on the electrical noise in the area. The PXI side of the mainframe holds a National Instruments PXI-6052E 16 channel multifunction DAQ module with a sampling rate of 333 kS/s and 16 bits resolution. It is connected to the SCXI-1531 via an SCXI-1349 cable to enable parallel readout of all channels. More modules can be added in order to operate up to 32 accelerometers per station.

Each station is connected to the data acquisition computer via a National Instruments PXI-PCI 8335 MXI-3 link, using a fibre cable up to 200m long. One PXI-1010 will be equipped with rack mount slides and a handle and feet kit. It can be rack-mounted in a wheeled rack together with the computer, or it can be carried to a remote location. The second PXI-1010 is considered to be a remote station and will only be equipped with a handle and feet kit. More stations can be added as needed.

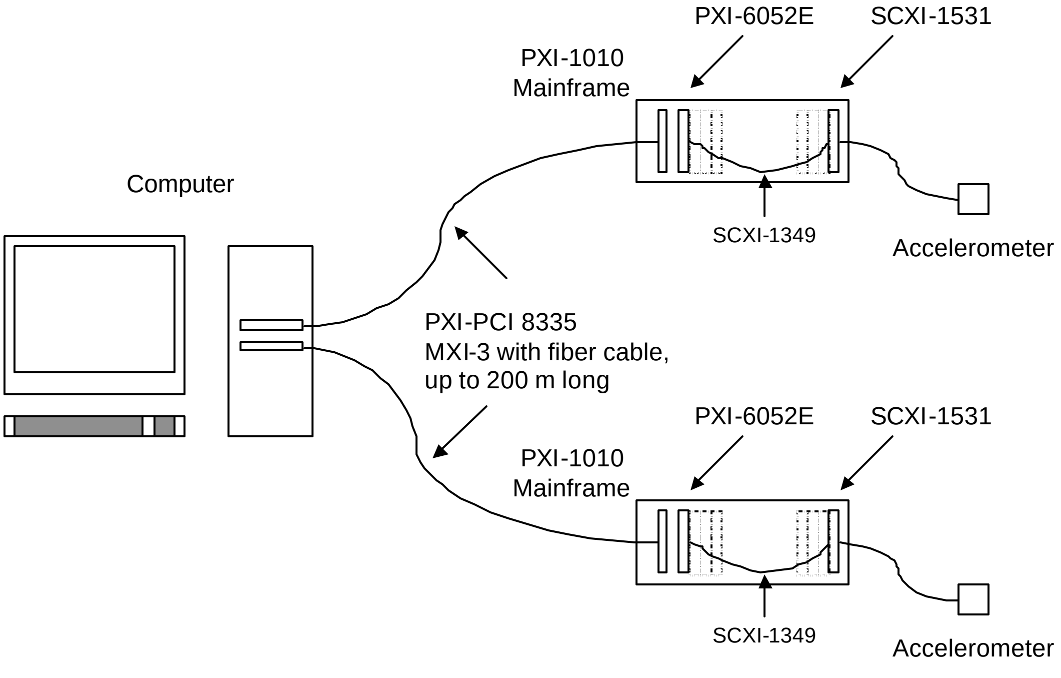


Fig.1: Data acquisition hardware for the vibration monitor system.

5.1.1 Piezoelectric Accelerometers

Piezoelectric accelerometers take advantage of the effect of stress applied to piezoelectric or polarized crystals (such as quartz). When a piezoelectric crystal experiences stress over the crystal's specified linear frequency range, an equal amount of charge of opposite polarity builds on opposite ends of the crystal. The crystal's capacitance tends to be very small, making the voltage sensitivity of the crystal very large.

$$\Delta V = \frac{\Delta Q}{C}$$

ΔQ - Charge that appears across the crystal

C -Capacitance of the crystal

ΔV -The resulting voltage across the crystal

The separation of charge or rather the voltage is proportional to the strain experienced by the crystal. A typical piezoelectric accelerometer contains a free-floating seismic mass that rests on top of a crystal. When a vibration wave stimulates the sensor, the seismic mass accelerates producing an applied dynamic force on the crystal proportional to the acceleration experienced and then this acceleration is proportional to the voltage that appears across the crystal. The voltages measured will only be dynamic voltages because the voltage produced by a static acceleration such as being in a gravitational field will tend to leak away through the path of least resistance. Due to the high-impedance of the crystal the resonant frequency tends to be very high compared to those of a mechanical accelerometer. The resonance frequency of the Model 393B31 tend to be in the order of 800 Hz. A high-impedance signal transmitted over a long co-axial cable in a noisy environment will be subject to a lot of interference. Within the sensor there is ICP circuitry that converts the high impedance charge to a low-impedance voltage. The low -impedance signal may be transmitted over long co-axial cable in a noisy environment. The ICP circuitry requires a constant current source that is provided by the signal-conditioning module. The bias voltage, typically 8-14V produced by the ICP amplifier may be used as a check to ensure the sensor is operating within spec.

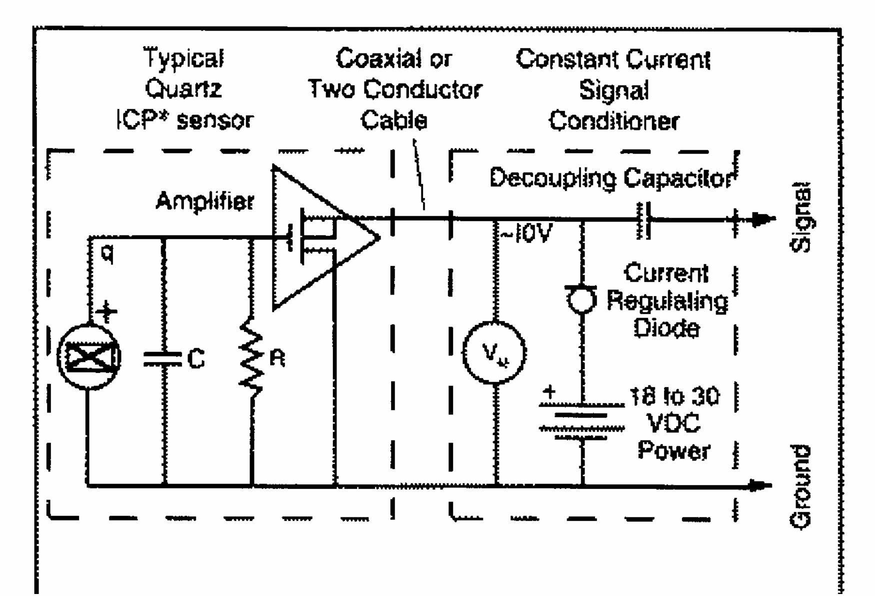


Figure 2. Accelerometer Schematic

The calibration sheets provided by the manufacturer specify the linear frequency range. The addition of mass to the accelerometer and or a change in the stiffness will alter the frequency response of the sensor for high frequencies. One must consider the accelerometer mounting technique used in taking a vibration measurement. At the CLS, the sensors will be screw mounted into mount points at specific locations at the site, a figure showing the current mount points is included in the appendix. Stud mounting tends to cause a sensitivity deviation far outside the linear frequency range of the accelerometer. The appendix contains a schematic of one of the mounting blocks used at the CLS to measure vibrations in the three orthogonal directions simultaneously. It is recommended that a thin layer of grease be used between the sensor and the contact surface to ensure transmissibility of high frequency vibrations. One must also be aware of the problems caused by mounting the accelerometer on electrically conductive surfaces or at points that are at a different electric potential relative to the signal-conditioning module. Electrical noise in the structure being measured may enter the path of the signal through the ground base of the accelerometer. One may insulate the accelerometer from the contact surface by placing a piece of paper between the two surfaces.

For low-impedance voltage signals over a short cable length the capacitive effects of the cable are not a concern. The cable of the accelerometer must also be fastened to the surface to avoid cable movement. Cable movement may result in electrical noise induced by the 'triboelectric' effect. It is also advised that the portion of the cable nearest the sensor be taped to the outside of the sensor to relieve possible connection breaks between the sensor and the cable connection.

The calibration sheets for the piezoelectric sensors are included in the appendix.

5.1.2 SCXI-1531 Signal Conditioning Module

The signal-conditioning module contains 8 programmable channels for analog input. A constant 4-mA current source option is available to each channel for current-excited accelerometers such as the model used at the CLS. The current source may be disabled to measure the noise within the system. An analog signal connected to a channel of the signal-conditioning module may be amplified by a gain of 1, 10, or 100 via the AC-coupled programmable amplifier. The signal is then passed through a 4-pole analog low-pass filter with programmable corner frequencies of 2.5 kHz, 5.0 kHz, 10 kHz, and 20 kHz. The analog filter is in place to ensure the signal is bandwidth limited and to reduce the aliasing of digitised data. The SCXI-1531 module also has simultaneous sample-and-hold circuitry to preserve the phase information and to ensure all the channels are sampled at the same time. Each input channel may be individually configured to single-ended or differential mode, to change the gains, or bandwidths.

Single-ended mode is recommended for signals that are greater than 1 V and the cable length for analog input signal is relatively short. Differential analog input uses two input channels and if the signal is floating with respect to ground it must be tied into the channels with bias resistors. Differential mode is recommended for analog input signals that are less than 1 V and over a long cable length.

One garners information from the analog signal by sampling the channels with the DAQ module. One may scan the channels in either parallel mode or multiplexed mode. The maximum scan rate in parallel mode with sample-and-hold turned on is 125 kS/s. It is the DAQ module or the SCXI module, whichever is lower, that limits the scan rate. The current configuration is for parallel mode; parallel mode is not limited by the settling time of the multiplexer and may make scanning more accurate and faster. When determining the rate at which all the channels are sampled simultaneously one must consider both the scan rate and the sample rate. Assuming a user is using n channels (n<8) then n samples are read in a single scan. The rate at which the channels are sampled leads to an inter-channel delay. The inter-channel delay is the reciprocal of the sampling rate.

$$\Delta t = \frac{1}{Sampling\_Rate} = \frac{1}{300000 Samples/s} = 3.3\times10^{-6} s/Sample$$

Δt - Inter-channel delay

If n channels are sampled then the overall delay is:

$$T = (n-1)(\Delta t) = (8-1)\,{}^{Samples}\!/_{Scan} \cdot (3.3\times10^{-6}\, s/Sample) = 2.31\times10^{-5}\, {}^{s}\!/_{Scan}$$

T-Overall Delay

n- number of channels sampled in a single scan

The overall delay must be less then the period of a single scan:

$$T < \frac{1}{Scan\_Rate} \Rightarrow 2.31\times10^{-5}\, {}^{s}\!/_{Scan} < \frac{1}{25000 Scans/s} \Rightarrow .0000231s < .00004s$$

If the scan rate used does not satisfy the above relationship then the software will report an error.

The SCXI module, as with all the hardware of the VMS, is software programmable and controlled by a data acquisition program written in C++. The device is calibrated in the factory and the calibration constants that correct for offset and gain errors are stored in the EEPROM.

The suggested minimum scan rate for a module is 10x the highest filter bandwidth setting (subject to the condition above). The recommendation of 10x the filter bandwidth is not a consideration due to the electronics but is a rule of thumb founded in the digital signal processing.

The specifications for the SCXI module are included in the Appendix.

5.1.3 DAQ 6052E Module

The DAQ module has 8 analog inputs with 16-bit resolution that may be sampled at a maximum rate of 333 kSamples/s. The module accepts two different types of signals, those that are referenced to the ground and those that are not. If a sensor does not have it's own ground reference (i.e. one that may be plugged into the building ground) then it is floating with respect to the ground. The accelerometers are floating with respect to ground so the module is run in referenced single ended mode, this ties the negative input of the PGIA to the analog input ground, the positive input is tied to the positive channel of the PGIA. The module may be run in differential mode using two analog input channels, one line that connects to the positive input of the PGIA and the other connects to the negative input of the PGIA. This mode allows for the elimination of common mode noise.

The module has two input ranges with respect to the reference voltage, a unipolar range of 0 to 10V (0 to V_{ref}) and a bipolar range of +/- 5 V ($-{}^{V_{ref}}\!/_{2}\ to\ {}^{V_{ref}}\!/_{2}$). The selection of the range depends on the type of measurement. With vibration sensors there will be both negative and positive accelerations corresponding to negative and positive voltages. The module has seven programmable gains to match the input signal range. The gain settings used depend on the range of the input signal used. To choose a gain setting one estimates the expected range of the input signal, selecting the gain setting that most closely matches this range to achieve the best resolution available.

The scan rate one programmatically selects for the module may not be the true scan rate. The scan rate depends on the resolution of the 20 MHz clock on the E-series board. 20MHz implies a 50ns resolution; to obtain the number of clock periods needed one must divide by the 'scan period' (reciprocal of the scan rate). The number of periods used must be an integer number so the resulting number is rounded up. This result, the actual period of a single scan is the integer number of pulses multiplied by the clock resolution for a specified scan rate. The actual scan rate is the reciprocal of the actual scan period.

For example:

$$ScanRate = 2545 Scan/s$$

$$ScanPeriod = {}^{1}\!/_{ScanRate} = \frac{1}{2545\,{}^{Scans}\!/_{s}} = 3.92927\times10^{-4}\ {}^{s}\!/_{scan}$$

$$ScanPeriod\!/_{ClockResolution} = \frac{3.92927\times10^{-4}\ {}^{s}\!/_{scan}}{50\times10^{-9}\ {}^{s}\!/_{pulse}} = 7858.4\ {}^{pulses}\!/_{scan}$$

$$\left\lceil 7858.4\ {}^{pulses}\!/_{scan} \right\rceil = 7859\ {}^{pulses}\!/_{scan}$$

$$ActualScanRate = \frac{1}{7859\ {}^{pulses}\!/_{scan}\cdot 50\times10^{-9}\ {}^{s}\!/_{pulse}} = \frac{1}{3.9295\times10^{-4}\ {}^{s}\!/_{scan}} = 2544.853\,{}^{Scans}\!/_{s}$$

5.2 Software

The system makes use of the following software:

(a) MS-Windows 2000 as the operating system.
(b) National Instruments MXI Interface configuration software is used to configure the MXI-3 interface to the remote PXI crates.
(c) National Instruments PXI and SCXI configuration software is used to perform internal calibration, configure properties of the conditioning module.
(d) Custom-written acquisition software is also used to log signal values.

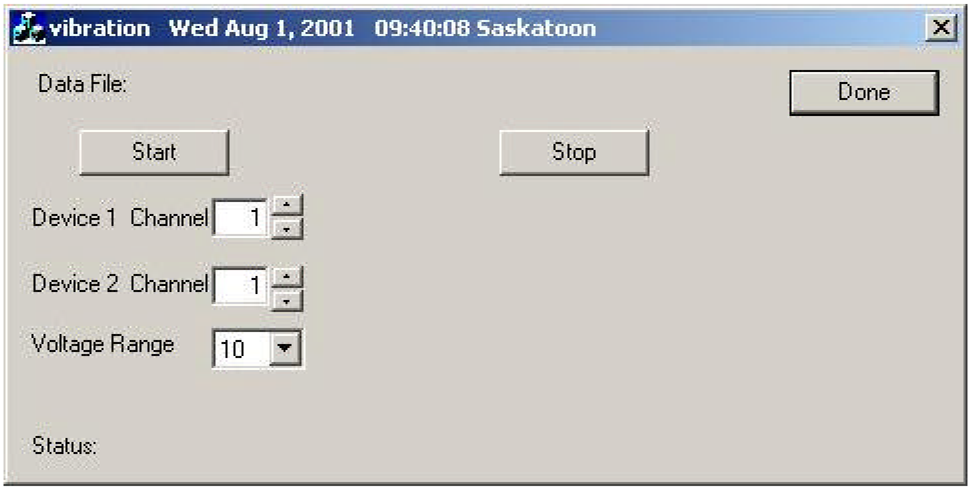
The custom-written acquisition software is written in C++ using the NI-DAQ driver libraries and Microsoft Visual C++ Studio. If a more complex window based user interface is required, future versions of the software may be written in LabView or alternatively switch to a Linux/EPICS based implementation. Initially, straight data acquisition will be performed with data logged in a format similar to what was used in the existing software. A future enhancement would be to incorporate a Fast Fourier Transform (FFT) algorithm (Press et al. 1992). In the short term, the use of the existing file format will permit the use of the pre-existing analysis and plotting software. The option to generate a space or table delimited output file that can be loaded into Math-CAD (or equivalent) will also be provided. A configuration file will be used to identify the channels that are to be sampled and the sampling period.

The computer has a 32 GByte hard-drive for data storage, a CD-ROM writer for data archival and a network connection to transfer sampled data onto the main CLS network.

5.2.1 Data Acquisition

The DAQ software configures, controls, and acquires data from the hardware is written in C++ and has a GUI interface for ease of use. A number of the programmable hardware features have been hard-coded into the software making the operation of the VMS transparent to the user. Given that the bandwidth of the accelerometers is .1Hz to 300Hz, 25000 S/s, is used to minimize the amount of data without producing aliasing. The sampling rate is hard-coded such that the overall inter-channel delay is less than the period of a single scan; the sample rate has been set as 300 kSamples/s. The input mode is set to referenced single-ended (meaning all measurements are made with respect to a common ground) and the input polarity is set to bipolar. The gains of the SCXI and DAQ module have been made adjustable from the GUI by allowing the user to select the voltage range expected from the input signal. For the first three ranges the DAQ gain is set to .5 and the SCXI gain is adjusted from 1, 10, 100. This corresponds to full range of 20 V, 2 V, .2 V. Again lowering the full range increases the resolution of the signal. The user may lower the range further, this is done in the software by holding the gain of the SCXI module at 100 and increasing the gain of the DAQ module to 50 or 100, corresponding to full voltage range of .02 V or .002 V respectively. It is not advised to increase resolution past the full range of .2V because the hardware is not guaranteed to operate within spec at lower ranges.

Fig. 3 shows an example of the user interface for the VMS system:



Data File:

Device 1 Channel 1

Device 2 Channel 1

Voltage Range 10

Status:

Figure 3: VMS Graphical User Interface

The device number corresponds to the number assigned to the chassis in the National Instruments Automation Explorer. The Channel widget is the number of channels used in a measurement. At present the software triggers on channel 0 and samples the channels consecutively. Therefore the channels used must begin at channel 0 and increase consecutively. The start and stop button correspond the beginning of a measurement and the end of a measurement.

Once the user presses the start button the hardware begins acquiring data by the parameters specified in the GUI. The flow chart shown in Fig. 4 summarizes the DAQ process.

The DAQ software uses a circular half buffer to store the sampled data. A data acquisition thread is started for each device. The data acquisition occurs in a loop that runs continuously. When the buffer is half-full a thread is begun that stores the half buffer to disk in binary format and then releases the half buffer to the DAQ loop. The data acquired in the loop is multiplexed in the sense that the first channel sampled corresponds to the first data point in the buffer, the second data sample corresponds to a sample from the second channel and so on. Before the half buffer stores the data to disk it first demultiplexes by changing the order of data points so the data in the buffer corresponds to all of the samples from the first channel, then all of the samples from the second channel and so on. This is the format that the data is printed to file in. It is obvious that the number of samples for a single channel output to disk in a single buffer emptying is the half the buffer size divided by the number of channels.

5.2.2 Data Formatting Software

A data file corresponding to a measurement output from the VMS contains the data from all of the channels on each device stored by successive dumps of the half-buffer and is in binary. It is desirable to have all of the data from a single channel stored in a single file in tab-delimited column text format. In-house software was written with a GUI interface to accommodate this need. The filename and directory are input into the GUI, the software takes the sampled data from each channel, creates a unique file for each channel, and stores it in the same directory as the input file. The unique file is named according to the original filename containing the sample number and day with an extension indicating the device and channel number. This format assumes that the data stored is a series of contiguous equally space sampled values of analog voltage.

For example:

H:\vibration\2001-10\21-000.dat

Refers to the first file (or the zeroth file represented by 000) created on the 21st day of the 10th month of the year 2001. If data in the file represented the sampling of one channel on device one and another two channels on device then the data formatting software would return three files.

H:\vibration\2001-10\21-000dev1chan0.txt

H:\vibration\2001-10\21-000dev2chan0.txt

H:\vibration\2001-10\21-000dev2chan1.txt

5.2.3 Digital Signal Processing Software

After a measurement is properly formatted then a data file corresponding to a single channel measurement is ready to be massaged by digital signal processing. In-house software was written to accomplish this task. The software divides the data from a single file into files corresponding to a minute of length and then processes the data as if each minute represented an isolated sample. The data is then converted by the accelerometer calibration constants, filtered, spectrum computed, and integrated twice. The acceleration and displacement data along with their corresponding spectra are stored to disk, four files are created each corresponding to a single minute of data for the sampling of a single channel. A more detailed discussion of the DSP process is given in section 7.

5.3 Measurement Capabilities of the VMS

The VMS is currently capable of acquiring simultaneous measurements from up to sixteen sensors. Eight sensors may be connected to the eight inputs available on each station by 1.8 m of co-axial cable. Therefore measurements may be taken about a small circle at each station, with a maximum distance of 200m between each station. The VMS may take measurements for a maximum of 24 hours from start time to stop time; this is limited by the hard drive space available. If required the disk space can be extended at a nominal cost

The upper bound magnitude and frequency limits of the VMS are constrained by the PCB accelerometers. In general the PCB sensors being used at the CLS measure accelerations with a frequency as high as 300 Hz. The calibrated sensitivity of the accelerometer determines the proportionality between acceleration and voltage. The VMS hardware in general is limited to a maximum input voltage of 10 V. The lower bound for frequency and magnitude of acceleration data is determined by the overall noise in the system. In terms of acceleration the lowest frequency that may be measured is .1 Hz and the lowest magnitude is $1mg$. When the data is integrated twice, noise in the low frequency range is greatly amplified and if it is the dominant signal in that range it must be suppressed. The characteristic noise of the system limits the high pass corner frequency to 1 Hz.

6. Operation Manual for Vibration Monitor

6.1 Overview:

This manual is intended for a user to obtain data of interest within the capabilities of the machine with ease. For information on specific hardware questions refer to National Instruments documentation for your device of interest.

6.2 Initialising the System

1. Ensure that all NI hardware components are connected and there is power to the system.
2. On each chassis move the PXI Power Switch and the SCXI Power Switch to the ON position. Each chassis must be connected to the computer via fibre optic cable and turned ON.
3. Open the panel to the computer chassis and move the Power Switch to the ON position.
4. Once Windows presents the log on screen enter the appropriate username and password for the system.
5. Connect the accelerometer(s) to be used to the BNC connections on the front of the SCXI module. Connections to the BNC posts of the SCXI module should begin at channel zero and increase in consecutively by channel number from channel zero.
6. On the windows desktop select MFC Short-Cut to Vibration icon and double-click.
7. The DAQ module and each accelerometer require 15min, and 30 seconds respectively for warm-up time to ensure accurate results. The minimum warm up time for each device may be exercised concurrently.
8. In the windows desktop select ‘Measurement and Automation’ icon and double click. Expand the ‘devices and interfaces’ branch. Select the chassis containing the device to be used and expand it’s branch. Right click on the SCXI module one will be using and select properties. Click on the ‘General’ tab and ensure under ‘Connected to’ the appropriate 6052E device is highlighted. Ensure in the box marked ‘This device will control the chassis’ has a check mark in it. Under ‘operating mode’ ensure that it is set to parallel mode. Click on the ‘Channel’ tab. Go through each channel to be used and ensure that the 4mA of current excitation is turned ON, the filter frequency is set to 2500 Hz, and the Ground Referencing is turned ON. After this is done press OK
9. Return to ‘devices and interfaces’ branch and select the 6052E device to be used. Once selected right click on it and select properties. Click on the AI tab and ensure that Polarity Range is set to –10 V to + 10 V and the Mode is set to Referenced Single ended. Click on the AO tab and ensure that the Polarity is set to Bipolar. Once this is done click on ‘Ok’.

6.3 Taking a Measurement:

1. Identify the range of values one is interested in. Typically one knows the stimulation that the device will receive to produce its respective voltage for that stimulation. One should identify the maximum voltage expected from the beginning of the measurement to the end of the measurement. Then adjust the voltage range of interest in the GUI that most closely matches the expected maximum voltage or is the closest value above the maximum voltage of interest. This allows for the best resolution within the limits of the hardware possible with out the clipping of one’s measurements. For example a user is interested in a series of vibrations stimulating a piezoelectric accelerometer to a maximum output .5 V. The range of interest should be adjusted to +/- 1V so that all values in the series of vibrations will be recorded using optimal resolution available.
2. If a station is not being used set the number of channels to 0 for that device.
3. For a station that is being used enter the number of channels being used in box for that particular device.
4. Press Start and the machine will begin taking data and recording it. Write down the filename for future reference, this filename belongs to the set of measurements that began when one presses start until one presses stop.
5. All data files are located on the D: drive of the vibration monitor computer each file being stored in a sub-directory referring to the year and month the data was taken. Each file name in the subdirectory consists of two parts. The first part refers to the day of the month the data was taken. Preceded by a hyphen the second part is an integer number referring to which consecutive measurement it is. For example D:\vibration data\2001-07\24-005.dat refers to the 6th file (zero based so the file 00 is the first file taken) taken on the 24th day of July 2001.
6. The output of the file contains samples of the analog signal measured, each separated by a time interval of equal size. The time interval depends on the scan rate hard coded into the software. The

discrete values are stored in binary and will be multiplexed if the number of channels simultaneously measured is greater than one.

6.4 Presentation of Processed Results

Processed data from a particular channel from a particular station is stored in a set of four unique files corresponding to the data sampled from that channel. The four files correspond to the acceleration data, displacement data, acceleration spectrum, and displacement spectrum. The number of files created for a single measurement is the total number of channels used multiplied by four.

Each filename contains the device, channel number, type of data (ie. Acceleration data), time-stamp, sampling rate, and re-sampling factor. For example:

20-001dev1chan0acceleration_0_min_R8.txt

20-001: second data file taken on the 20th day (stored in the directory dated by month and year)

dev1chan0: Voltages were read from Device 1 Channel 0

acceleration: Acceleration data

_0_min: First minute of data taken

R8: 8 samples were taken and averaged to produce one sample

7. Digital Signal Processing

A series of Q points is read into an array, the Q points represent the raw data collected from the VMS. The data is converted from units of voltage to units of acceleration by the calibration values given in the accelerometer calibration sheet:

$$a_n = C \cdot v_n$$
$$n = 0,1,2..(Q-1)$$

v_n - Discretely sampled voltage data

C - Accelerometer calibration constant

a_n - Discretely sampled acceleration data

The mean of the data set is determined and subtracted point by point from the data set. If there was a constant offset in the data that is not representative of the measurement but is caused by the system then this effect is removed. This is equivalent to removing the D.C. component from the DFT frequency spectrum adjusting the average value of the data set to zero. In terms of vibrations this is what is physically expected, if a vibration data set has a non-zero mean it would imply the object being measured is experiencing an overall acceleration:

$$s = \frac{\sum_{n=0}^{Q-1} a_n}{Q}$$

$$a_n \equiv a_n - s = \{a_0 - s, a_1 - s, ..., a_{Q-1} - s\}$$

s - Mean of the data set

The data set is then zero-padded 1:1 with N-1 zero's at the end of the data set, the zero padding is done in preparation for DFT frequency filtering:

$$a_n \equiv \begin{matrix} a_n & 0 \leq n \leq Q-1 \\ 0 & Q-1 \prec n \prec 2Q-1 \end{matrix}$$

$$n = 0,1,2,...,(2Q-1)$$

$$N = 2Q$$

The N point zero-padded data set is then transformed via DFT to N complex points in the frequency domain. The zero padding has no effect on the frequency values themselves but does allow for interpolation between frequencies. One could pad with as many zero's as one wanted to interpolate between discrete frequency points.

The DFT is given below:

$$H_j = \frac{1}{N}\sum_{n=0}^{N-1} a_n \cdot e^{-2pinj/N}$$

$$j = 0,1,2,...,N$$

The symbol DFT[] will be used to imply a mapping of N real data points in time space to N complex points in frequency space.

An FFT (FFT being an algorithm for performing a DFT) library is used to perform the harmonic analysis. The FFT software libraries used are MIT's Fastest Fourier Transform in the West (FFTw) library. The specific algorithm used computes a one-dimensional N point FFT that is un-normalized. To normalize the frequency values computed one must divide through by a scaling factor, the scaling factor used in this case is $1/N$. The resulting transformation is symmetric and one may discard half of the frequency values leaving $N/2$ points to consider.

The amplitude values are with respect to hertz and are given by:

$$f_j = \frac{ScanRate * j}{N}$$

By the Nyquist criterion the sampling rate must be twice as high as the highest frequency present in one's data to ensure there is at least 2 points per cycle sampled in the highest frequency. Assuming an even number of data points the so-called Nyquist frequency is at $f_{N/2}$. Frequencies above the lower half of the Nyquist interval will be aliased back in. By using a sufficiently high sampling rate aliasing will be reduced. The magnitude spectrum of the complex frequency values is calculated by computing the sum of the square real and imaginary of the jth complex value.

$$\left|H_j\right| = \sqrt{H^2{}_{real_j} + H^2{}_{imaginary_j}}$$

The phase is contained in the sequence:

$$J_j = \tan^{-1}\frac{H_{imaginary_j}}{H_{real_j}}$$

The phase is preserved in an array and the magnitude spectrum is multiplied by the following filter function:

$$\left|F_j\right| = \frac{1}{1+\left({}^{j}\!\!\diagup\!\!{}_{j_0} \right)^{2p}}$$

j_0 is defined as the corner frequency of the filter function. In the case of this filter function the corner frequency will always have an amplitude ½. The variable p adjusts the order of the filter causing the transition band to attenuate frequencies more rapidly. A value of p=8 is an 8th order filter that attenuates high frequencies rapidly with little aliasing in the transition band. The inverse Fourier Transform is defined IDFT[] and is given as:

$$a_n = \sum_{k=0}^{k=N-1} H_k e^{2pikn/N}$$

The data set must be zero padded 1:1 due to the discrete circular convolution implied in DFT frequency filtering as shown below:

$$DFT(a_n)\cdot DFT[f_b] = H_k \cdot F_k$$

$$IDFT[H_k \cdot F_k]_l = \left[\sum_{n=0}^{n=N-1} a_n f_{l-n}\right]$$

If the original data set was filtered circular convolution and one simply convolved the data sequence with a filter that has a finite impulse response of M coefficients, M-1 points of the original data set would be polluted with the transient response of the filter. If one pads the original data set by the length of the finite impulse response then the end points of the original data will not be polluted. Likewise with DFT multiplication one pads the end of the data set with zeros before transforming and multiplying, since the DFT transforms M points of data to M points one may consider DFT multiplication as circular convolution with the finite impulse response of the filter (the inverse DFT of the frequency response).

The IDFT of the DFT frequency filtered data is taken and the last half of the data set is removed, as it is the transient response of the filter.

The data is then integrated twice using Simpson's rule as a running integral, shown as a finite difference equation below:

$$y_n = y_{n-2} + \frac{\Delta t}{3}\left[x_n + 4x_{n-1} + x_{n-2}\right]$$

y_n - Numerically integrated data set

x_n - Input for the numerical integration rule

The frequency response of Simpson's rule only follows the frequency response of the ideal integrator to about .15*(Scan Rate).

The value of y at time n=1 demands a value for y at n=0, to satisfy this the first point of the integrated data is set artificially to zero. Setting this value to zero introduces a sharp discontinuity into the data; this is done for lack of an alternative. After a single integration the mean of the data is subtracted each time to ensure that the data has an overall zero mean.

Once the displacement data is obtained from the twice over integration the acceleration data is re-sampled by a factor of 8. This changes the original sampling rate from 25000 Scans/s to 3125 Scans/s and the Nyquist frequency is changed to 1562.5Hz. If one did not perform the previous low pass filtering then frequencies above the new Nyquist frequency would be aliased into the lower half of the Nyquist interval. The data is still well over-sampled by roughly 5x the corner frequency of the filter. Given that the original data set contained Q points where Q is equal to 1.5million, the re-sampled data set is reduced to 187500 points.

The DFT of the re-sampled acceleration data set is taken to determine the actual effect of the DFT frequency filtering. This is done because the ideal frequency response of the filter used is rarely the response that one gets. The re-sampled acceleration magnitude spectrum and the re-sampled acceleration data is then output to disk in tab-delimited format with respect to frequency and time.

The displacement data calculated must be high pass filtered to remove any data below the high pass corner frequency that may cause slow moving trends that are not representative of the signal one measures. The corner frequency chosen depends on the nature of the data and is best decided by trial and error. The magnitude spectrum N point DFT of the displacement data is computed and multiplied by the ideal high pass filter (the phase is preserved) as seen below:

$$D_j \equiv \begin{matrix} 0 & 0 \le j \le \left\lfloor \dfrac{f_{corner} \cdot N}{SampRate} \right\rfloor \\ D_j & \left\lfloor \dfrac{f_{corner} \cdot N}{SampRate} \right\rfloor \prec j \le N \end{matrix}$$

D_j - Filtered displacement magnitude spectrum

The high pass filtering will remove low frequency components below the corner frequency of the filter also removing the D.C. component. The ideal case is of course not realized and one will have ripples in the filtered portion corresponding to convolution with the sinc function. A filter with a smoother transition band was considered but when tested the result was a transition band containing enough energy to cause slow moving trends in the data. It is found that the density of the DFT points is sufficient enough that the high pass filtering approaches the ideal case sufficiently.

The IDFT of the filtered displacement spectrum is computed and re-sampled by a factor of 8, thus reducing the sampling rate and the Nyquist frequency. The DFT of the re-sampled data is computed, then the actual magnitude spectrum and displacement data are output to disk.

The interpretation of the DFT is critical when performing spectral analysis. The DFT may be thought of in two ways.

1. Values of the DFT may be interpreted as an approximation to the Fourier coefficients of the Fourier series for a single cycle of a periodic function.

2. Areas of samples of delta functions that are the real and imaginary parts of an approximation to the continuous Fourier transform of a function.

In the first case the DFT amplitudes of an acceleration data set will have units of [length/time^2]. In the second case the DFT amplitudes will have units of [length/ time^2/Hz].

Thus one may calculate either the Fourier Transform or the Fourier Coefficients depending on the scaling factor one multiplies the DFT by. Multiplying the DFT spectral estimates by $N\Delta t$ will yield samples of the Fourier transform.

If one wanted to view the magnitude spectrum of either the acceleration data or the displacement data in Amplitude rms simply multiplying through by the following:

$$\left|H_j\right|[Unit]\sqrt{2} = \left|H_j\right|[Unit_{RMS}] \Rightarrow j = 1,2,\ldots N/2$$

The single-sided power spectrum may be computed simply by squaring the single sided rms amplitude spectrum. The units in this case are volts amplitude squared.

Spectral averaging may be employed to decrease the random noise in a spectrum. The noise in the spectrum will tend to decrease with the square root of the number of averages.

If a non-integral number of cycles of a periodic signal exist in the data set then spectral leakage may occur, energy will pollute adjacent spectra. This is not a concern for non-stationary vibrations that tend to be transient, but stationary vibrations such as cooling fan will extend past the duration of a sample. Windowing may be used to decrease spectral leakage at the expense of resolution of the spectral lines. Although a window is not employed in the current DSP software if one were to be used the Hanning window is recommended. It is suitable for narrow band processes such as vibrations.

Future adjustments to the software may be to compute continuous sets of digitally processed data by using the overlap-and-save method. Specific time intervals could be obtained by employing a time indexing interface and using GNU plot as a front-end output.

8. Technical Problems Encountered:

8.1 Voltage Range of The DAQ Module:

The documentation on the bipolar range of the DAQ module is ambiguous. This was the first problem encountered with the module. The specifications of the module are given as +/- 5V in some places and documentation lists the voltage range as +/- 10 V in other places. In actual fact a bipolar range of +/- 10V may be achieved by setting the gain to .5. With this done any analog input signal with a range of +/- 10V is divided in two internally such that it falls into the +/- 5 V specs of the DAQ module and then is scaled back up to +/- 10 V at the end. This process is transparent to the user.

8.2 Simultaneous Sample & Hold Capabilities of the SCXI module:

The original interpretation of the documentation gave the SCXI device as not simultaneous sample-and-hold but that there was a delay between channel samples; phase is not preserved in this case. The device is capable of simultaneous sample-and-hold phase preservation and a proper interpretation of scanning and sampling facilitates the use of this capability. The SCXI module is scanned and in a single scan the channels specified for use are sampled. The rate at which these channels are sampled corresponds to the sampling rate specified in the software. In a single scan the voltages held internally across a capacitor do not change but will change in the next scan. Thus it is the scan rate that is the rate of consideration in the

DSP work, it determines the discrete time interval between each data sample for a channel. The sampling rate is simply the inter-channel delay. As long as the overall inter-channel delay is less than the scan delay then the simultaneous sample-and-hold capabilities will be properly used. Originally the scan interval was set to zero and the sampling rate was the assumed rate that is used to sample the analog signal. In this case the sample-and-hold capabilities are not being used and the phase is not preserved. Of course a scan rate of zero may be used for a single channel and the sampling rate may be set to what one desires but one will have to use a scan rate value subject to the above conditions preserving the phase for use of multiple channels.

8.3 Using Multiple Chassis

Originally the system would not initialise more than one chassis at a time. The current documentation does not address the problem. The solution to the problem is to turn on the chassis that is physically farthest from the main computer.

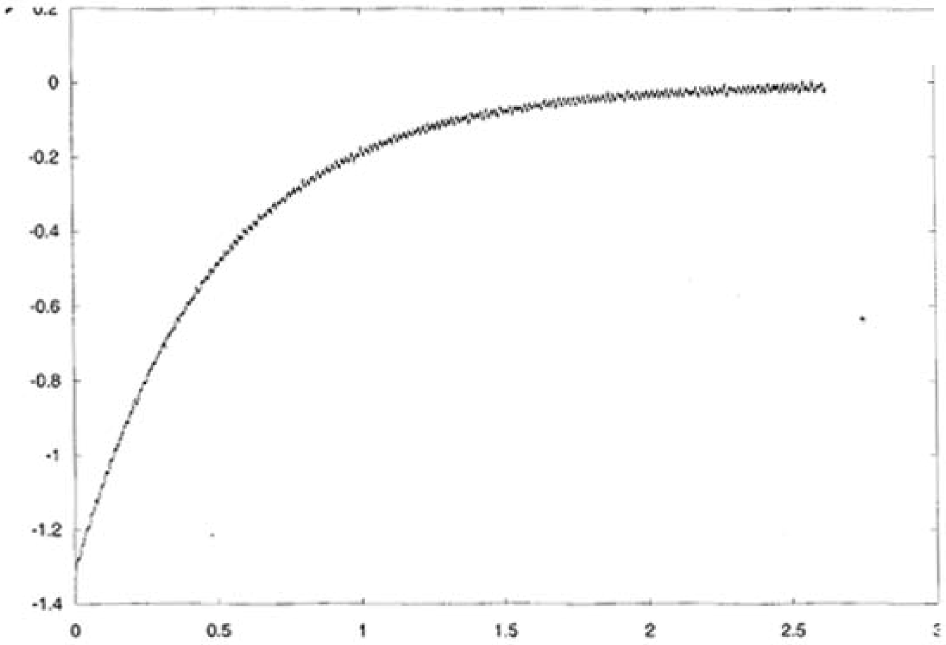
8.4 Low Frequency Integration

The output of the accelerometer is a voltage that is proportional to acceleration. To obtain displacement data the acceleration data must be integrated numerically twice. Integration at low frequencies tends amplify low frequency signals and attenuate high frequency signals. If noise is the dominant signal in the low frequency range then the data must be high pass filtered to eliminate the large contribution the noise would make to the integrated data.

8.5 Transient Voltage Decay

Transient voltage decay was identified when the gain settings were changed for a channel and a measurement was taken. At the beginning of a measurement exponential voltage decay was apparent. It was found that the voltage decay occurred when the gain settings of the amplifier were changed. The solution to the problem was to insert a ten second delay in the software at the beginning of each measurement to ensure that transients had dissipated and the system has settled. This is a short-term solution to a problem that warrants further analysis. If it is transient voltage decay the time constant of the event should be determined. In the interim it is recommended that two minutes is left between each measurement to ensure the system has settled. As can be seen in Fig. 6 the transient has decayed in about three seconds.

Figure 6. Transient Voltage Decay



8.6 Low Frequency Noise

Noise analysis has been done for the system and it is found that the noise level goes down by a factor of ten for each step wise increase in gain setting. Noise measurements of 1 min in length were taken simultaneously on all 8 channels of a device with each channel shorted. A channel short behaves the same as a 50-Ohm load and thus it is reasonable to use it in place of an accelerometer. Micrometer displacements are taken using the maximum resolution of the VMS so this discussion will be restricted to the noise amplitudes at this gain setting. If the noise were a random and Gaussian being uniformly distributed then the frequency spectrum of the noise should be entirely broadband. This is not the case for low frequency noise in the VMS. It is found that the low frequency noise is not a concern in the acceleration data but when integrated twice the process of integration tends to amplify low frequencies. Large amplitudes < 1Hz when integrated resulted in slow moving curves in the displacement data. It is inherently difficult to integrate data at low frequencies but it was expected that problems should occur at < .1Hz as opposed to < 1Hz. There are a number of possible sources of low frequency noise.

The low frequency noise may be an artifact of the numerical noise taking place in the large summations involved in integration. If this were the case, low pass filtering and re-sampling the data at a lower sampling rate would decrease the number of summations therefore decreasing the numerical error. Performing the aforementioned operation easily tests this. It was found that down sampling does not affect the low frequency problem.

The noise may be caused by a signal within the system such as a beat frequency caused by one of the three fans in the chassis or perhaps voltage leakage before the ADC. This may be tested by turning off one of the fans for a short enough period of time to take a measurement but not long enough to cause damage to the system. The hardware manufacturers advise against turning the fans off programmatically.

The noise may be due to a linear time varying offset in the system often called a baseline offset. If this is the case the baseline offset should appear as a large D.C. component in the frequency domain and energy about the D.C. component.

If the noise is simply due to the nature of the system and is random but not uniformly distributed then a potential solution is to increase the signal to noise ratio. The signal to noise ratio may be increased by averaging spectra. For example breaking up a 1-minute data set containing 187500 points into 100 segments each contains 1875 data points. If the noise is random, averaging their spectra will tend to reduce the noise by the square root of the number of segments averaged. This may allow for a lowering of the high pass filter corner frequency used to eliminate the slow moving curves brought on by low frequency amplification. This type of noise reduction is effective for measurements only concerning long term averages but is not useful for analysis of an identifiable transient event.

One solution to the problem is to high pass filter the displacement data using an ideal high pass filter having a corner frequency of 1 Hz. This removes all frequency content below 1 Hz and has the effect of decreasing the slow moving trends in the displacement data.

Low frequency content is apparent in the acceleration data in Fig. 7. This data set is treated as acceleration noise despite the fact that it is voltage noise. A calibration constant of 9.8 V/g is used for an order-of-magnitude calculation because it is roughly the calibration constant of each accelerometer. The data set has been down-sampled and properly-low-pass filtered to 300 Hz. The peak-to-peak amplitude of the noise falls within specifications of the data acquisition system but the specifications do not give any information about the frequency content.

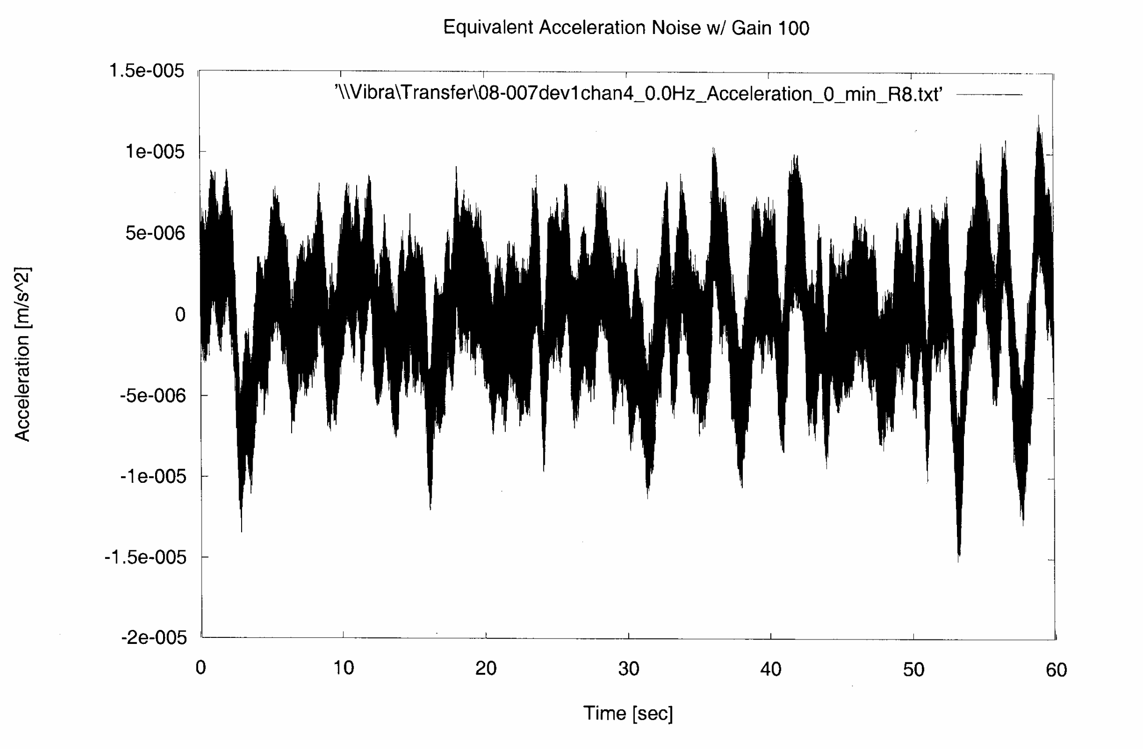


Figure 7. Equivalent Acceleration Noise w/ Gain 100

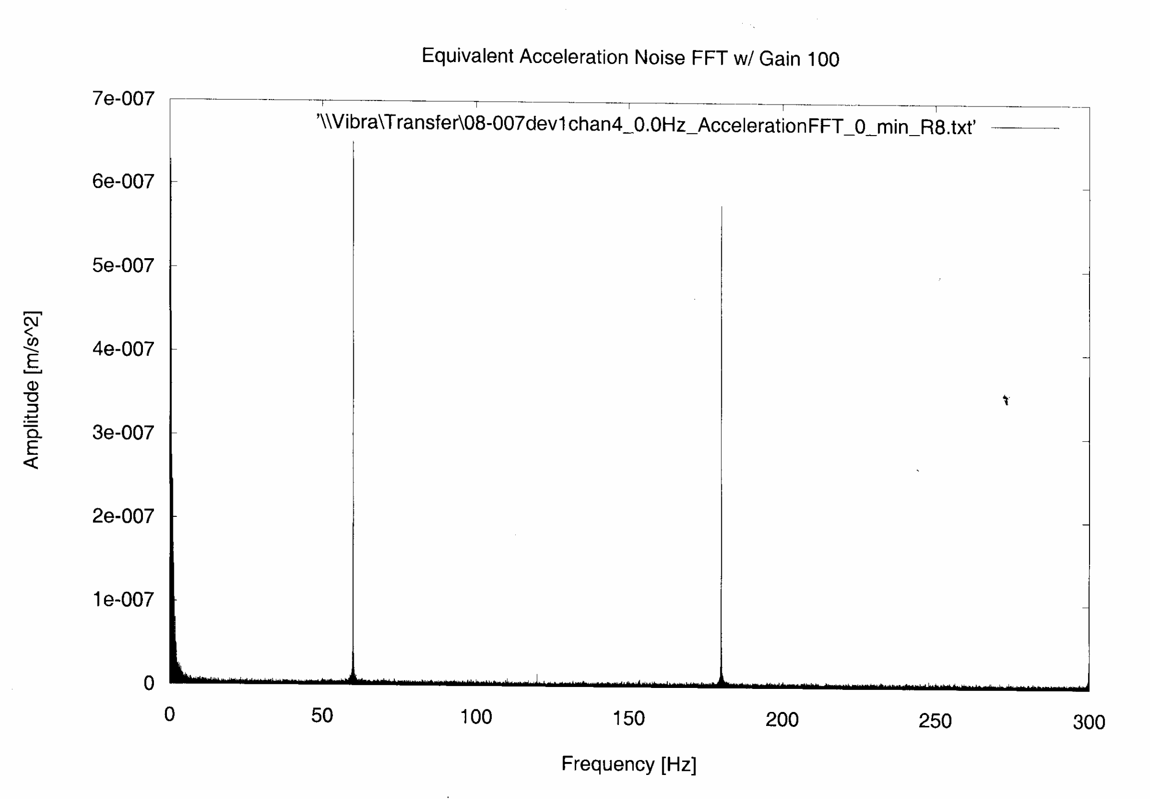


Figure. 8 Frequency Spectrum of Acceleration Noise

The spectrum of the acceleration noise shows strong electrical components at 60 Hz and again at 180 Hz. The frequency components may be due to the nature of the system or may be caused by grounding problems perhaps caused by a ground loop. If the electrical components in the spectra are due to a ground problem, then the grounding problem may also be the cause of the low frequency noise.

A closer inspection of the spectrum in Fig. 9 show frequencies between D.C. and 1 Hz shows large amplitudes compared to the broadband constant amplitude frequencies greater than 1 Hz. This is consistent with the low frequency discussion above.

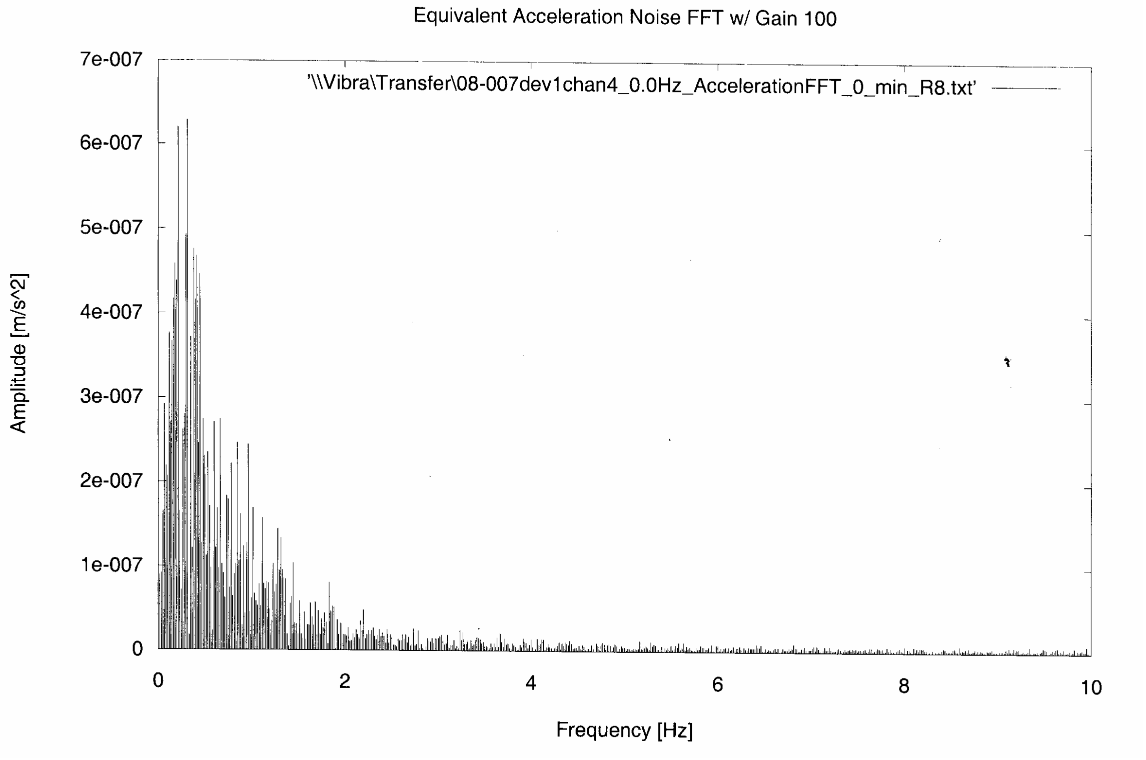


Figure 8. Lower Range of Frequency Acceleration Frequency Spectrum

Integration of the acceleration data by applying Simpson's rule shows slow moving curves that result from low frequency integration. This is seen in Fig. 10 showing the equivalent displacement noise.

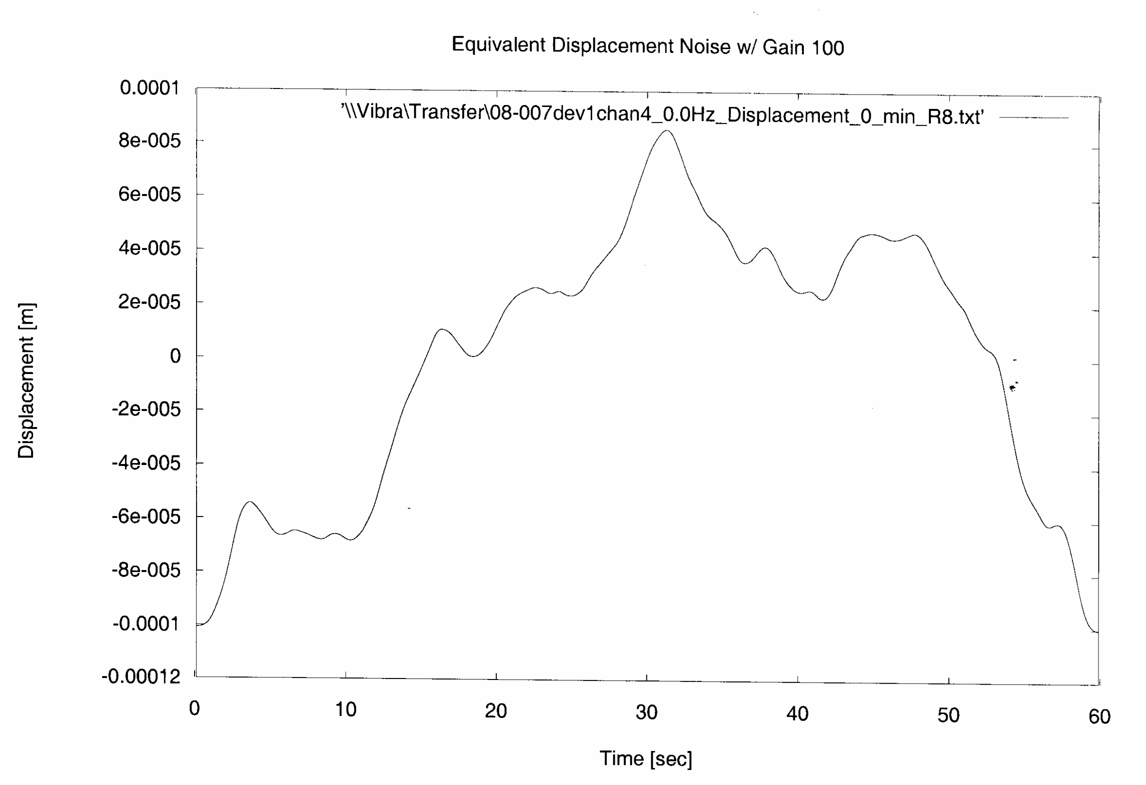


Figure 9. Equivalent Displacement Noise

The displacement spectrum in Fig. 11 shows the expected $1/w^2$ multiplication in frequency from the acceleration spectrum. The frequencies lower than 1 Hz are amplified and those above 1 Hz are attenuated.

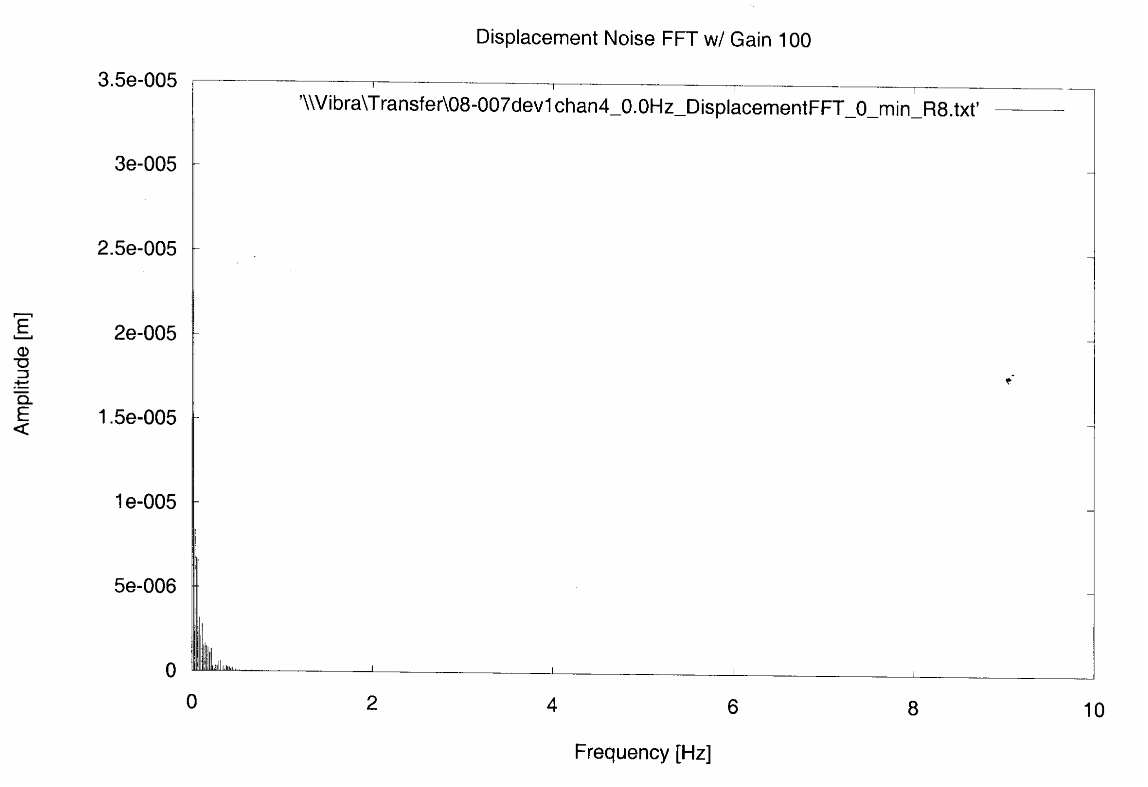


Figure 10. Spectrum of Displacement Noise

Closer inspection of the lower frequency interval in Fig 10. shows the D.C. component removed from the displacement data, this is consistent with a constant offset being removed from the data.

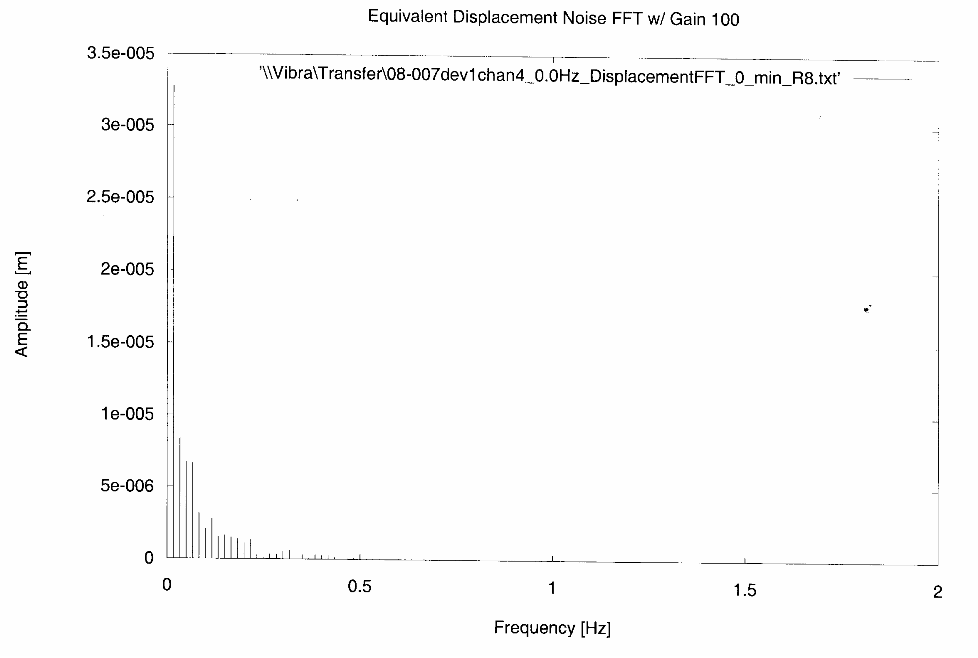


Figure 11. Lower Range of Displacement Noise Spectrum

High pass filtering will remove the slow moving curves allowing the higher frequency components present to be shown. High pass filtering with a corner frequency of 1 Hz lowers the noise amplitude well below the necessary $1mm$ resolution. This may be seen in Fig. 12 where the high pass filter has been applied to the displacement spectrum shown above.

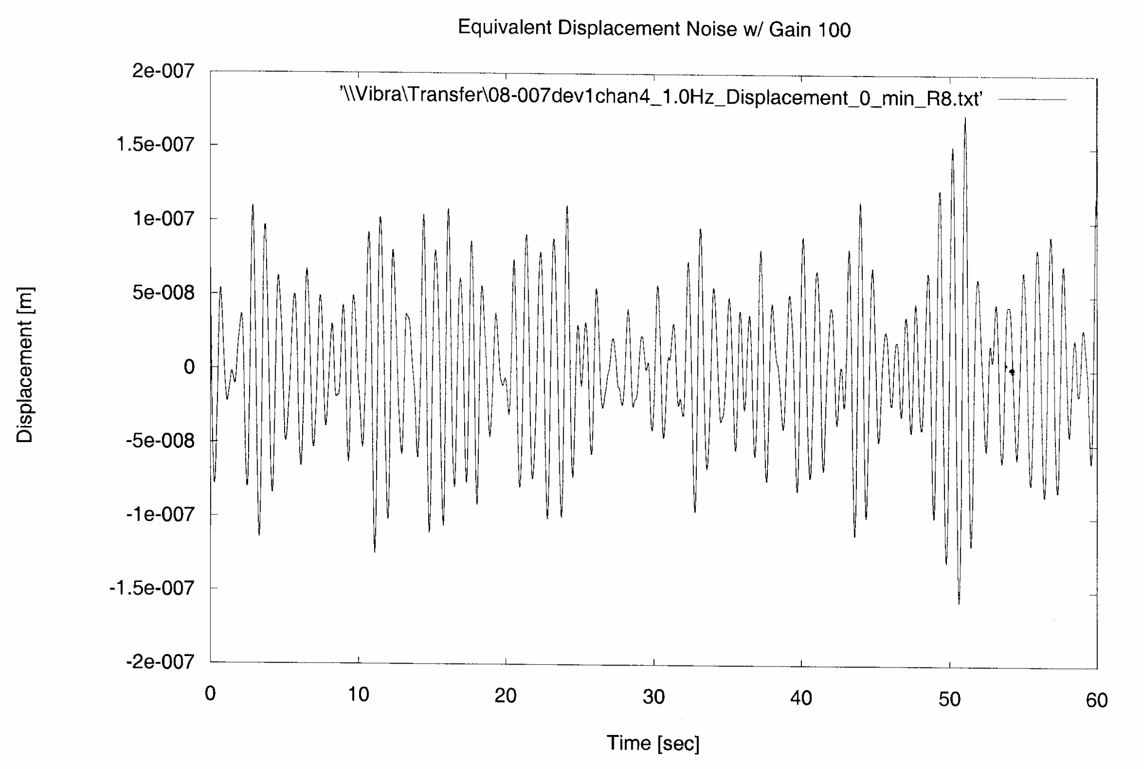
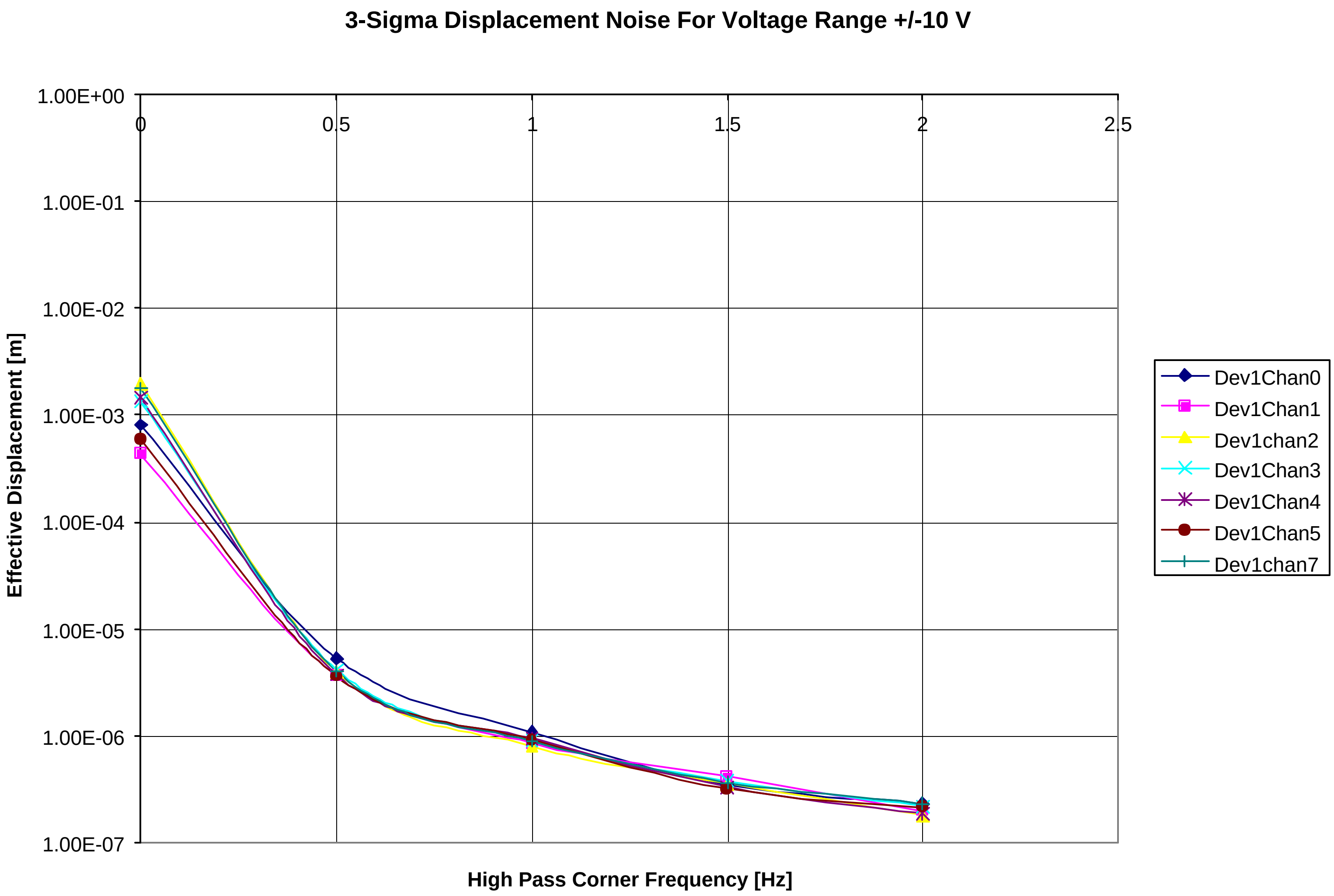


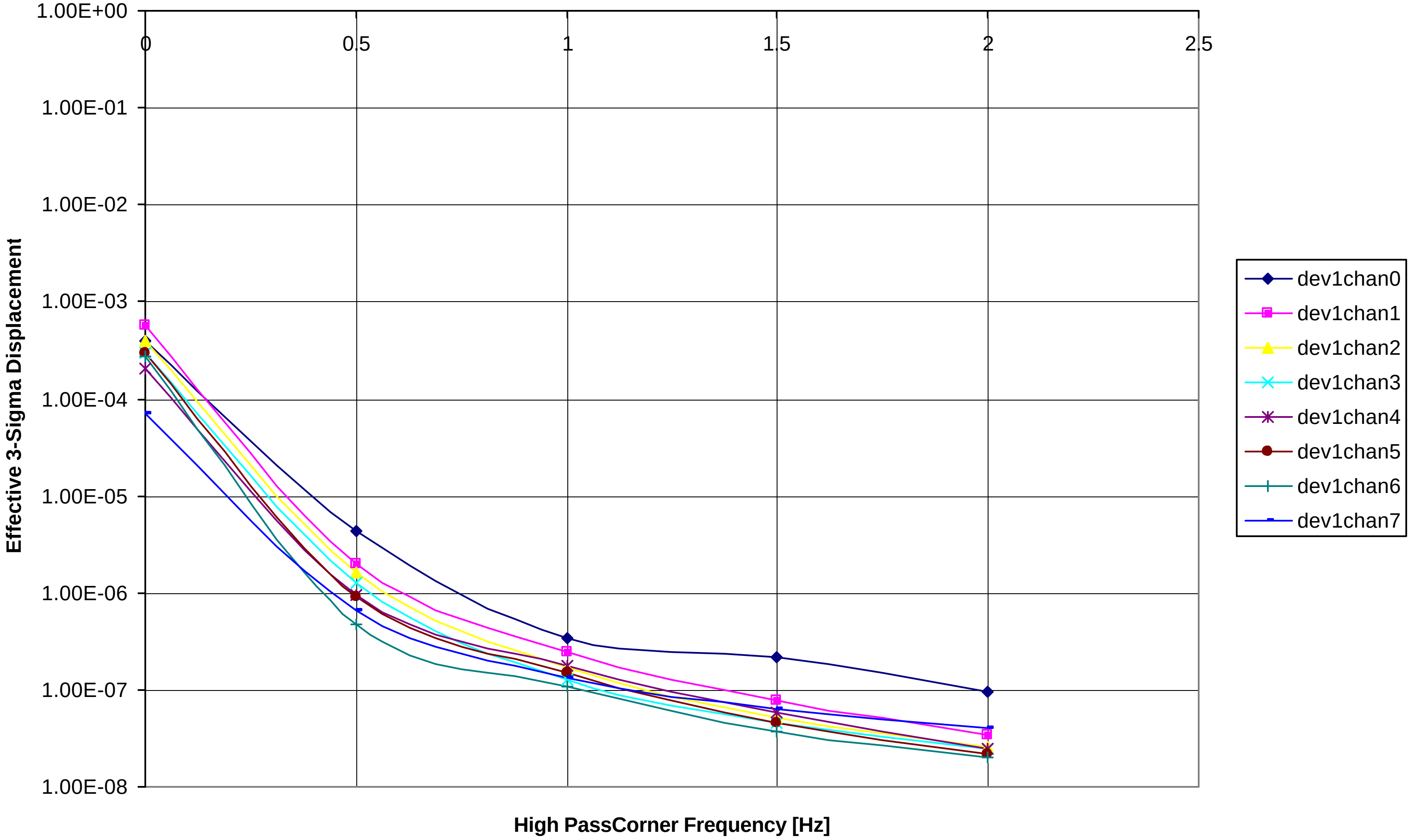
Figure 12. Displacement Noise Spectrum High Pass Filtered

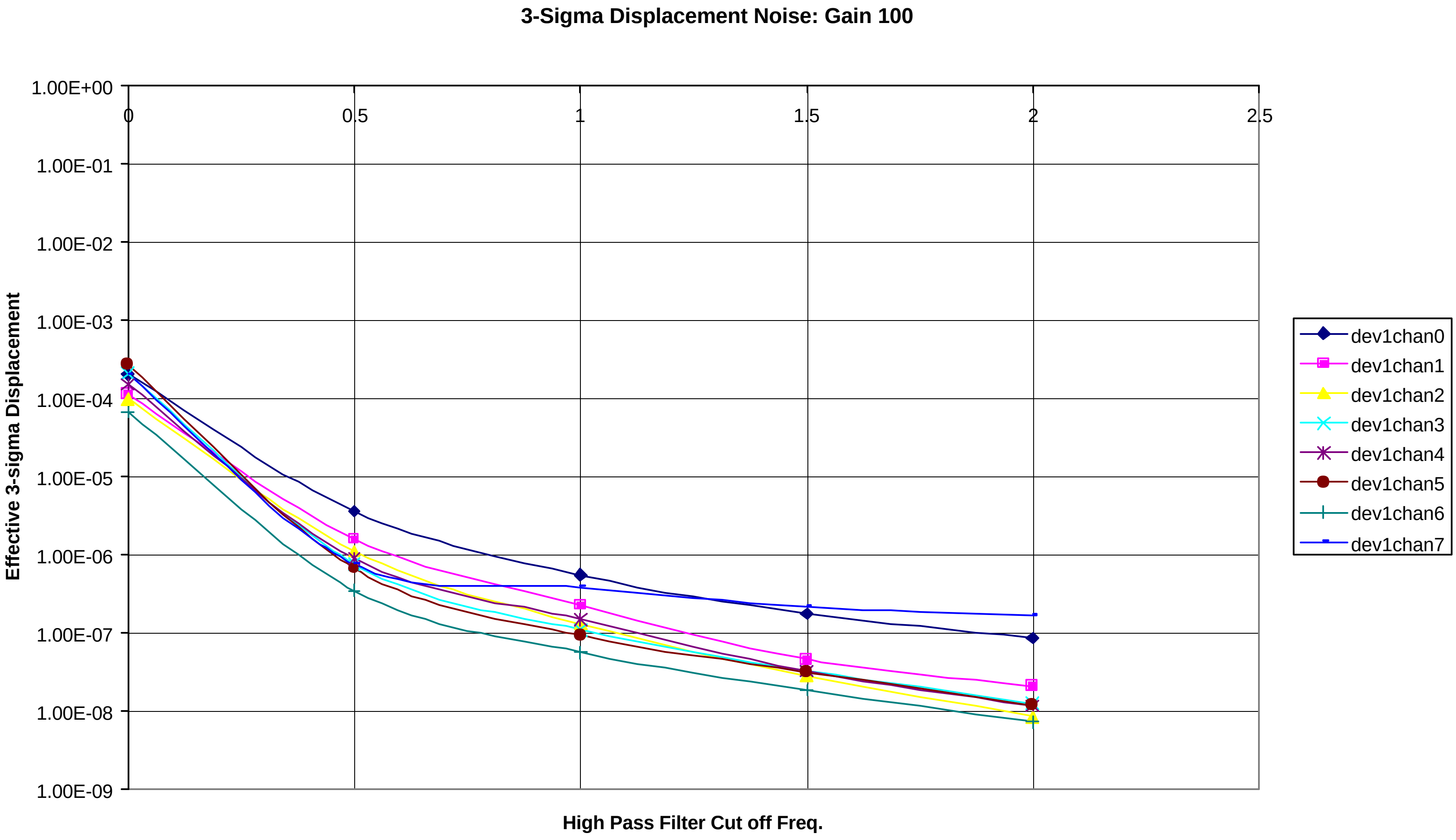
8.7 System Noise:

An analysis of the system noise was done using a voltage conversion factor of 9.8 V/g to obtain the effective displacement noise contributed by the system as a function of high pass filter corner frequency. Three sets of measurements were taken for the three gain settings of 1, 10, and 100 (DAQ gain set to .5). Each data set was roughly 60 seconds long sampled at a rate of 25000 Scans/s yielding about 1.5million data points. All the channels on each chassis were measured simultaneously and each data set was run through the DSP software for analysis. For a gain setting of 1 eight data sets were obtained for channels 0-7 each 1-minute in length. Each raw data set from a channel was run through the DSP software six times for six different high pass corner frequencies. The third standard deviation of each displacement data set is calculated to obtain the 3-sigma value for the displacement noise for that particular gain setting and channel. This yields for a particular gain setting 6 data points per channel and for 8 channels gives 48 data points in all. The results are summarized in three graphs below.


3-Sigma Displacement Noise For Voltage Range +/-10 V
Effective Displacement [m]
1.00E+00
1.00E-01
1.00E-02
1.00E-03
1.00E-04
1.00E-05
1.00E-06
1.00E-07
0
0.5
1
1.5
2
2.5
High Pass Corner Frequency [Hz]
Dev1Chan0
Dev1Chan1
Dev1chan2
Dev1Chan3
Dev1Chan4
Dev1Chan5
Dev1chan7

3-Sigma Displacement Noise: Gain 10





3-Sigma Displacement Noise: Gain 100
Effective 3-sigma Displacement
1.00E+00
1.00E-01
1.00E-02
1.00E-03
1.00E-04
1.00E-05
1.00E-06
1.00E-07
1.00E-08
1.00E-09
0
0.5
1
1.5
2
2.5
High Pass Filter Cut off Freq.
dev1chan0
dev1chan1
dev1chan2
dev1chan3
dev1chan4
dev1chan5
dev1chan6
dev1chan7

8.8 Summary of Measurement Capabilities of the VMS

The VMS is currently capable of measuring a total of 16 accelerometers distributed amongst two different locations simultaneously. With the use of a proper mount it is capable of measuring vibrations in the three orthogonal directions. It is capable of measuring dynamic displacements with a resolution of $1mm$ between 1 Hz and 300 Hz. If spectral averaging is used to decrease the signal to noise ration and thus decrease the lower bound of the frequency range then the sampling rate will change and the upper bound may have to be adjusted. The VMS is capable of addressing the CLS vibration measurement needs. The system may be expanded if the vibration monitoring requirements of the CLS changes.

Appendix A- Spec Sheet

1.1 PXI-1010 Mainframe

1.2 SCXI-1531 Accelerometer Module

1.3 PXI-6052E DAQ Module

1.4 PXI-PCI 8335 MXI-3 Fiber Optic Link

1.5 393B31 Seismic ICP Accelerometers

1.6 Hardware Test Procedure

The hardware set-up may be verified through the National Instruments Measurement & Automation Explorer. The software allows for verification of the system’s Chassis, DAQ module, and SCXI module.

1. Run the NI-MAX software:
2. Expand Devices & Interfaces
3. Right click on Each PXI-1010 Chassis and select test.
4. Similarly each module may be verified.

Table: 1 Module Verification	
Chassis (1)	Verified
Chassis (2)	Verified
6052E (1)	Verified
6052E (2)	Verified
1531 (1)	Verified
1531 (2)	Verified

1.6.1 Accelerometer Test Procedure

The input of the accelerometers that are connected to the SCXI-1531 module on the system may be tested and viewed in NI-MAX. Selecting the 6052E module that is controlling the particular SCXI module to which the accelerometer(s) are connected may do this. Right clicking on the 6052E module and selecting ‘test’ will bring up a window that allows for viewing of the signal.

Both the ADC & the Accelerometer frequency calibration may be tested by using tuning forks calibrated to an accuracy of +/- .01% of full scale. By taking an FFT of the vibration measurement one may identify the peaks at the frequencies of the tuning forks. Since the tuning fork oscillations tend to damp quickly one must make the measurement time short, say ten seconds, so one may resolve the tuning fork peak against the background noise. Assuming the background noise is small a large peak should be identifiable at the tuning fork frequency. If there were an integer number of oscillations in the total sample period then there would be a single sharp peak in the frequency domain. If this is not the case then the frequency peak may fall between two discrete frequency points and it’s energy will leak to other frequencies about that point. Identification of the centre frequency is the primary concern. A pedantic approach may be to zero pad the data such that the frequency in question falls precisely on a discrete frequency point therefore reducing the energy leakage.

If the peak is offset from the expected frequency it may either be due to the clock frequency of the ADC or it may be due to a damaged accelerometer. If it is due to a damaged transducer then one may test the accelerometer in question against a calibrated accelerometer to identify whether it is in fact of the accelerometer. If it is the ADC the user is reminded the sampling frequency input to the system may not be the resulting sampling frequency. Refer to a 5.1.3 on the actual sampling rate of the ADC.

The accelerometers may be tested themselves by using a T-Connector on the input channel of the accelerometer and using an oscilloscope or multi-meter to measure the bias voltage of the accelerometer.

Table 2: Accelerometer Calibration Tests

Acc. SN#	Spec. Bias (V)	Measured Bias (V)	Tuning Fork Freq. Hz	Measured Freq. [+/- .1 Hz]
8767	11.90	10.99	128	128
8780	11.40	10.80	128	128
8781	12.20	11.32	128	128
8777	11.80	10.84	128	128
8763	9.93	10.85	128	128
8768	10.07	11.20	128	128.25
6606	10.30	9.97	128	128
5847	12.00	11.21	128	128
5848	9.80	10.50	128	128

Fig. 13 shows the expected peak at the tuning fork frequency. The is most sharp at 128 Hz and has other typical features of digitised data such as spectral leakage. Fig. 14 is a close-up of that the same peak.

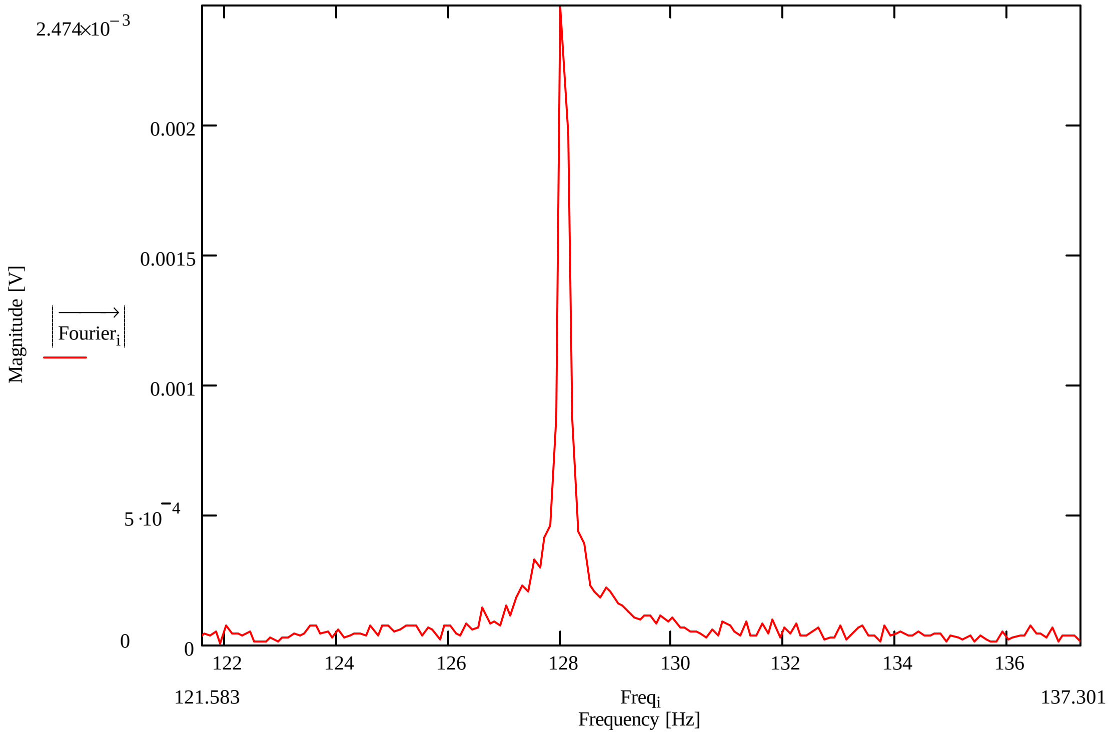


Figure 13 Example of Accelerometer 8767 Frequency Calibration

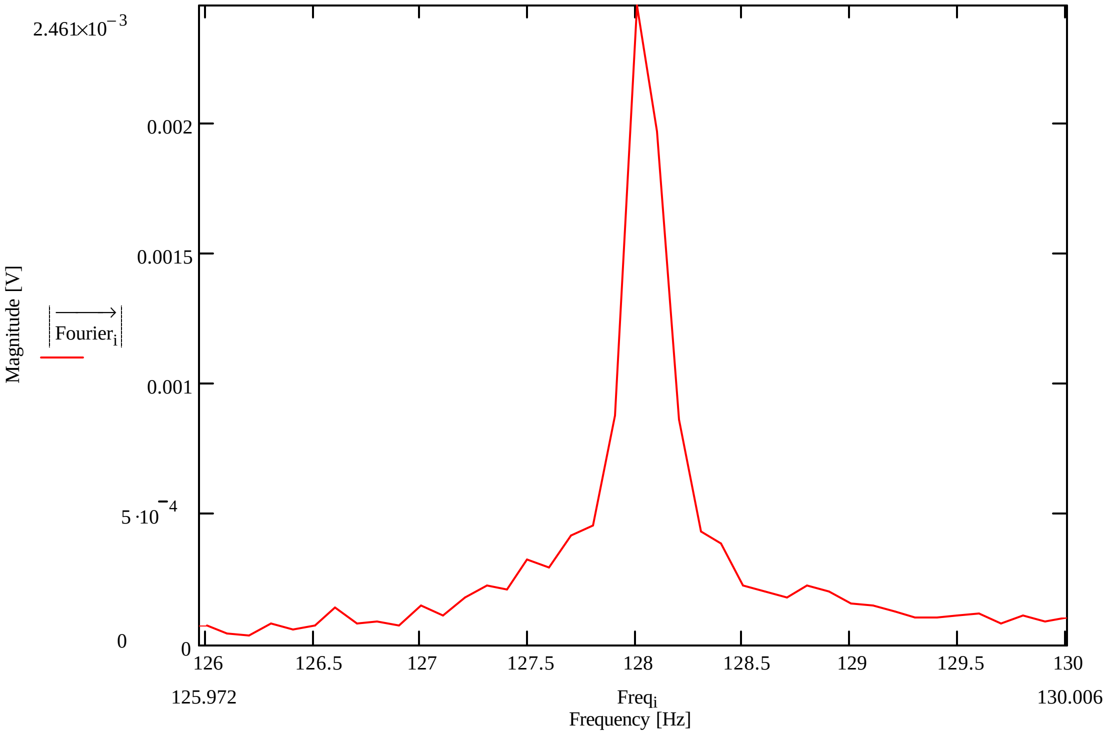


Figure 14 Higher Resolution Example of Accelerometer 8767 Frequency Calibration

Fig. 15 is the displaced peak of accelerometer 8768. The peak is measured to be consistently high by roughly .25 Hz for two different tuning forks of the same resonant frequency of 128 Hz. This feature may be due to a damaged crystal.

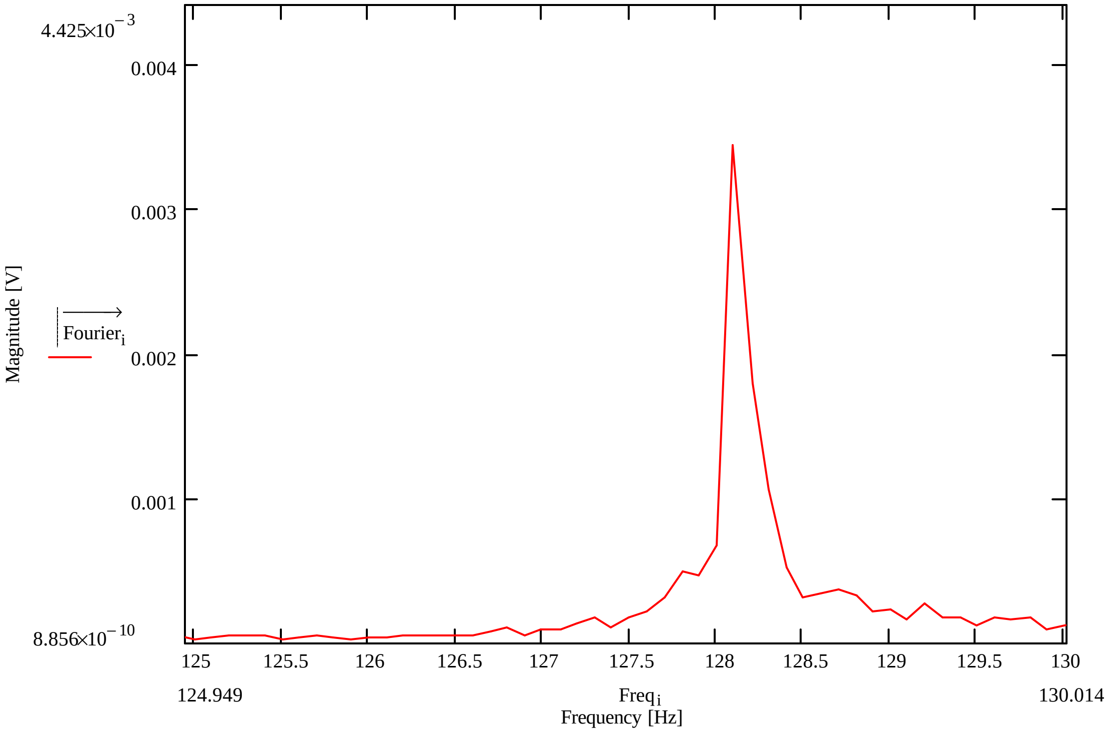
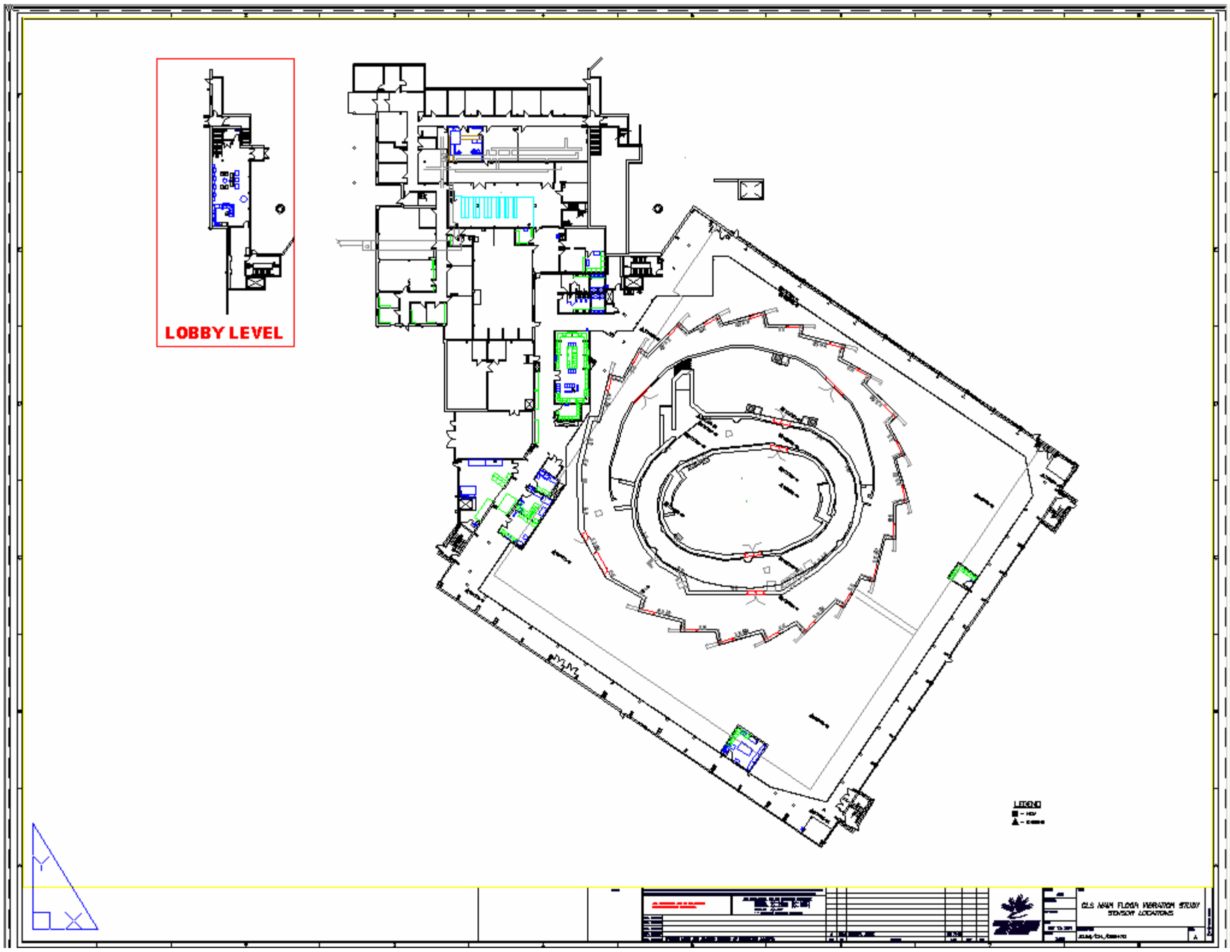
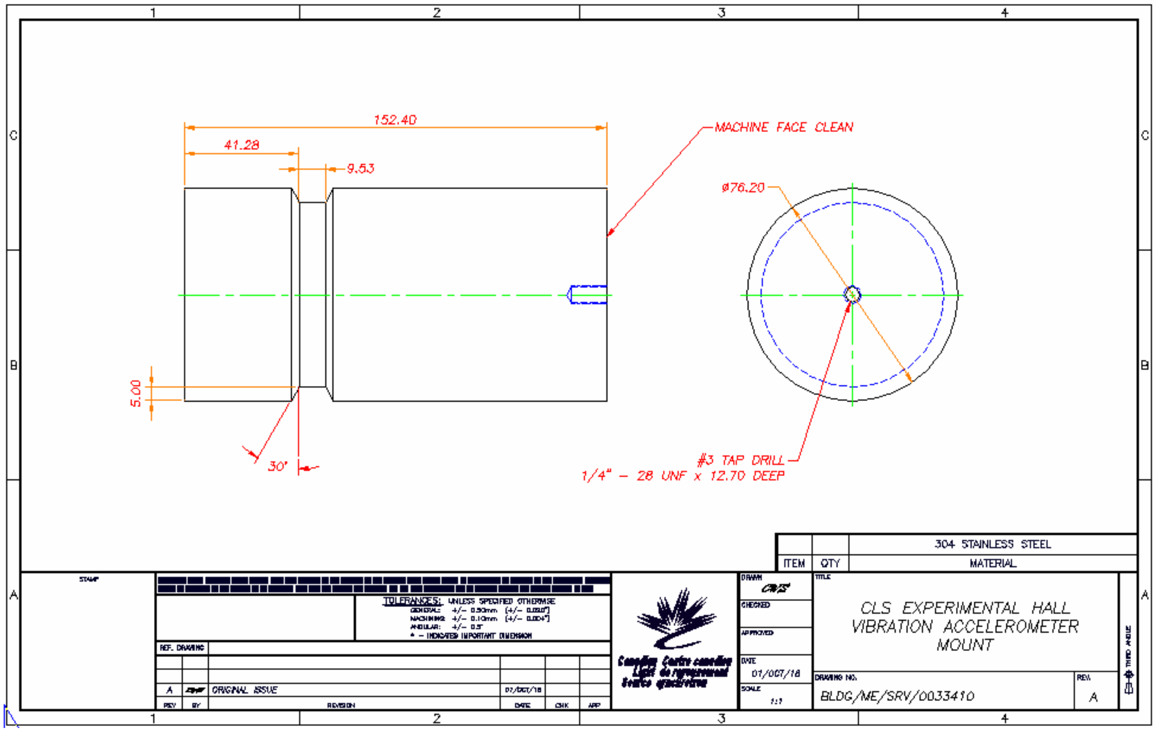


Figure 15 Displaced Peak of Accelerometer 8768

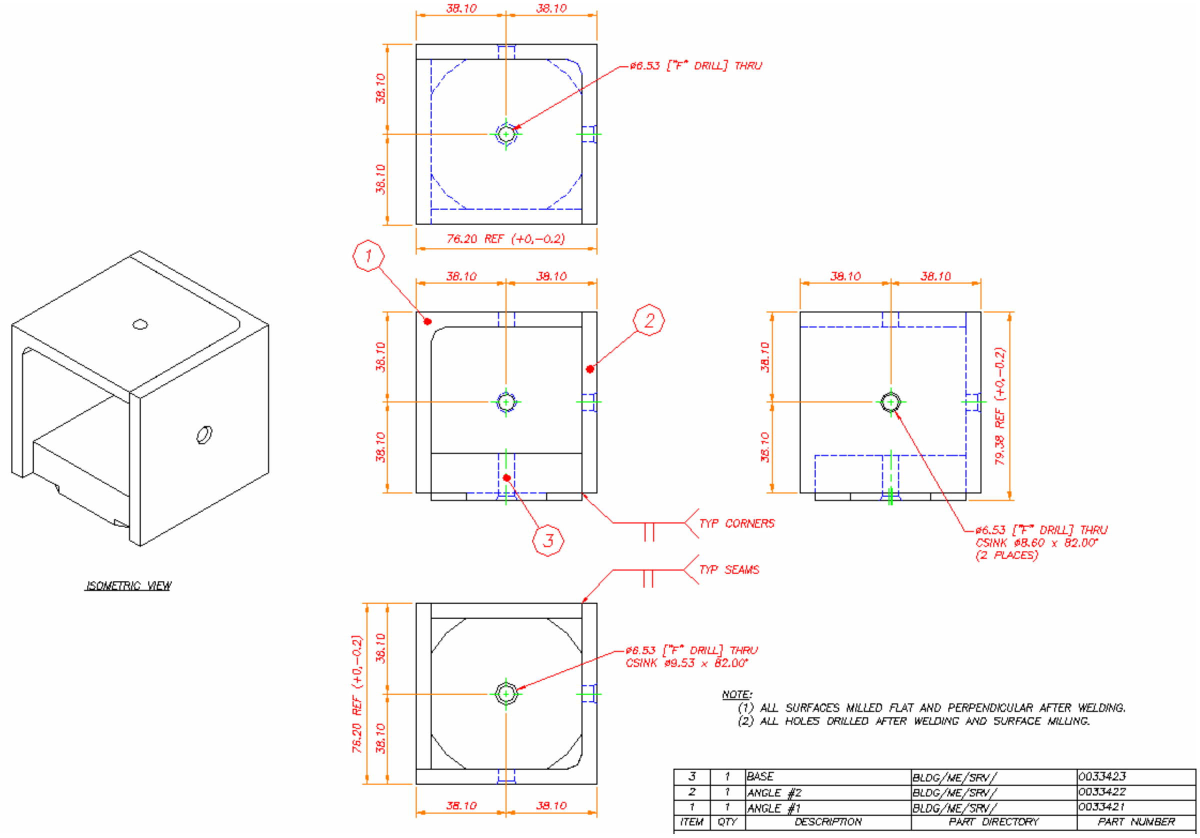
1.7 CLS Accelerometer Mount Point Locations



1.8 Accelerometer Mount



1.9 Accelerometer Block Mount



NOTE:
(1) ALL SURFACES MILLED FLAT AND PERPENDICULAR AFTER WELDING.
(2) ALL HOLES DRILLED AFTER WELDING AND SURFACE MILLING.

ITEM	QTY	DESCRIPTION	PART DIRECTORY	PART NUMBER
3	1	BASE	BLDG/ME/SRV/	0033423
2	1	ANGLE #2	BLDG/ME/SRV/	0033422
1	1	ANGLE #1	BLDG/ME/SRV/	0033421