

POLITECNICO DI TORINO

Department of Control and Computer Engineering

Master degree in MECHATRONIC ENGINEERING

THESIS:

Development of control system in LabVIEW environment for the study of
the random fatigue behavior of metallic materials



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1 Introduction

Nowadays the huge application of alloy materials in different mechanical system leads to known how characteristics of different metal alloy responds to random vibration excitation. During the years it was of great interest the technique based on modal parameters measurement of material and study the changing of characteristics related to fatigue stress test.

The aim of the thesis is to develop a control system, in NI LabVIEW environment, for fatigue testing of alloy materials on a test bench with electromechanical actuation system called Modal Shaker. The functionalities of the software include generation of sinusoidal signal controlling the pneumatic actuator and acquire signal coming from sensors collocated on the alloy specimen using real time device. Consequently, the software does a modal frequency analysis according to the data provided by accelerometer with the aim to find different modal frequency of specimen considered. Then it is possible to generate a stress signal on it to simulate the fatigue life and evaluate changing on physical alloy characteristic.

2 State of art

The main objective of this thesis work is centered on the development of a software to let examine the characteristic of various material. This is related to a field of applied mechanics, called 'structural mechanics', which investigate about deformations, stresses, and strains in solid materials. Often, the purpose is to determine the strength of a structure, such as a bridge, to prevent damage or accidents. On the other hand, structural mechanics analyses include determining the flexibility of a structure and computing dynamic properties, such as natural frequencies and responses to time-dependent loads.

After these considerations is possible to study fatigue phenomena of material, it depends by many factors statistically determined such as type of stress, number of cycles, stress amplitude, mean stress that material undergoes. Fatigue is considered the most common cause of structural failures in mechanical system application. Among the functionalities of the requested software, is crucial to consider the type of stress interested the specimen white noise generation signal provided to Modal shaker.

Although in literature different control system to study material characteristics are proposed, there is no commercial software be able to implement the functionalities requested considering the hardware system type, this justify the necessity to develop entirely a new software control system to implement on the given hardware. The choice of LabVIEW as environment is related to possibility of employ different hardware devices provided by National Instrument that are well integrated and suitable with sensors used final aim of this study case.

In details, there are advantages in term of good management of real time generation of signal and acquisition processes that is well integrated and supported among functionalities of LabVIEW environment.

The mechatronic system used to perform the test involve the know of many concepts of signal theory and structural mechanics. According to this consideration in the next chapter a brief theoretical explanation of the analyzed phenomena is done.

2.1 Signal theory

The signal theory studies the mathematical and statistical properties of the signal, describing how it can evolve with space time domain through mathematical functions. In this section are introduce some fundamental concepts useful to have a clear vision of observed phenomena of vibration and justify the decisions take in software development.

2.1.1 Statistics

Some consideration on statistical parameters is necessary to have good start point of analysis of signal involved in the application of signal generation part of the software.

A signal is a time variation of physical state of a system or physical quantity used to transmit some information between two different point in the space. The signal it is represented by a time definite mathematical function of which it is possible to evaluate some characteristics: the mean value, its variance, root mean square value and autocorrelation function of a signal.

The mean value of a function is given by:

$$\mu_x = \lim_{T \rightarrow \infty} \int_{-T/2}^{T/2} x(t) dt$$

The variance of a function is given by:

$$\sigma_x^2 = \lim_{T \rightarrow \infty} \int_{-T/2}^{T/2} [x(t) - \mu_x]^2 dt$$

The positive square root of the variance is known as standard deviation σ_x , and it is a measurement of how spread from the mean values of the function are.

The root mean square value of a function $x(t)$, is defined as:

$$x_{RMS} = \lim_{T \rightarrow \infty} \sqrt{\frac{1}{T} \int_{-T/2}^{T/2} [x(t)]^2 dt}$$

The autocorrelation function of a signal is:

$$R_x(t_1, t_1 + \tau) = \lim_{T \rightarrow \infty} \int_{-T/2}^{T/2} x(t)x(t + \tau) dt$$

The autocorrelation of a function describes the general dependency of $x(t+\tau)$ with its value at an instant before, $x(t)$. It's a parameter that represents the “memory” of that function, as far as a future value is strongly or weakly related to the previous ones.

2.1.2 Numerical processing of signals or Digital Signal Processing (DSP)

Introduction about how a real continuous time signal is often processed it is necessary to clarify some aspect in terms of computer signal elaboration. Indeed, any signal processed by computer is a discrete representation of a continuous time one and as results of this it is necessary to introduce some concepts about discretization of continuous time signal.

Sampling operation allows to obtain a discrete time signal, that is a sequence $x[n]$ of numbers which can be represented by a relative integer variable having real or complex values, starting from the analog signal. Any generic signal $x(t)$ is sampled when an extraction of the value that it assumes in equispaced time instant is done, the time interval considered T is called the sampling period. This operation creates a sequence whose n -th

values $x[n]$ is the value assumed by the continuous-time signal at instant nt : $x[n] = x(nT)$

The sampling operation is symbolically carried out by a device, the sampler, indicated with a sort of switch that closes for an infinitesimal duration interval. The rate at which the switch closes is equal to:

$$f_c = \frac{1}{T}$$

and takes the name sampling frequency, measured in Hz or in samples / second. Figure shows a generic analogue signal as a function of time t , it can be seen how it varies with continuity. The ordinate represents the signal level, which for acoustic signals is usually expressed in decibels (dB).

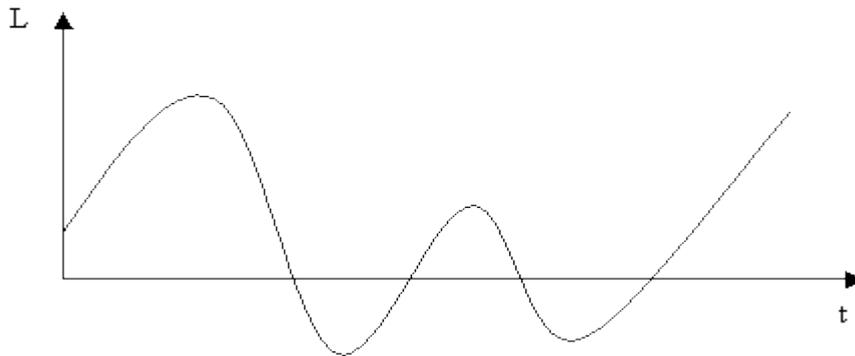


Figure 1: Analog signal

The Figure represents the digital counterpart of the previous analogue signal: in this case it is clearly seen that we are dealing with a discrete sequence of values and no longer with a continuum, with the values of the sequence that are distant from each other a time T equal to the period of sampling.

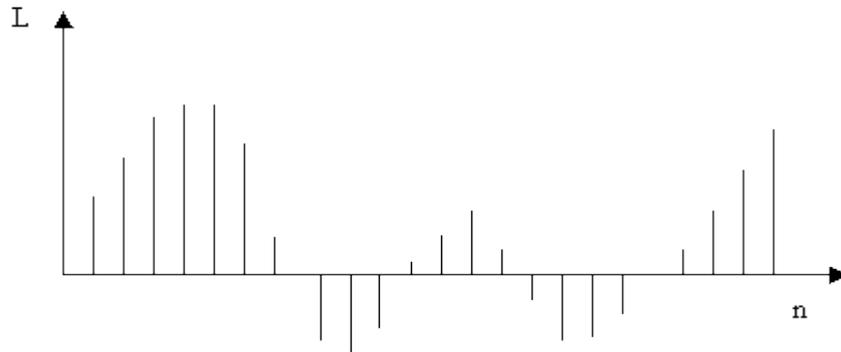


Figure 2: Digital signal

Usually, the discretization process is executed by A/D converter. This device is controlled by a clock signal (timing) at the sampling frequency. The ideal sampler extracts at each clock pulse the value of the input signal at the sampling instant, which is in general a real number with infinite decimal digits. Unlike the ideal sampler, the A / D converter instead makes a finite representation of this real number (digital signal), and precisely in binary arithmetic on a finite number of digits (bits), variable from 8 to 24. The small error inherent in the representation of the real number on a finite number of digits represents the second operation of the converter: quantization. In fact, therefore, the converter carries out a double "discretization" operation: the first on the time axis, where it determines the instants in which the signal must be considered; the second on the amplitude axis where the signal is approximated to the nearest "how much" based on the converter's bit resolution.

From Shannon's Theorem (or Nyquist) it is possible to obtain the following relation:

$$f_c \geq 2f_{max}$$

Where f_{max} represent the maximum frequency reached by signal, also called as Nyquist frequency, while f_c is the sampled frequency. This is so that a signal after sampling can

again be converted back to analog. For example, a signal with frequencies up to 20000 Hz must be sampled with a frequency equal to or greater than 40000 Hz

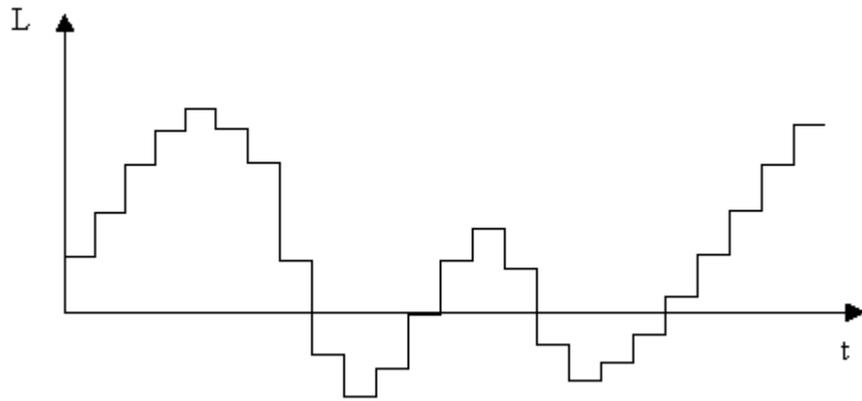


Figure 3: Sampled signal

The width of the steps of the sampled signal (Figure 4) is given by the sampling interval over time while the height of the steps is given by the resolution of the A / D converter

At the end of the sampling a numeric vector is obtained that enters the computer stored in memory cells and divided into n-points blocks (Figure below) since the algorithm applied act on a number n of points. Next, the spectrum will be calculated for each block.

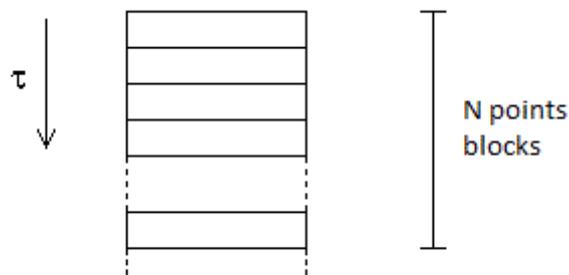


Figure 4: Data vector in memory cells

2.1.3 Frequency domain analysis

FOURIER TRANSFORM

As defined previously, a signal is represented by mathematical function varying in time domain. However, considering many reasons in some application better study signals represented on frequency domain.

Any periodic function representing a time variant signal, can be decomposed as sum of harmonic waves (sine and cosine waveform), when the signals are no periodic behavior the sum operation is replaced by integral one. Using Fourier transformation, it is possible to find what terms compose sum or integral operation and for each component is associated a frequency and an amplitude. Thanks to Fourier transformation it is possible to obtain a relationship between signal frequency and its amplitude, this is defined as spectral representation of the original time domain signal considered.

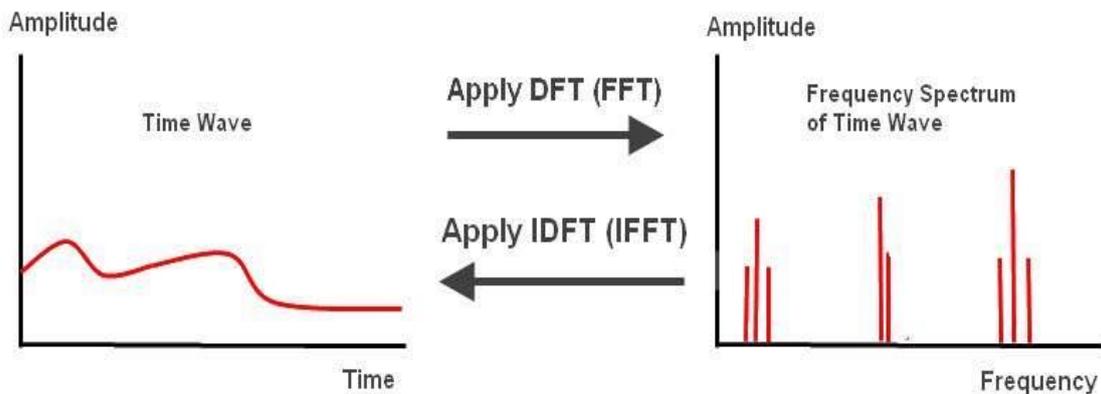


Figure 5: Time domain-frequency domain relation

The Fourier transformation of a time domain signal $x(t)$ is denoted by the symbology $X(f) = F\{x(t)\}$ and its analytical definition is:

$$X(f) = \int_{-\infty}^{+\infty} x(t) e^{-i2\pi ft} dt$$

On the contrary, starting from a signal in frequency domain, the Fourier anti-transformation let to obtain its time domain signal representation:

$$x(t) = \int_{-\infty}^{+\infty} X(f) e^{-i2\pi ft} df$$

DISCRETE FOURIER TRANSFORM AND FAST FOURIER TRASFORM

Discrete Fourier transforms

Related to computer signal processing some considerations must be done. Indeed, the continuous Fourier transformation cannot be applied having discrete signal form. Therefore, a discrete-time Fourier transformation is used to realize the spectral analysis of signal considered.

The discrete-time Fourier transform has the following formal expression:

$$X_{2\pi}(\omega) = \sum_{n=-\infty}^{+\infty} x[n]e^{-i\omega t}$$

Where $x[n]$ is a discrete set of real or complex numbers and the frequency variable, ω , has normalized units of radians/sample, with periodicity 2π .

Fast Fourier Transform elaboration

The mathematical algorithm computes signal spectrum starting from quantitated sample is called FFT (Fast Fourier Transform). It operates on N input real samples in time domain transforming them into N/2+1 complex sample in frequency domain following a constant spacing, each one composed of real and complex part: therefore, the algorithm produces a linear shaping on the frequencies axis. This representation type confers to both high and low frequencies the same heaviness.

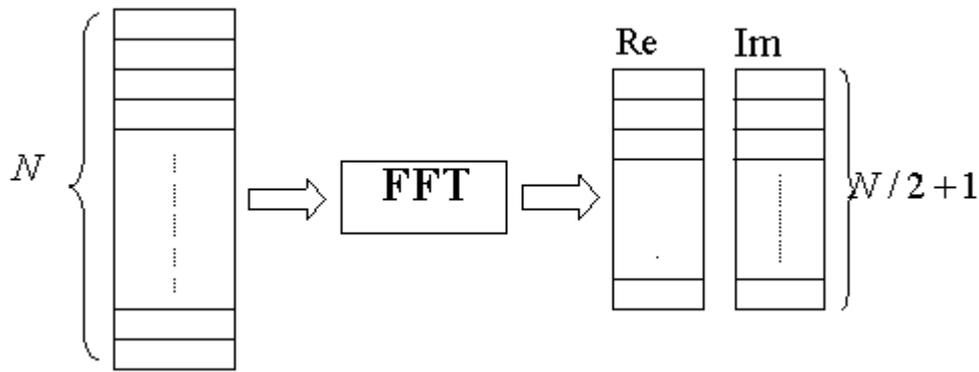


Figure 6: FFT Algorithm scheme

The first obtained element has real value and represents the signal mean value, in other words its continuous component, which is obviously a null frequency component. The second element has a frequency defined as:

$$f_1 = \frac{1}{N \cdot T}$$

Where T is sampling period. The other next frequency components are:

$$f_i = i \cdot f_1$$

Considering $i = 1, 2, 3 \dots N/2$ where the last is Nyquist frequency.

Related to transform elaboration execution, it possible do two considerations to obtain better performance:

- FFT algorithm employs real and complex part instead of amplitude and phase data form.
- N is selected as power of two

Indeed, if N is chosen N=4096, applying the algorithm to this samples number and considering the correctness sampling implementation 2049·(N/2+1) spectral lines are generated. As results of this, 2048 is the real number of spectral lines obtained because the continuous component of acoustic signal is null.

Another consideration useful is necessary about resolution of FFT computation. Indeed, it is related to sampling frequency and the number of samples generated. The following relation resume this concept:

$$\Delta f = \frac{\text{Frequency}}{\text{Samples}}$$

For example, using a sampling frequency of 48kHz the frequency range goes from 0 to 24kHz (Nyquist Frequency), the resolution result to be:

$$\Delta f = \frac{24\text{kHz}}{2048} = 11,71 \text{ Hz}$$

GAUSSIAN AND WHITE NOISE DISTRIBUTION

The signal theory studies the different signals representing various phenomena. According the signal amplitude changing in time, a process may be defined deterministic or nondeterministic. Indeed, in deterministic signal the amplitude variation in time is exactly known; on the other hands in a nondeterministic or random signal the amplitude trend can be characterized only in statistical terms.

A random process is defined white related the absence of periodicity in time and the amplitude spectral representation is uniformly distributed entirely on all frequency range. As result of this, the signal could change its value without any particular law: indeed, the components of this type of process change over time with a low correlation with respect to their trend in previous times. In practice, however, white noise does not exist: it is a theoretical idealization, since no system is able to generate a uniform spectrum for all frequencies extended from zero to infinity, while in real cases the white noise is at more referable to a range of frequencies (finite or limited band white noise)

A process is defined Gaussian when the amplitude of the process is characterized by well-known behavior in time, in other words the random amplitude values following a Gaussian distribution.

As shown in the following figure, the two different process type are not much different considering in time domain especially considering a small temporal windows observation.

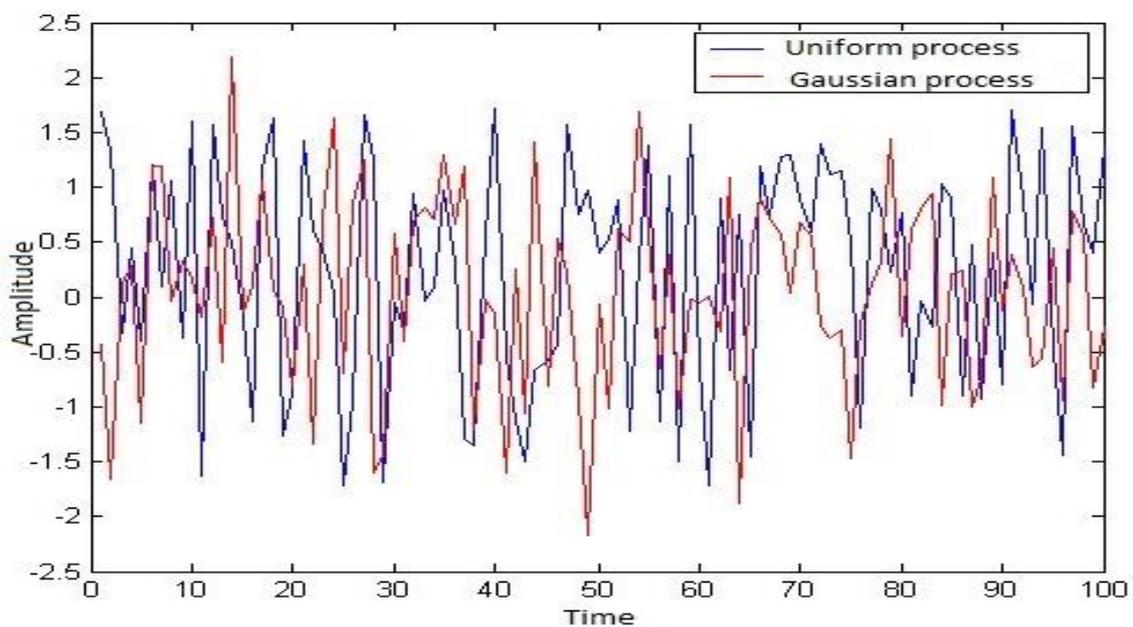


Figure 7: Gaussian and Uniform waveform trend in time

Starting from the previous processes, considering their amplitude distribution it is possible to build a histogram useful to illustrate how often the signal assumes a given amplitude during a defined observation time.

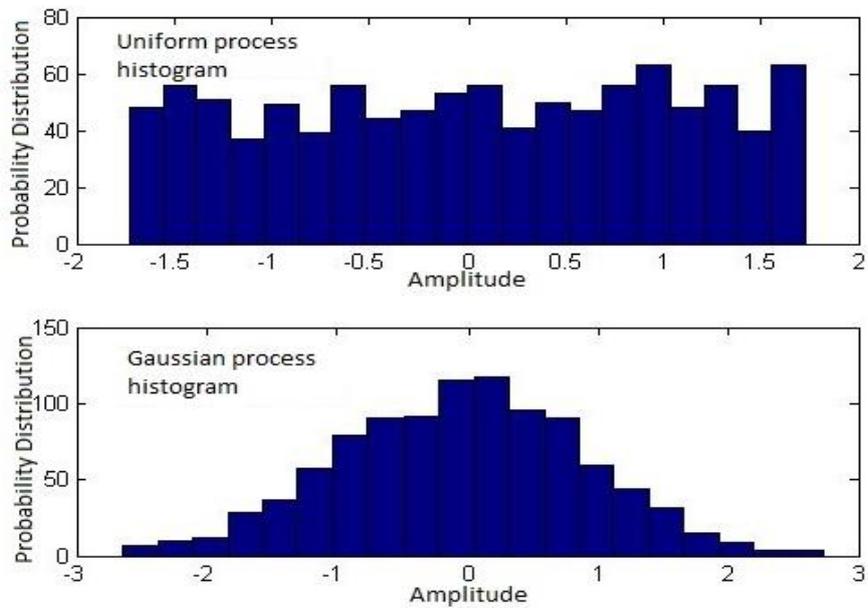


Figure 8: Uniform and Gaussian process histograms

As it is clearly illustrated in previous charts: uniform white noise, from instant t to $t + 1$, the signal can change value in any way (the process has a low correlation); gaussian noise signal cannot change from one instant to the other more than is allowed by the maximum frequency of its spectrum.

2.1.4 Filters

The filter, in electronics, is a system or device which realizes the elaboration of signal provided as input of it. Filter transfer function is the frequency domain relationship describing the elaboration process done by filter providing an output according a given input. Filter transfer function is graphically representable by means of Bode diagram in which is shown signal variation in terms of amplitude and frequency called frequency response of filter. According bode diagram it is possible to distinguish low pass filter, high pass, band-pass filter, band-stop or band-rejection filter, all-pass filter. According technological aspect electronics filter can be defined as: active o passive, analog or digital, discrete or continuous time, linear or not linear filter.

ANTI-ALIASING FILTER

Discretization process of continuous time signal let to better manage and elaborate signal using electronic device such as computers or others instruments such as oscilloscope. However, digital signal processing lead to some problems in terms of good and reliability of reconstructed signal. To guarantee this, a peculiar filter type called Anti-aliasing filter is used. In the following picture is illustrated the steps of analog to digital conversion of a given analog signal:

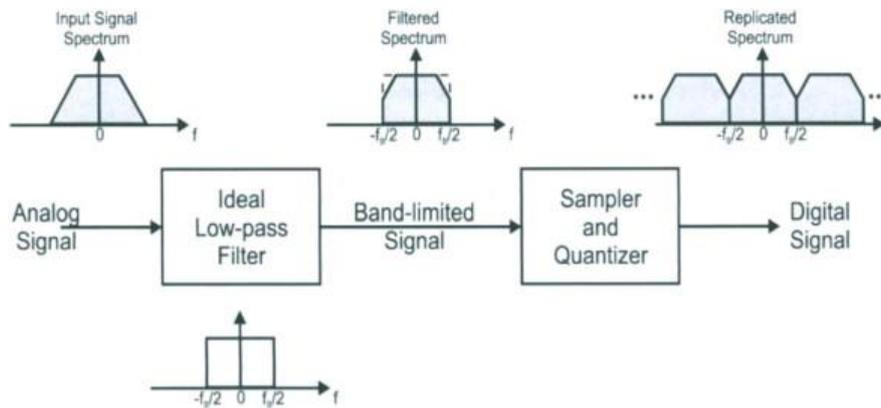


Figure 9: Analog to digital conversion process

An anti-aliasing filter (AAF) is an analog filter used before an ADC and before the signal sampler. This electronic component is used to restrict the bandwidth of a signal to approximately or completely satisfy the Nyquist-Shannon sampling theorem over the band of interest. Since the theorem states that unambiguous reconstruction of the signal from its samples is possible when the power of frequencies above Nyquist frequency is zero, a real anti-aliasing filter trades off between bandwidth and aliasing. A realizable anti-aliasing filter will typically either permit some aliasing to occur or else attenuate some in-band frequencies close to the Nyquist limit. For this reason, many practical systems sample higher than would be theoretically required by a perfect AAF to ensure that all interested frequency can be reconstructed, a practice called oversampling. Often, an anti-aliasing filter is a low-pass filter, however, generalizations of the Nyquist-Shannon sampling theorem allow sampling of other band-limited passband signals instead of baseband signals. For signals that are bandwidth limited, but not centered at zero, a band-pass filter can be used as an anti-aliasing filter.

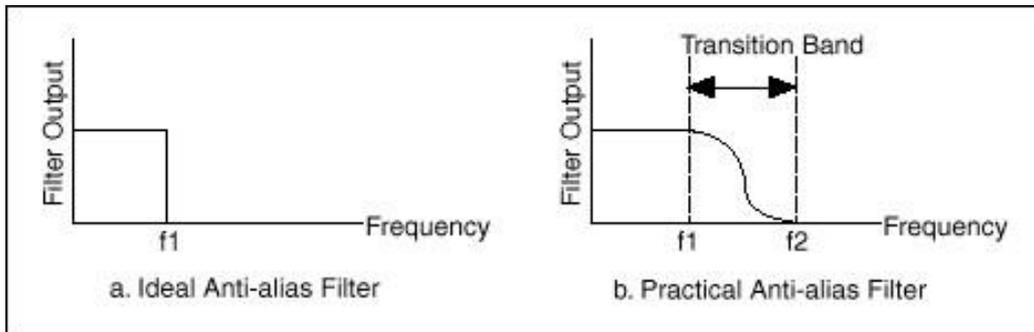


Figure 10: Anti-alias filter behaviour

As is shown in previous image the ideal Anti-alias Filter An ideal anti-alias filter passes all the appropriate input frequencies (below f_1) and cuts off all the undesired frequencies (above f_1). On the other hands, the practical Anti-alias Filter allows to pass all frequency minor of f_1 and cut off all frequency major of f_2 . The transition band contains a gradual attenuation of the input frequencies. Although you want to pass only signals with frequencies lower than f_1 , those signals in the transition band could still cause aliasing. Therefore, in practice, the sampling frequency should be greater than two times the highest frequency in the transition band. This turns out to be more than two times the maximum input frequency (f_1). That is one reason why you may see that the sampling rate is more than twice the maximum input frequency.

FILTER TYPES

As mentioned previously, a filter is described by transfer function, often defined in the domain of the complex frequencies. The back and forth passage to/from this domain is operated by the Laplace transform and its inverse. The transfer function $H(s)$ of a filter is the ratio of the output signal $Y(s)$ to that of the input signal $X(s)$ as a function of the complex frequency s :

$$H(s) = \frac{Y(s)}{X(s)}$$

Considering $s = \sigma + j\omega$

Filter may be differing by the approximating polynomial used and each lead to certain characteristics of the transfer function of the filter. Refers to the particular characteristics it is possible to resume main filter families:

- Butterworth filter
- Chebyshev filter
- Bessel filter
- Elliptic filter
- Optimum “L” filter
- Gaussian filter

For each filter family it is possible to specify a proper order, which represent the grade of “imitation” of ideal filter behavior. The higher is the order, the more filter step response is like the ideal one. On the other side, the impulse response and the latency of the filter is longer when its order increasing.

Butterworth filter

The Butterworth filter is designed to have a frequency response as flat as possible in the passband. indeed, the frequency response of the Butterworth filter is maximally flat, absence of ripples, in the passband region and goes immediately to zero in the

stopband. The following picture illustrate bode diagram of a first-order Butterworth low-pass filter.

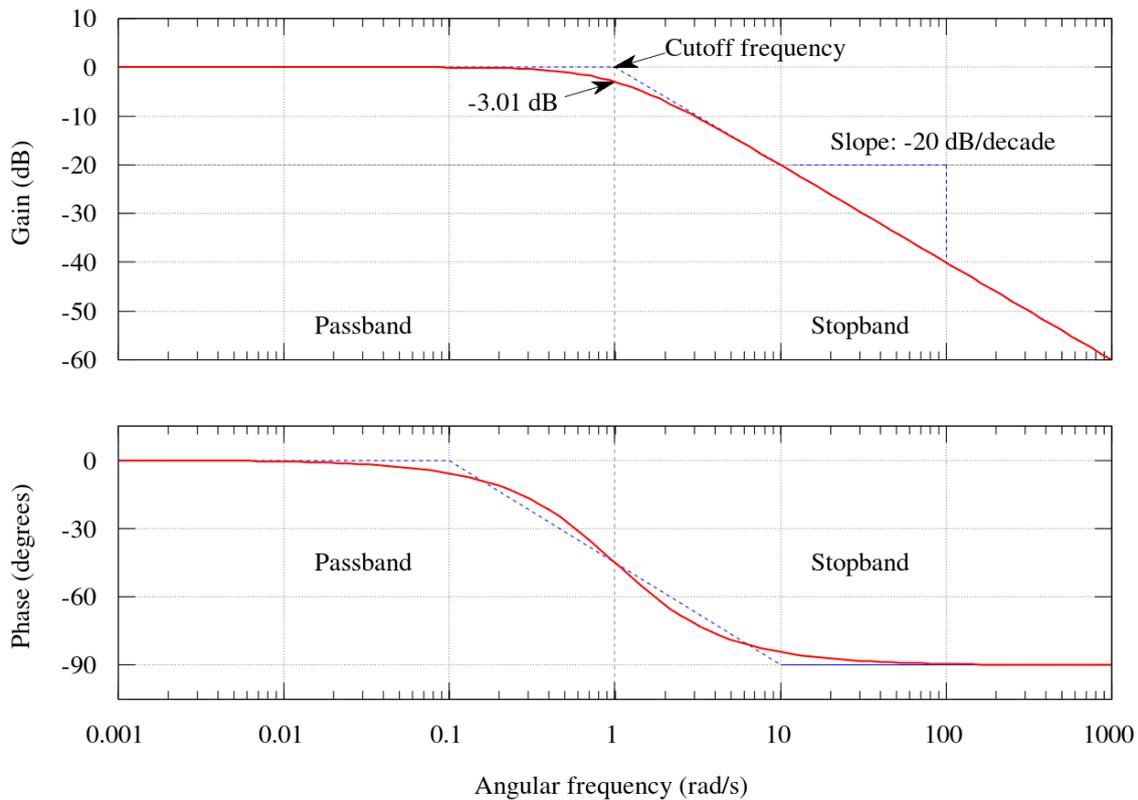


Figure 11: Bode diagram of the Butterworth low-pass filter

As shown in previous figure, the response slopes off linearly towards negative infinity. A first order filter's response rolls off at -20dB per decade, like as all first order filter. Butterworth filter is characterized by a slow roll-off respect to other filter type, as result of this it requires a high order to implement a given stopband specification, on the contrary Butterworth filters have a more linear phase response in the pass-band than Chebyshev filters.

Butterworth low pass filter transfer function has frequency response varying according the n order filter considered. It is possible to obtain the gain $G(\omega)$ of a filter through its transfer function and as will be show in the next figure the slope of it is $20n\text{dB}/\text{decade}$.

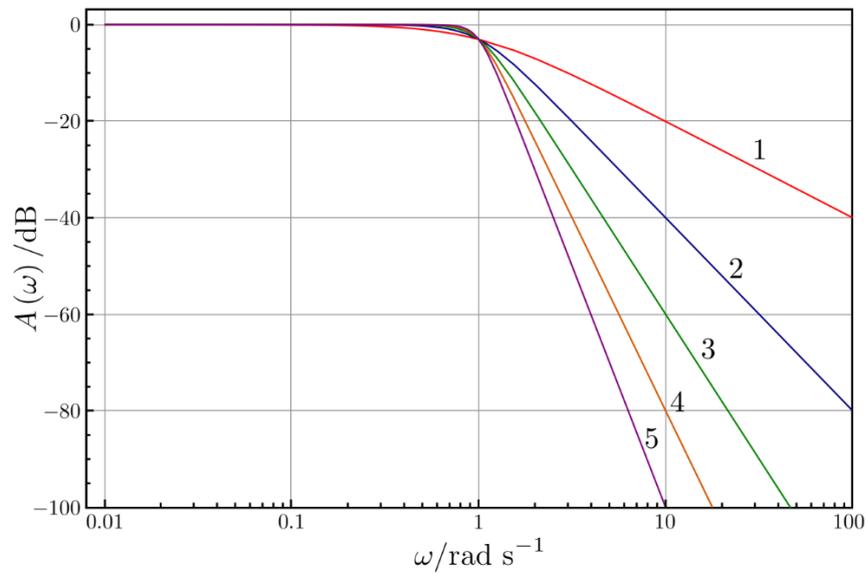


Figure 12: Transfer function of the Butterworth low pass filter

The relationship linking two quantities is:

$$G^2(\omega) = |H(j\omega)|^2 = \frac{G_0^2}{1 + \left(\frac{j\omega}{\omega_c}\right)^{2n}}$$

Where:

- n is filter order
- ω_c is cut-off frequency (approximately the -3dB frequency)
- G_0 is the DC gain of transfer function

Chebyshev filter

Chebyshev filters can be distinguished in type I and type II with different characteristics. Chebyshev type I has a steeper roll-off and more passband ripple than Butterworth filters, while type II filter has more stopband ripple.

Type I Chebyshev filters are the most common types of Chebyshev filter. The gain response $G_n(\omega)$, as a function of angular frequency ω of the n -order low pass filter is equal to:

$$G_n(\omega) = |H(i\omega)| = \frac{1}{\sqrt{1 + \varepsilon^2 T_n^2\left(\frac{\omega}{\omega_0}\right)}}$$

Where :

- ε is the ripple factor
- ω_0 is the cut-off frequency
- T_n is a Chebyshev polynomial of the n -th order

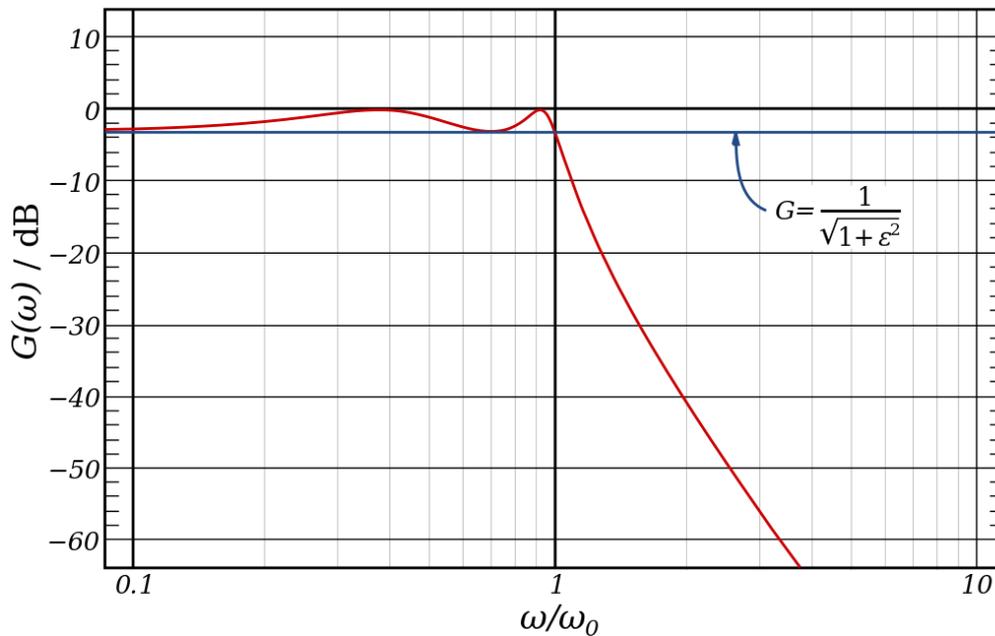


Figure 13: Transfer function of Chebyshev filter

Type II Chebyshev filter, also called inverse Chebyshev filters, is less used than the previous one because its implementation requires more components. It is characterized by absence of ripple in the passband, while has equiripple in the stopband.

The gain response $G_n(\omega)$ has the following equation:

$$G_n(\omega) = \frac{1}{\sqrt{1 + \frac{1}{\varepsilon^2 T_n^2 \left(\frac{\omega_0}{\omega} \right)}}}$$

Elliptic filter

An elliptic filter type is characterized by the same ripple in both passband and stopband region, quick transition in gain between the passband and stopband. Indeed, it is the fastest filter respect to the others mentioned with a given ripple.

The gain frequency response of a low pass elliptic filter can be written as:

$$G_n(\omega) = \frac{1}{\sqrt{1 + \varepsilon^2 R_n^2 \left(\xi, \frac{\omega}{\omega_0} \right)}}$$

Where:

- R_n is the elliptic rational function
- ω_0 is the cutoff frequency
- ε is the ripple factor
- ξ is the selectivity factor

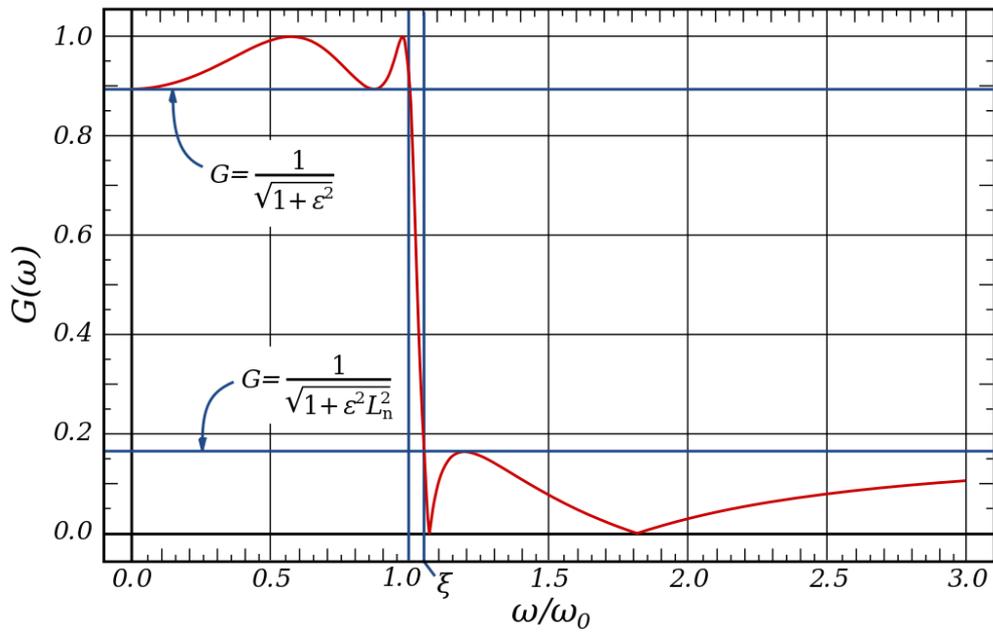


Figure 14: Transfer function of Elliptic filter

As can be seen from the previous figure in the passband the gain varies between 1 and $1/\sqrt{1+\epsilon^2}$, while in the stopband region the gain oscillates between 0 and $1/\sqrt{1+\epsilon^2 L_n^2}$ (where $L_n = R_n(\xi, \omega)$). In the following figure it is shown the comparison between the linear filter mentioned and their gain frequency response:

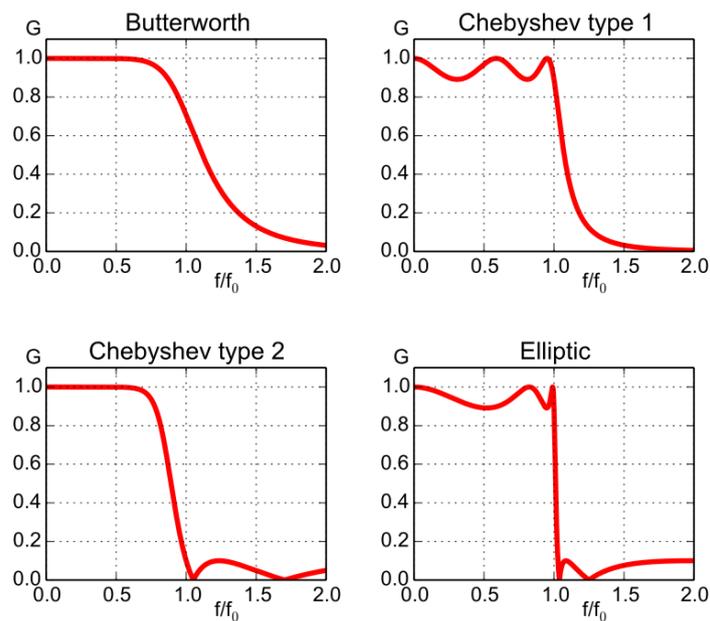


Figure 15: Frequency response comparison

2.2 Fatigue

Fatigue is a phenomenon related to huge quantity of objects and mechanical structures; it can change their internal plastic properties causing damage at different level. Material fatigue can be defined as structure failure occurring when it is subjected to a cyclic load. Material fatigue phenomenon is the most common cause of damage and failure in mechanical structures and it occurs even when the experienced stress range is far below the static material strength.

Permanent damage caused by fatigue can occur with a no definite number of cycles, indeed it can verify after a few cycles or up to many millions of cycles. The parameters influencing fatigue life can be many such as the stress amplitude of each load cycle; the value of the mean stress; the type of stress, for example a tensile stress state has stronger effect on damaging respect to compressive stress. Also, surface roughness and environment condition are parameter able to characterize fatigue properties. Considering the moment in which a crack happens in a mechanical structure, the damage continuously increases after each load cycle until break down condition where the structure does not support the peak load amplitude of stress cycle.

Different failure process can be distinguished according to how the material damage is propagated. It is possible to divide fatigue damage in three stages:

- The damage develops on the microscopic level and grows until a macroscopic crack is formed, due to many cycles.
- The macroscopic crack grows for each cycle until it reaches a critical length.
- The cracked component breaks because it can no longer sustain the peak load.

The fatigue is considered the main causes of failure in any mechanical structure, it could interest various component related to automotive, jet engine, ships bridges and others section. The ageing phenomenon on all part is commonly caused by oscillating stress interesting the considered mechanical part. The oscillation load can be characterized by deterministic or random properties; they are respectively related to the probability to

know the future stress amplitude value following defined mathematical law or the actual oscillation has not any correlation with the successive one.

The studies based on vibration theory provide some instruments to make simpler the analysis on system subject to vibration stress condition. The modal analysis approach is considered an effective method to study changes in structural characteristic of metallic material. Indeed, it has been observed modification of modal parameter such as modal frequency, damping factor and mode shape consequently to stress condition caused by external vibrations.

Fatigue Variables

When a material is subjected to stress, its internal structure state varies in time. Using a series of specific variables, it is possible to describe the internal material condition at given time instant. Simplify the concept, a load cycle can be defined as the duration from the duration from one peak in the studied variable to the next peak (even if all cycles are characterized by have not the same amplitude). On the other hands, it is necessary to consider the external load nature (most common is random or periodic load) characterizing the material response to a sequence load cycle.

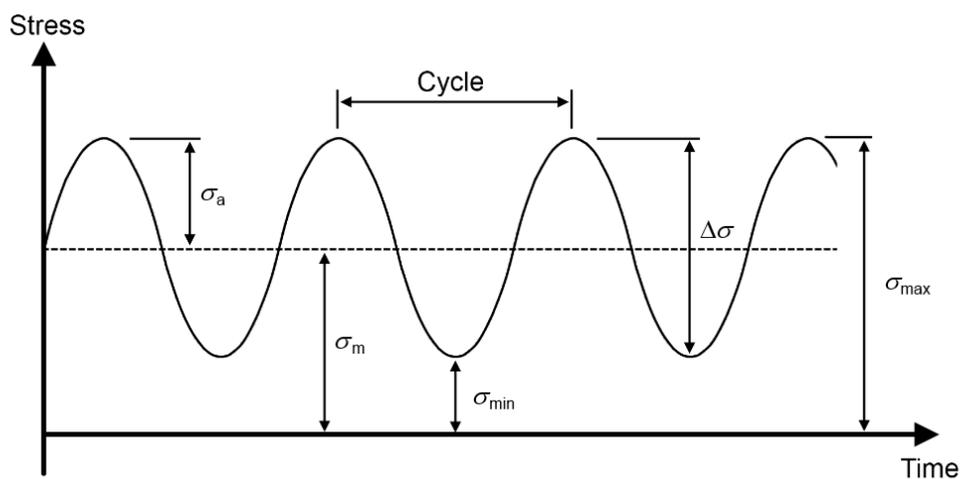


Figure 16: Fatigue parameters

As illustrated in the previous figure, it is possible to define a load cycle by mean the following variable:

- σ_{max} maximum stress
- σ_{min} minimum stress
- σ_a stress amplitude
- σ_m mean stress
- $\Delta\sigma$ stress range

It is possible to describe relationship exist between the mentioned variable. In detail, can be write:

The mean stress is:

$$\sigma_m = \frac{\sigma_{max} + \sigma_{min}}{2}$$

The stress amplitude, considers as the most important parameter for fatigue damage, can be written as:

$$\sigma_a = \frac{\sigma_{max} - \sigma_{min}}{2}$$

The stress range is equal to difference between maximum minimum stress:

$$\Delta\sigma = \sigma_{max} - \sigma_{min}$$

Another common parameter used to describe stress cycle is the *R - value*, defined as:

$$R = \frac{\sigma_{min}}{\sigma_{max}}$$

Fatigue Models

The analysis of material fatigue can be done using different approach, among the all possible methods the most common is called S-N curve. This model type considers the relationship between the number of cycles until failure, N (i.e. material lifetime), and the stress amplitude in uniaxial loading. Indeed, a longer lifetime is strictly related to lower value of stress amplitude. On the other hands, there are materials that if stressed below a threshold (also called endurance limit) no fatigue damage is observed, and components can operate for an infinite lifetime.

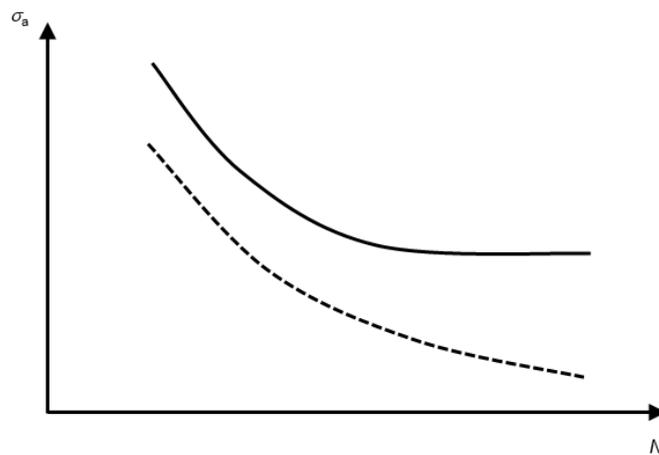


Figure 17: Typical S-N curve

In the previous figure is illustrated a typical S-N curve for a material with an endurance limit (solid line) and without an endurance limit (dashed line).

In random loading, the parameters of each stress cycle are different from the others. This leads to consider another approach to predict fatigue, the full stress history must be transformed into a stress spectrum that can be related to fatigue in the next step of the analysis. The Rainflow counting algorithm is commonly used to define a set of stress amplitude with corresponding mean stresses. In case of certain material, fatigue life is strongly influenced by the number of micromechanical defects, indeed component lifetime can vary according the material position of defects. To study this phenomena type can be used probabilistic methods. The applicability of prediction model depends both material and loading type. An application of Rainflow counting algorithm describing

stress cycle distribution is shown in the following figure:

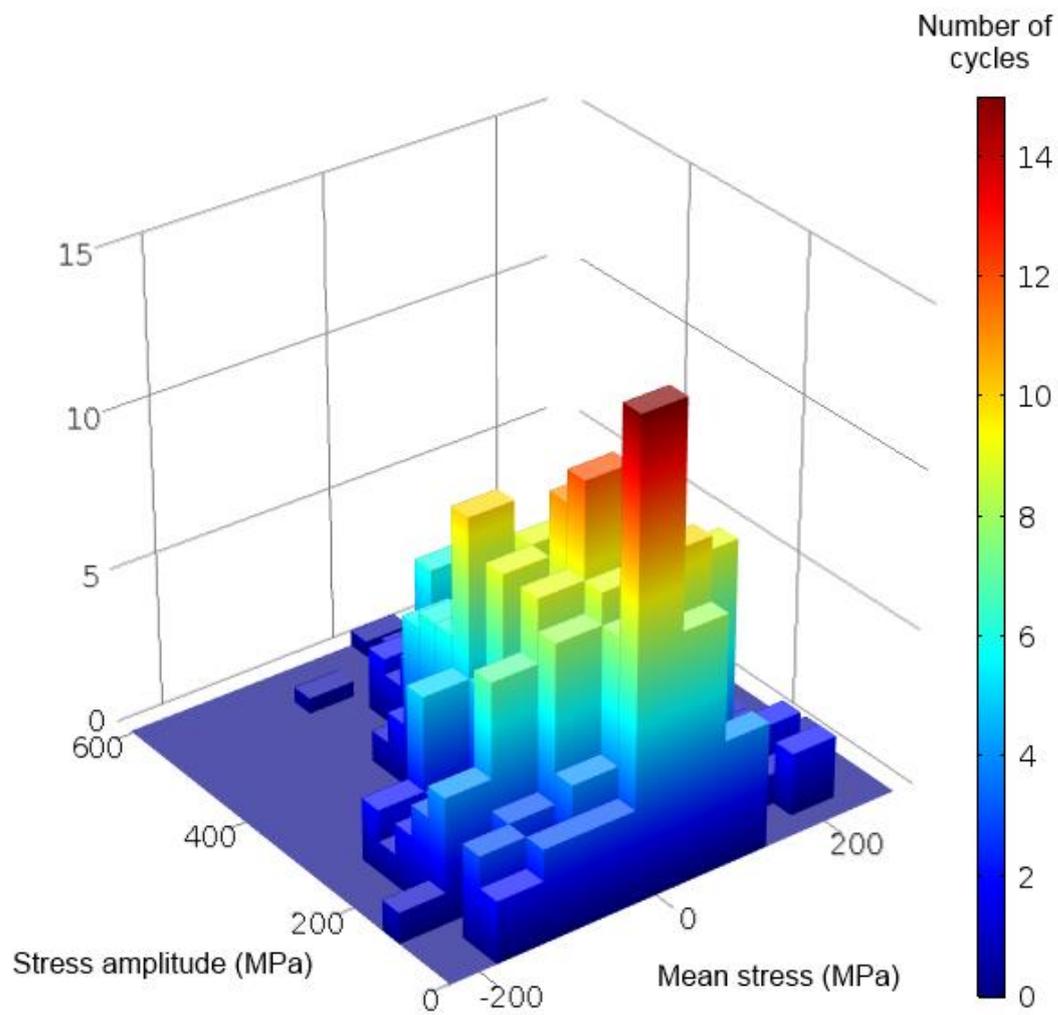


Figure 18: Stress cycle distribution

A common application of random loading is studies of material vibrational fatigue. Since the stress depends on the excitation frequency, the fatigue evaluation can be made in the frequency domain using, for example, power-spectral density methods.

2.3 Vibrations

Vibration is a mechanical phenomenon in which is present oscillations respect to an equilibrium point. The vibration theory concerns the study of oscillatory motions of bodies and its associated forces. Generally, vibration is undesirable event because it can be the cause of many misbehavior regarding various mechanical device, for example engines, electric motors and others.

It is possible to distinguish vibration according the cause of phenomena verifying: free vibration, forced vibration, damped vibration.

- **Free vibration** occurs when a mechanical system is set in motion with an initial input and allowed to vibrate freely. The mechanical system vibrates at one or more of its natural frequencies and damps.
- **Forced vibration** take place when a time-varying disturbance is applied to a mechanical system. The disturbance can be a periodic and steady-state input, a transient input, or a random input.
- **Damped vibration** when the energy of a vibrating system is gradually dissipated by friction and other resistances. The vibrations gradually reduce or change in frequency or intensity until the system rests in its equilibrium position.

Forced vibration phenomena, usually, is interested in vibration testing of structural mechanics. It is possible to notice that testing is executed by means of some type of shaker, it is linked to a device under test (i.e. metallic specimen). Vibration testing is performed to examine the behavior of device under test related to a definite vibration environment.

According the frequency range execution of the test, shaker characteristic change to satisfy the various testing requirements. Indeed, for low frequency forced vibrations a servo hydraulic or electrohydraulic shaker are used; on the contrary for high frequencies, usually electrodynamic shakers are adopted to perform the test. As for the vibration tests performed in the laboratory, a device under test can be subjected to more types of vibrations. Among the most common ones used to perform the tests it is possible to

mention: sinusoidal and random vibrations. Sinusoidal vibration test is performed to investigate the structural response of the device under test, on the other hand random excitation executed by shaker is generally used to replicate real world environment stresses to a mechanical system.

VIBRATION ANALYSIS

A significant aspect of test execution is represented by vibration analysis, with the aim to detect any possible fault occurs to a mechanical system identifying unbalance, misalignment, rolling element bearing faults and resonance condition. Vibration analysis is centered to study displacement, velocity and acceleration of specimen, represented as time domain waveform. Starting from time waveform, spectrum analysis is executed by means of Fast Fourier transform (FFT) operation to obtain important frequency information about the system analyzed.

It is possible to explain vibration motion through its similarity respect to the harmonic motion of a body. Harmonic motion can be represented by means of a vector \overline{OP} of magnitude A , rotating at a constant angular velocity ω . In the following figure it is shown the two representation of the same harmonic motion:

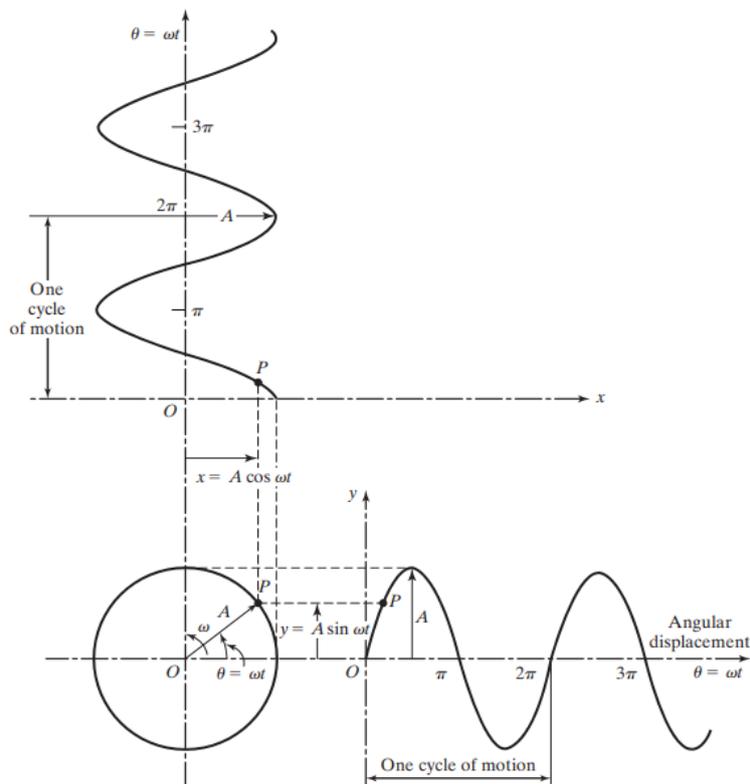


Figure 19: Harmonic motion as the projection of the end of a rotating vector (From L. Meirovitch, Fundamentals of vibrations. Boston: McGraw-Hill, 2001)

As it is possible to observe in the previous picture, the projection of the tip of the vector \overrightarrow{OP} on the vertical axis is given by $y = A \sin \omega t$, while its projection on the horizontal axis is $x = A \cos \omega t$.

Using complex-number representation, the rotating vector $\vec{X} = \overrightarrow{OP}$ can be written as $\vec{X} = Ae^{i\omega t}$, where ω denotes the circular frequency of vector \vec{X} .

Applying the differentiation operation respect to time, it is possible to obtain the expression of displacement, velocity and acceleration. They can be expressed as:

- Displacement: $x(t) = \text{Re}[Ae^{i\omega t}] = A \cos \omega t$
- Velocity: $\dot{x}(t) = \text{Re}[i\omega Ae^{i\omega t}] = -\omega A \sin \omega t = \omega A \cos(\omega t + 90^\circ)$
- Acceleration: $\ddot{x}(t) = \text{Re}[-\omega^2 Ae^{i\omega t}] = -\omega^2 A \cos(\omega t + 180^\circ)$

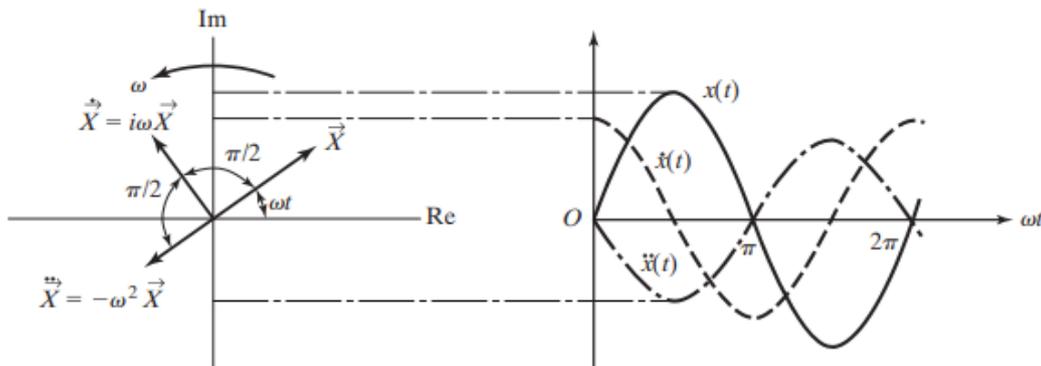


Figure 20: Displacement, velocity, and acceleration as rotating vectors (From L. Meirovitch, Fundamentals of vibrations. Boston: McGraw-Hill, 2001)

Any mechanical system interested by vibration can be described using harmonic motion. It is possible to write the following definition and terminology useful to study phenomena characteristics:

- **Cycle of vibration** is defined as the movement of a vibrating body from its undisturbed or equilibrium position to its extreme position in one direction, then to the equilibrium position, then to its extreme position in the other direction, and back to equilibrium position. One revolution of the vector \overrightarrow{OP} constitutes a cycle.

- **Amplitude of vibration** can be explained like the maximum displacement of a vibration body from its equilibrium position. In the previous figure the amplitude of vibration is equal to A .
- **Period of oscillation** is the time needed to complete one cycle of motion, it is equal to the time required for the vector \overrightarrow{OP} to rotate through an angle of 2π and hence

$$\tau = \frac{2\pi}{\omega}$$

Where ω is called the circular frequency.

- **Frequency of oscillation**, it is defined as the number of cycles that a vibrating body completes per unit time. It is denoted as f and it is equal to:

$$f = \frac{1}{\tau} = \frac{\omega}{2\pi}$$

f is measured in cycles per second (Hertz), ω is measured in radians per second.

- **Natural frequency** can be explain observing vibrational motion. Indeed, if a system after an initial disturbance, is left to vibrate on its own, the frequency with which it oscillates without external forces is defined as its natural frequency.
- **Mechanical resonance** is the tendency of a mechanical system to respond at greater amplitude when the frequency of its oscillations matches the natural frequencies of the system subjected to vibrations, than it does at other frequencies. Furthermore, many resonant objects have more than one resonance frequency and it will vibrate easily at those frequencies, and less so at other frequencies.

If resonance occurs in a mechanical system, it can lead to system failure. Often, vibration analysis is used to predict and to avoid failure system phenomena.

MODAL ANALYSIS

Modal analysis is the study of dynamic properties of systems in frequency domain, providing information about natural modes of the observed system. Modal analysis is usually the starting point to understand the natural frequencies, damping and mode shapes of the structure. An example of this is measuring the vibration of a body when it is attached to a shaker or the noise pattern in a room when excited by a loudspeaker. The modal analysis is useful to obtain many information about structural characteristic of a mechanical element, the analysis results are influenced by mass, stiffness and damping properties of structure considered.

The first step in experimental modal analysis is to measure the excitation and responses of the structure under test. The structure must be excited to have the vibrating responses. There are two common methods of excitation: impact hammer and modal shaker. An impact hammer shown in Figure (21) is a specialized measurement tool that produces short duration of excitation levels by striking the structure at certain.



Figure 21: Impact hammer instrumented with a force sensor with different hardness tips (from PCB Piezotronics official catalogue model: 086C03)

The hammer is instrumented with a force sensor which generates a voltage signal proportional to excitation force so that the excitation force is measured during the test. Impact hammer is often used for modal analysis on simple structures, or where attaching a modal shaker is not practical. Different hardness impact tips can be used with the hammer to change the measurement frequency range of the impact. When low frequency measurements are needed, a soft rubber tip may be used, and when high frequency measurements are needed, a hard metal tip may be used.

Typically, a modal shaker is connected to the structure with a small thin metal rod called “stinger”. A force sensor is placed on the structure at the driving point, and connected to one end of the stinger, to measure the excitation force. Quite often, an accelerometer is also mounted on the structure at the driving point to measure the acceleration levels. And this can be easily done by applying an Impedance head (force and acceleration sensor in one to the driving point on the structure).



Figure 22: Dongling ESD-045 Modal shaker

A shaker is often driven by the dynamic signal analyzer’s DAC (digital to analog converter), which is an electronic device to generates carefully tuned electronic signals

which are then amplified and converted into the excitation signals. There are several different types of excitation signal profiles can be used to excite the structure, including pure random (white noise), burst random, pseudo random, periodic random, chirp (burst chirp), etc.

Nowadays, the experimental modal analysis systems use to analyze a mechanical body are composed by the following elements:

- Excitation instrument: impact hammer or modal shaker
- Sensors such as transducer (usually accelerometer, extensimeter, load cells)
- Data acquisition system and analog to digital converter
- Host Pc to visualize the data and analyze them.

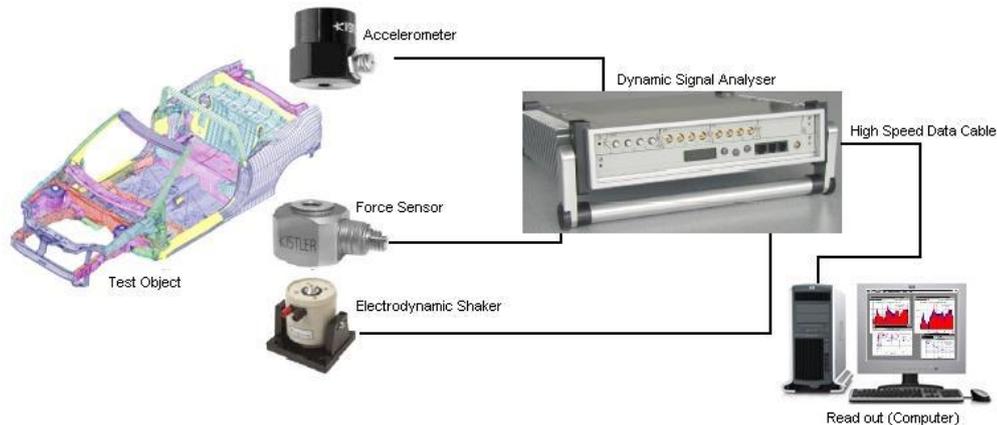


Figure 23: Example of a test bench system for modal analysis

Modal analysis can be executed following different approach:

- SIMO (single-input, multiple-output) system is characterized by one excitation point, and then the response is measured at many other points. This approach can be inappropriate considering some mechanical system, indeed in recent years the MIMO bench system have become more popular.
- MIMO (multi-input, multi-output) approach, where partial coherence analysis identifies which part of the response comes from which excitation source. Using

multiple shakers leads to a uniform distribution of the energy over the entire structure and a better coherence in the measurement. A single shaker may not effectively excite all the modes of a structure.

The analysis of the signals typically is based on Fourier analysis, generally Fast Fourier Transform (FFT) is applied. The resulting transfer function will show one or more resonances, whose characteristic mass, frequency and damping can be estimated from the measurements.

Enhancing Measurement Resolution

One important consideration in modal testing measurements is the frequency resolution, that is fundamental especially in some frequencies. Indeed, when natural frequencies occur close together, a finer frequency resolution may be needed to accurately represent the frequency and damping of the two peaks. There are two ways to increase the frequency resolution: increasing frame size or using FFT zoom analysis. The frequency resolution depends on the number of points in the time frame, more points in the time block results in more frequency lines in the frequency domain spectrum. Thus, the frequency resolution can be increased by increasing the number of points in the time block. The drawback is that a longer time frame requires more time of data acquiring, which increases the overall measurement time. This is even more noticeable when the frequency span is low, such as below 50 Hz. The second method for increasing frequency resolution is to use FFT zoom analysis. This technique uses a special algorithm to compute the spectrum within a band that does not start from frequency zero (or DC), unlike standard “base band” spectra that are normally used when FFT zoom is not used.

When measurement data is acquired, and complete data set is saved the acquisition process is not complete. Indeed, it is necessary to export the data to the format which modal analysis software supports. Usually, a dynamic signal analyzer is used to obtain the data and manage it in a format readable by the modal analysis software. The data set dimension acquired is related to the quantity of sensors used and the resolution selected. Usually, the high large dimension of total data acquired lead to avoid manual editing of

them and is convenient to export data in format compatible with the most popular modal analysis software.

The next data elaboration process consists of extraction of modal parameters from saved data, it includes the natural frequencies, damping factor and mode shapes. The last one can be built as animation starting from structure geometry.

3 Methodology

The thesis has been development in partnership between DIMEAS Laboratory (Dipartimento di Ingegneria Meccanica e Aerospaziale) and DAUIN (Dipartimento di Automatica Informatica) research laboratories. The aim is to implement a software control system to perform modal analysis of alloy specimen to study its structural characteristic, providing sensible data useful to describe a series of material characteristics. Thesis work is composed essentially of three moments:

- Software development, performed in DAUIN

- Software testing in real bench system, executed in DIMEAS laboratory
- Software validation

Software development has been implemented using the software has been made available by DAUIN of *Politecnico di Torino*, in details NI LabVIEW software and its suitable extension, MathWorks® MATLAB and Microsoft Excel.

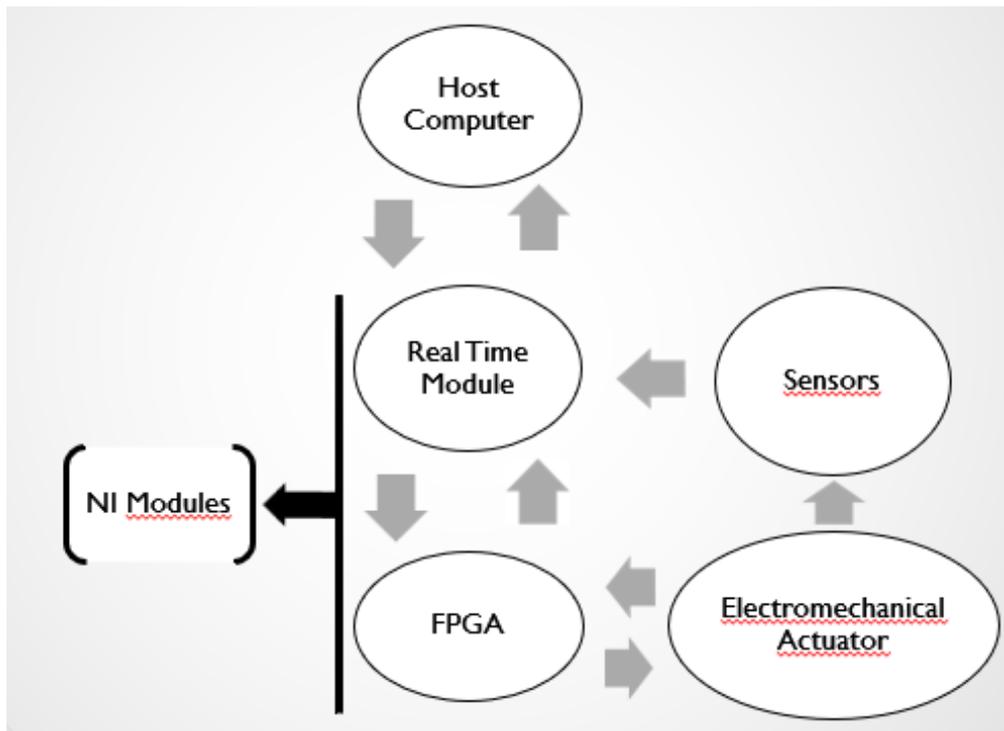


Figure 24: Test Bench system flowchart

Software testing has been performed following the typical test bench system configuration used to implement structural modal analysis, made of modal shaker, accelerometers, load cell and real time hardware for data managed.

Software validation step has been focused on the software data acquisition part, to evaluate effectively the absence of data loss coming from all sensors used in test bench system.

3.1 Test bench system

As illustrated in the previous lines(paragraph), test bench system used to implement the software and to execute the test is quite like the one commonly used for modal analysis. Indeed, in this case has been chosen to use only National Instrument module, sensors and modal shaker tanks to optimum integration with sensors necessary to perform modal analysis. As result of this, there is no presence of digital analog converter hardware dedicated because already integrated in the used NI modules.

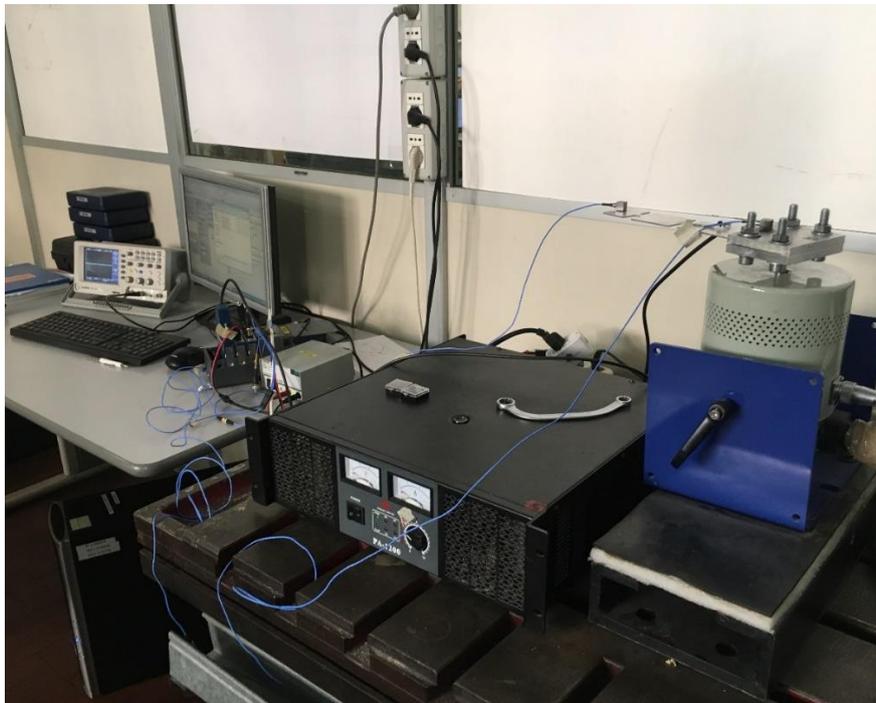


Figure 25: Complete Test bench System

In the previous figure it is possible to see the test bench system used, in details:

- Host PC in which run LabVIEW software to manage all devices used
- NI modules where is implemented real-time functionalities
- Linear power amplifier necessary to raises power of the analog signal for Modal Shaker
- Modal Shaker to transform the signal into desired mechanical vibrations

- Sensors that are distributed in various zone of specimen to analyze

3.2 Hardware platform

NATIONAL INSTRUMENT MODULES

Considering the test bench system described in the previous lines, all data exchanged between host computer and all devices composed hardware platform are managed by only National Instrument modules. In details, the following modules has been used:

- NI Compact RIO Controller (NI cRIO-9024)
- NI 9234
- NI 9263
- NI 9237

CompactRIO

CompactRIO combines an open-embedded architecture with small size, extreme ruggedness, and C Series modules in a platform powered by the NI LabVIEW reconfigurable I/O (RIO) architecture. Each system contains an FPGA for custom timing, triggering, and processing with a wide array of available modular I/O to meet any embedded application requirement

NI cRIO-9024 description and specification

The NI cRIO-9024 device belongs to the family of NI CompactRIO Control and Mechatronics Bundle. This National Instrument device is an embedded control platform composed by four components: real time controller, reconfigurable chassis with FPGA and I/O modules.



Figure 26: NI cRIO-9024 module

The Real-Time Controller is connected to a Chassis in which are allocated the I/O modules. The target can have different slot number to placed more than one input/output modules, in the following figure is shown a common chassis connected to NI-cRIO 9024.



Figure 27: NI 9024 and Reconfigurable Embedded Chassis

The CompactRIO real-time controller has an 800 MHz processor that can managed real time control algorithms, signal analysis, and data logging. Thanks to this device it is possible to deploy the control algorithm and run LabVIEW code without the code generation step.

Another fundamental characteristic of this device is the presence of Reconfigurable Chassis with FPGA (Field-Programmable Gate Array), that is the reprogrammable silicon chips. In other words, when an FPGA is programmed it rewires the chips itself inside it to implement desired functionality rather than run a software application.

The controller can manage the I/O module by means of the reconfigurable embedded chassis (shown in the previous picture). All circuitry required for measurement is contained in the module itself and all A/D – D/A conversion occur in the module before the data reaches the chassis. Each I/O module contains built-in signal conditioning and screw terminal, BNC, or D-Sub connectors.

In the following table are resumed the most important cRIO-9024 specification:

Network

Network Interface

Ethernet port 1	10BaseT, 100BaseTX, 1000BaseTX Ethernet
Ethernet port 2	10BaseT and 100BaseTX Ethernet
Compatibility	IEEE 802.3
Communication rates	
Ethernet port 1	10Mbps, 100Mbps, 1000Mbps auto-negotiated
Ethernet port 2	10Mbps, 100Mbps, auto-negotiated
Maximum cabling distance	100m/segment
RS-232 DTE Serial Port	
Baud rate	600bps to 230,400 bps
Data bits	5,6,7,8
Stop bits	1,1.5,2
Parity	Odd, even, mark, space, none
Flow Control	RTS/CTS,XON/XOFF,DTR/DSR,none
USB Port	
Maximum data rate	480Mb/s
Maximum current	500mA
Memory	
Nonvolatile	4GB
DRAM	512MB
Power Requirements	
Recommended power supply	55W secondary, 35VDC maximum
Power consumption	
cRIO-9024 only	17W
cRIO-9024 supply power to eight C series modules	35W
Voltage requirement	
On powerup	9V to 35V
After powerup	6V to 35V

National Instruments C series

The NI C series modules are suitable for measurement, control, and communication applications. Thanks to their well versality C series module can connect to any sensor or bus allowing a high-accuracy measurements suitable for data acquisition and control applications.

NI 9234 module

The NI 9234 is a four-channel dynamic signal acquisition module for making high accuracy measurements for IEPE (Integrate Electronics Piezoelectric) sensors, with 102dB of dynamic range and signal conditioning at 2mA constant current for accelerometers and microphone. Other characteristic of this module is the presence of built-in antialiasing filters allowing to obtain an automatically sampling rate adjustment.



Figure 28: NI 9234 module

The NI 9234 analog input signal on each channel is buffered, conditioned, and the sampled by a 24.bit Delta-Sigma ADC. In the following picture can see the Input Circuitry for channel:

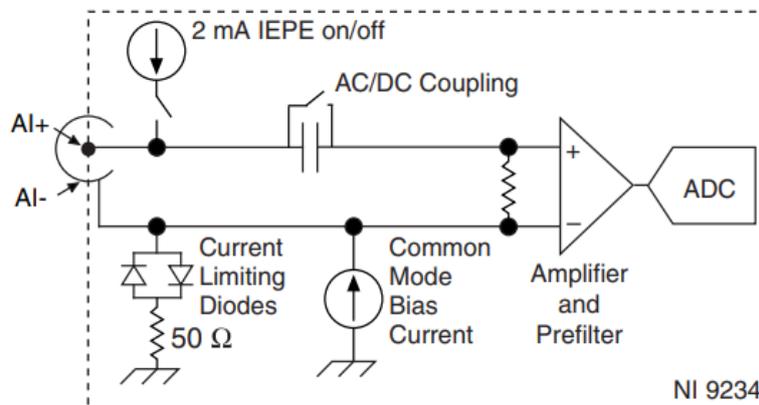


Figure 29: NI 9234 Input Circuitry for One Channel

The analog input channels are referenced to chassis ground through a 50Ω resistor, and is protected from overvoltage. The AC/DC coupling can be set to enabling the IEPE excitation current on or off.

Concerning the filtering and the relative passband frequency response, the NI 9234 is characterized by a combination of analog and digital filtering to provide an accurate representation of in-band signals and reject out-of-band signals. The signals within the passband have frequency-dependent gain or attenuation. The digital filters of mentioned NI module adjust the frequency range of the passband to match the data rate, as result of this the gain or attenuation at a given frequency depends on the data rate. The NI 9234 includes an internal master timebase with a frequency (f_M) of 13.1072 MHz, it controls the data rate (f_s) of the module.

The following equation provides the available data rates of the NI 9234:

$$f_s = \frac{f_M \div 256}{n}$$

However, it is possible to have an external or export own master timebase to obtain a synchronization between two or more C series modules. Indeed, to synchronize the data rate of NI 9234 module with others the use master time-based to control sampling, all the modules must share a single master time-based source.

The NI 9234 module most important specifications are shown in the following table, valid in range -40 °C to 70 °C, unless expressed noted.

Input Characteristics

Number of channels	4 analog input channels
ADC resolution	24 bits
Type of ADC	Delta-Sigma (with analog prefiltering)
Sampling mode	Simultaneous
Type of TEDS supported	IEEE 1451.4 TEDS Class I
Internal master timebase(f_M)	
Frequency	13.1072 MHz
Accuracy	±50ppm maximum
Data rate range (f_s)	
Using internal master timebase	
Minimum	1.652 kS/s
Maximum	51.2 kS/s
Using external master timebase	
Minimum	0.391 kS/s
Maximum	52.734 kS/s
Input coupling	AC/DC (software-selectable)
AC cutoff frequency	
-3dB	0.5 Hz
-0.1dB	4.6 Hz maximum
Input range	±5 V
AC voltage full-scale range	
Minimum	±5 Vpk
Typical	±5.1 Vpk
Maximum	±5.2 Vpk
Common-mode voltage range (AI-to earth ground)	±2 V maximum
IEPE excitation current (software-selectable on/off)	
Minimum	2.0mA
Typical	2.1mA
Power-on glitch	90μ for 10μs
IEPE compliance voltage	19 V maximum

When IEPE sensors are used, to make sure the configuration meets the IEPE compliance voltage range the following equation must be considered:

$$V_{common-mode} + V_{bias} \pm V_{full-scale}$$

Where $V_{common-mode}$ is the common-mode voltage applied to the NI 9234, V_{bias} and $V_{full-scale}$ are respectively the bias voltage and the full-scale voltage of IEPE sensors considered.

The NI 9234 module has been used to read input data coming from accelerometer and from load cell, through BNC cables, as shown in the following figure:

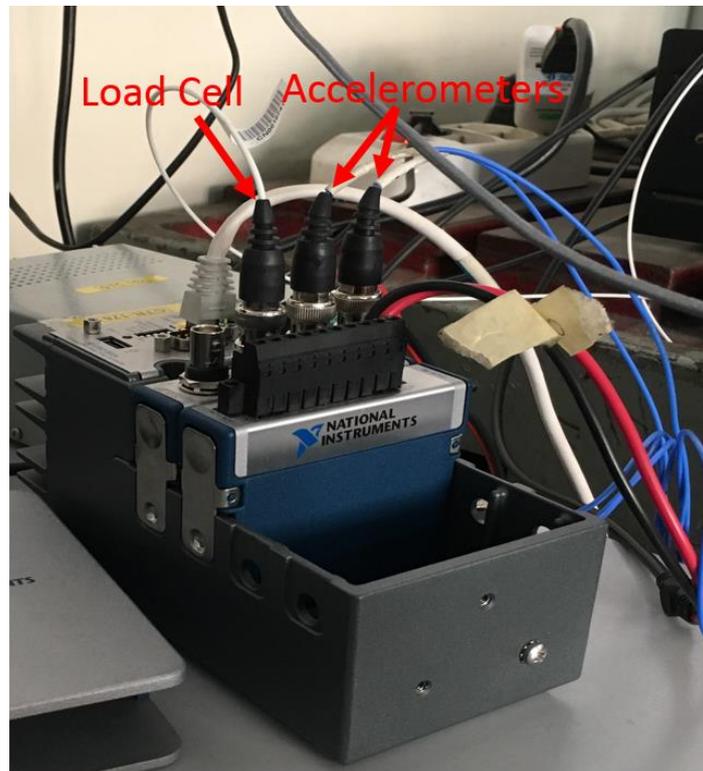


Figure 30: BNC connection of Load Cell and Accelerometers

NI 9263 module

The NI 9263 is an analog output module for any CompactDAQ and CompactRIO systems. Among all characteristics from electric point of view, it is possible to note $\pm 30V$ overvoltage protection, short-circuit protection, low crosstalk and a channel-to-earth ground double isolation barrier for safety and noise immunity. Regards to dynamic behavior the NI 9263 module has fast slew rate, high relative accuracy and NIST-traceable calibration.



Figure 31: NI 9263 module

Each channel has a digital-to-analog converter (DAC) that produces a voltage signal. In the following images is shown the circuitry scheme of one channel.

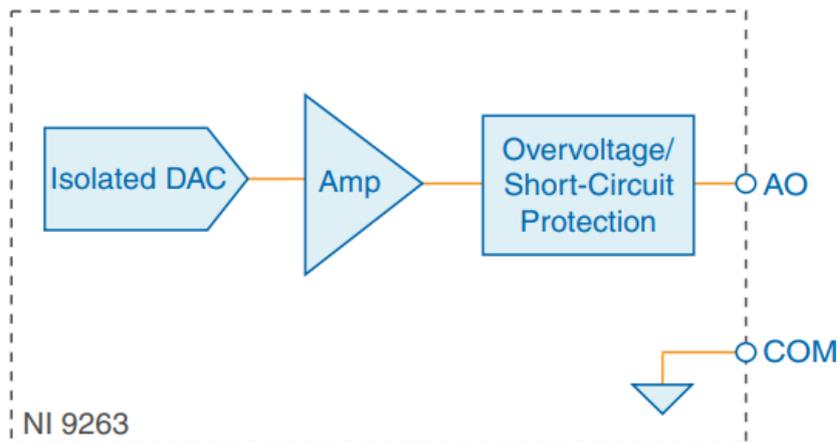


Figure 32: Output Circuitry for One channel of the NI 9263

Output Characteristics

Number of channels	4 analog output channels
DAC resolution	16 bits
Type of DAC	String
Power-on output state	Channel off
Startup voltage	0 V
Power-down voltage	0 V
Output voltage range	
Nominal	± 10 V
Minimum	± 10.4 V
Typical	± 10.7 V
Maximum	± 11 V
Current drive	± 1 mA per channel maximum
Output impedance	2Ω
Stability	
Gain drift	11 ppm/ $^{\circ}C$
Offset drift	110 μ V/ $^{\circ}C$
Protection	
Overvoltage	± 30 V
Short-circuit	Indefinitely
Noise	
Updating at 100 kS/s	600 μ Vrms
Not updating	260 μ Vrms
Slew rate	4 V/ μ s
Crosstalk	76dB
Settling time(100pF load ,to 1LSB)	
Full-scale step	20 μ s
1 V step	13 μ s
Power Requirements	
Power consumption from chassis	
Active mode (at $-40^{\circ}C$)	500mW maximum
Sleep mode	25 μ W maximum
Thermal dissipation (at $70^{\circ}C$)	
Active mode	750mW maximum
Sleep mode	25 μ W maximum

The NI 9263 module thanks to the mentioned characteristic has been used to implement the various signal excitation for the Modal Shaker. In details, this module produces two different stimuli: a sinusoidal sweep and white noise signal.

In the next figure, it is possible to see the connection configuration realized according the pin distribution functionality explained in National Instrument NI 9263 datasheet.

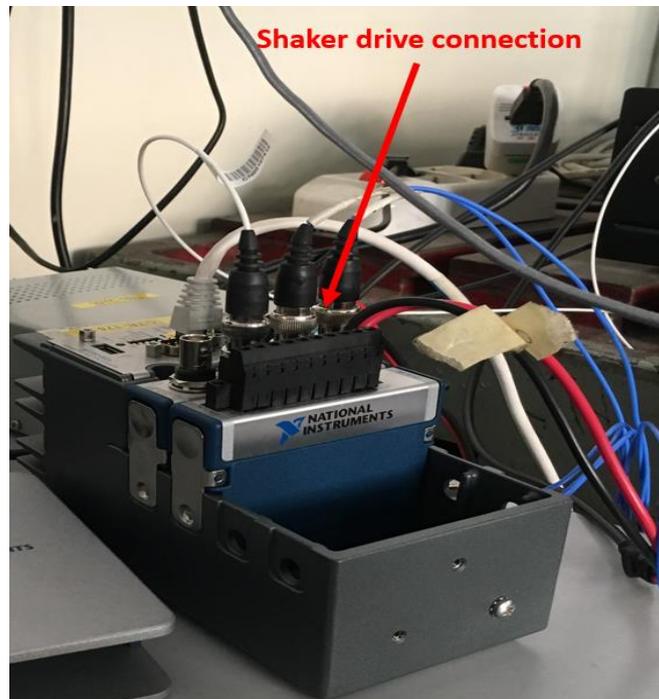


Figure 33: NI 9263 with Modal Shaker driving connection

NI 9237 module

The NI 9237 simultaneous bridge module contains all the signal conditioning required to power up and measure up to four bridge-based sensors simultaneously. This device is characterized by high sampling rate and bandwidth, allowing to obtain high-quality, high speed strain or load measurement system with zero interchannel phase delay. Moreover, NI 9237 has a 60 VDC isolation and 1,000 V_{rms} transient isolation, with high common-mode noise rejection and increased safety for both the operator and test system.



Figure 34: NI 9237 with DSUB and NI 9923 connector

From circuitry point of view, each channel on the NI 9237 has an independent 24-bit ADC and an input amplifier to sample signals coming from all four channels simultaneously.

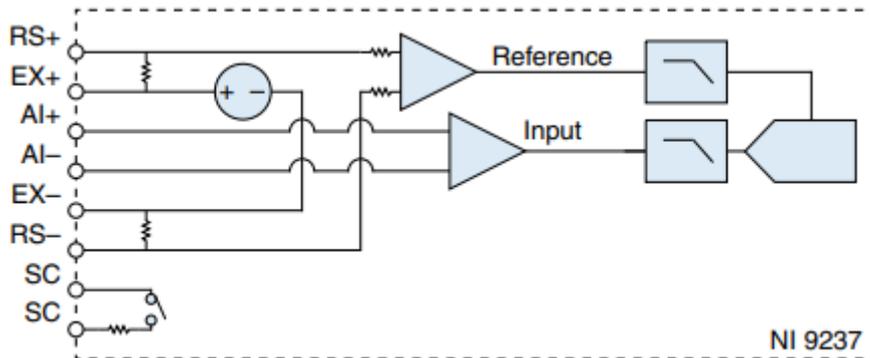


Figure 35: Input Circuitry for each channel of NI 9237

The module is isolated from earth ground, on the contrary there are not isolation among the channels. The connection to one or more sensors the wiring resistance can create errors in bridge circuits, to compensate this phenomenon in the NI 9237 module is implemented two correction mechanism: remote sensing and shut calibration.

The remote sensing continuously and automatically corrects for error in excitation leads, for this characteristic is suitable for half and full bridge sensors. Considering configuration of connection, may happen to manage long wire and small gauge wire with considerably value of resistance, leading to gain error. The gain error depends on voltage

drop caused by the wires that connect the excitation voltage to the bridge. To implement the remote sensing operation, it is necessary remote sense wires to the point where excitation voltage wires connect to the bridge circuit.

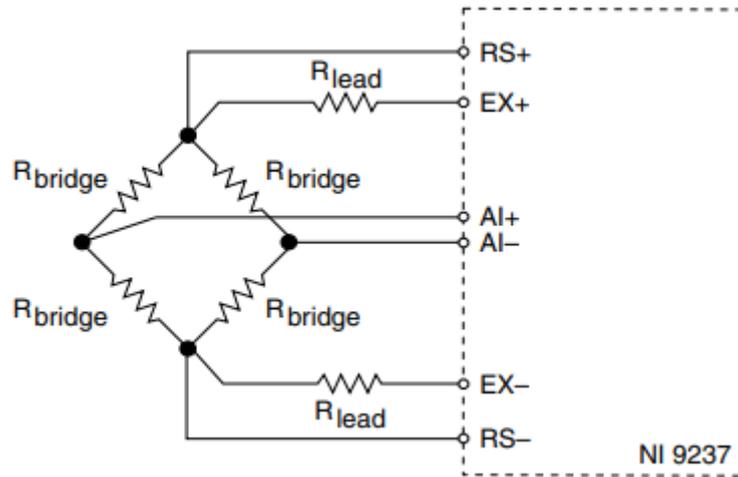


Figure 36: Connection scheme for Remote Sensing

The electric scheme shown before explaining how to realize the connection to take advantage of remote sensing. As result of this, the actual bridge excitation voltage is smaller than the voltage at the $EX+$ and $EX-$ -avoiding the gain error. Indeed, connecting remote sense signals directly to the bridge resistor, the NI 9237 sense the actual bridge voltage and eliminates the gain errors caused by the resistance of the $EX+$ and $EX-$ leads.

Shunt calibration functionality can solve errors from the resistance of both the excitation wiring and wiring in the individual resistors of the bridge. In details, shunt calibration corrects errors caused by wire resistance within an arm of the bridge. Shunt calibration is commonly used with quarter-bridge sensors because they may have significant resistance in the wiring to the active resistor in the bridge.

In the next table is shown a summary of NI 9237 module specification:

Input Characteristics	
Number of channels	4 analog output channels
Bridge completion	
Half and Full	Internal
Quarter	External
ADC resolution	24 bits
Type of ADC	Delta-Sigma (with analog prefiltering)
Sampling mode	Simultaneous
Internal master timebase (f_M)	
Frequency	12.8 MHz
Accuracy	± 100 ppm maximum
Data rate range (f_s) using internal master timebase	
Minimum	1.613 kS/s
Maximum	50kS/s
Data rate range (f_s) using external master timebase	
Minimum	391 S/s
Maximum	51.36kS/s
Data rates (f_s)	$(f_M \div 256) \div n$, where $n=1,2,\dots,31$
Typical input range	± 25 mV/V
Scaling coefficient	2.9802 nV/V per LSB
Overvoltage protection between any two pins	± 30 V
Phase nonlinearity	
$f_{in} = 0$ to 1 kHz	$<0.001^\circ$
$f_{in} = 0$ to 20kHz	$\pm 0.1^\circ$
Input delay	$(40 \div 5/512) / f_s + 4.5\mu\text{s}$
Passband	
Frequency	$0.45 \cdot f_s$
Flatness	0.1 dB maximum
Stopband	
Frequency	$0.55 \cdot f_s$
Rejection	100 dB
Alias-free bandwidth	$0.45 \cdot f_s$
Oversample rate	$64 \cdot f_s$
Rejection at oversample rate	
Excitation noise	100 μ Vrms
Crosstalk (not including cable effects)	
$f_{in}=1$ kHz	110 dB
$f_{in}=10$ kHz	100 dB
Excitation	
Internal voltage	2.5 V, 3.3 V, 5.0 V, 10.0 V
Internal power	150mW maximum

External voltage	2 V to 10 V
Shunt calibration	
Resistance	100 kΩ
Resistor accuracy	
25°C	± 110 Ω
-40°C to 70°C	±200 Ω
Power Requirements	
Power consumption from chassis	
Active mode	740 mW maximum
Sleep mode	25 μW maximum
Thermal dissipation (at 70°C)	
Active mode	740 mW maximum
Sleep mode	25 μW maximum

The NI 9237 module has been used for reading data coming from strain gauge placed on the specimen analyzed. According to considered strain gauge characteristics, a Half Bridge configuration is chosen.

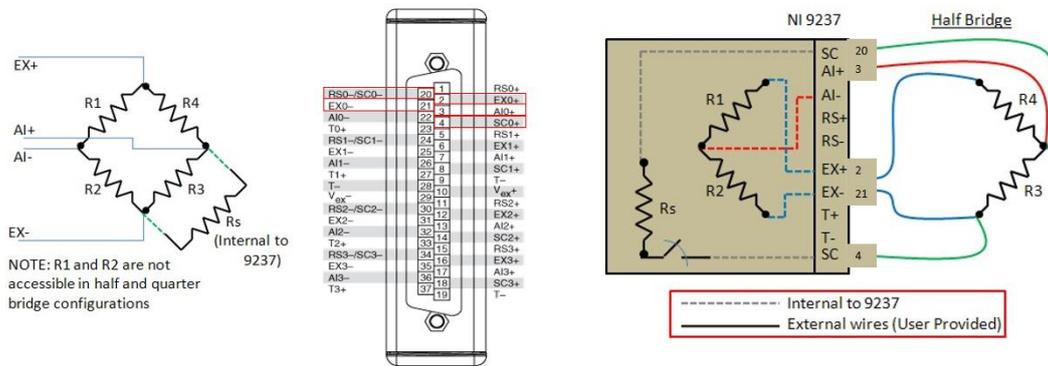


Figure 37: Half Bridge configuration connection

The half bridge configuration leads to obtain the following equation valid Output of the bridge ϵ_3 , ϵ_4 and the output voltage AI

$$AI = \frac{1}{4} \left(\frac{\Delta R_3}{R_3} - \frac{\Delta R_4}{R_4} \right) EX = \frac{1}{4} (\epsilon_3 - \epsilon_4) \rightarrow \begin{cases} \epsilon_3 = \frac{4AI}{EX}, \text{ if } \frac{4AI}{EX} > 0 \\ \epsilon_4 = -\frac{4AI}{EX}, \text{ if } \frac{4AI}{EX} < 0 \end{cases}$$

Signal	Description	9923 pins	Wire color
AI+	Positive analog input signal connection	3/7	White
AI-	Negative analog input signal connection		
EX+	Positive sensor excitation connection	2/6	Brown
EX-	Negative sensor excitation connection	21/25	Green
RS+	Positive remote sensing connection	1	Grey
RS-	Negative remote sensing connection	20	Yellow
SC+	Positive shunt calibration connection	4	
SC-	Negative shunt calibration connection	20	Yellow
T+	TEDS data connection		
T-	TEDS return connection		
V _{ex+}	Positive external excitation source connection		
V _{ex-}	Negative external excitation source connection		

AMPLIFIER

The analog signal providing by NI 9263 is characterized by low energy. As result of this, it is necessary to amplify the signal to obtain a suitable response from Modal Shaker actuator and perform the desired analysis. For this purpose, has been chosen a Dongling linear power amplifier PA-1200 to produce a enough high energy of analog signal.



Figure 38: Dongling power amplifier PA-1200

The technical characteristics of the amplifier are summarized in the following table:

AMPLIFIER	SPECIFICATIONS
Rated power (10 Ohms)	1200 W
Frequency Response	5-20kHz \pm 2dB
Signal-to-noise ratio	>90dB
Power supply	220VAC 50Hz
Power consumption	2400 W
Weight	20 Kg
Dimension(L×W×H)	480×740×132 mm

MODAL SHAKER

As explained in the previous paragraph the test bench system considered has a Modal Shaker to execute testing operation. In detail, to excite the vibration on the specimen has been selected a Modal Shaker model ESD-045. This model has a permanent magnet inside, which allows it to be driven directly by the power amplifier analog signal and reduce the heat generation, thanks to air-cooling system equipment.



Figure 39: Dongling ESD-045 Modal Shaker

It is possible resume the technical specification of Modal Shaker shown in the previous picture, in a table represented below:

MODAL SHAKER	SPECIFICATIONS
Sine force(peak)	450 N
Usable Frequency Range	5 to 3000Hz
Max. Displacement	25 mm
Max. Velocity	1,6 m/s
Max. Acceleration	1000 m/s ²
Effective Mass of Moving Elements	450 g
Weight	25 kg
Dimension(L×W×H)	239×152×220 mm
Power Amplifier Model	PA-1200
Cooling Type	Air-Cooled
Power Supply Requirement	AC 220 V ± 10%, 50 Hz, 1300 VA
Working Environment Requirement	Temperature Range 0-40°C
	Humidity range ≤ 80%

The Modal Shaker is linked to specimen through a clamping structure, composed by two metallic supports realized according to custom specification given by actuator characteristics. The specimen is kept between two supports through four screws, nuts and washers. One of the supports has a groove in its inner face, where the specimen is placed, with a thickness smaller than the specimen's, in order to guarantee a suitable fixation. In the next image, it is possible to see the clamping structure described:

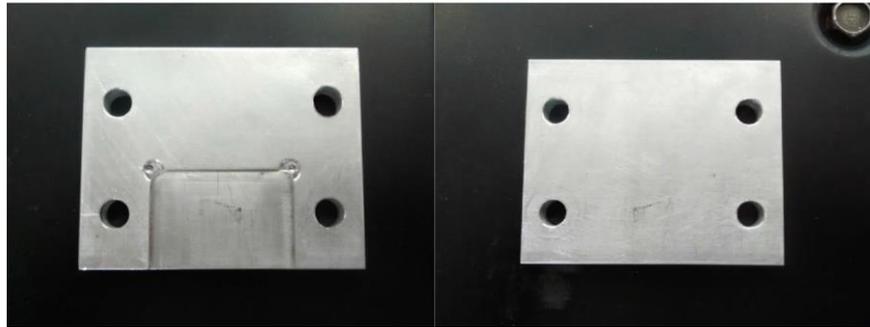


Figure 40: Clamping structure

Through the clamping structure, an alloy specimen can be attached to the Modal Shaker; it is made with aluminum alloy 6182-T4, commonly used in many industrial applications. The geometry has been chosen considering the state of the art coming from previous studies.



Figure 41: Alloy specimen geometry

The specimen shape shown before is useful to understand what the most stressed concentration points among all possible zones are. Indeed, it is placed closest to the clamped end, characterized by a high deformation when the specimen is excited across the second modal frequency. In the next figure, it is illustrated the most stressed specimen area, in which after a certain amount of fatigue stress test the specimen goes to break.

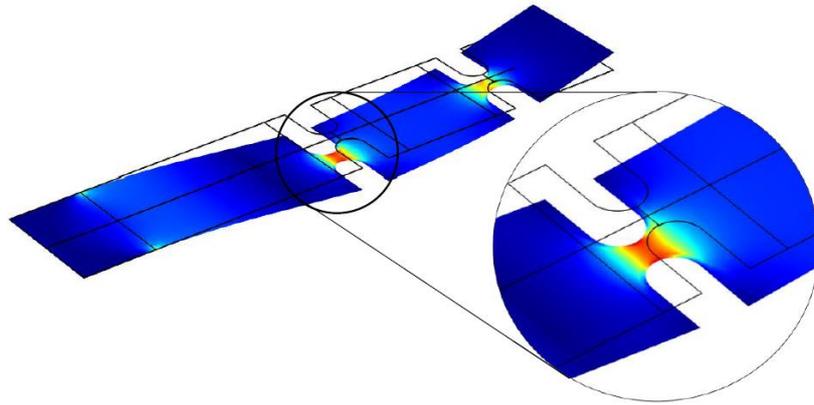


Figure 42: Second modal frequency specimen stress point

ACCELEROMETERS

The accelerometers are devices that measure proper acceleration of a body., which is the physical acceleration experienced by an object. It is thus acceleration relative to a free-fall, or inertial, observer who is momentarily at rest relative to the object being measured. At any point in space-time the equivalence principle guarantees the existence of a local inertial frame, and an accelerometer measures the acceleration relative to that frame. Usually the acceleration is denoted as g-force in relationship to standard gravity.

Simplifying the operating principle of accelerometer, it behaves as a damped mass on a spring. When the accelerometer experiences an acceleration, the mass is displaced to the point that the spring can accelerate the mass at the same rate as the casing. The mainly accelerometer used in engineering applications are piezoelectric, piezo resistive and capacitive to obtain a conversion of the mechanical motion into an electrical signal.

To implement the desired test bench performance, has been chosen piezoelectric ICP® sensors. This accelerometer type is characterized by a mass displacement detection, as an electric signal generate from a piezoelectric crystal when a compression force is applied on it. Conceptually, in piezoelectric accelerometer the mass is suspended above piezoelectric crystal, represented both sensor and elastic element. When an acceleration

occurs, the mass compresses the crystal generating an electric signal proportional to the compression.

According the crystal physical characteristics, the piezoelectric accelerometer presents:

- Considerably low sensitivity
- Very high acceleration detection without damaged
- Impossibility to measure time constant acceleration

Piezoelectric accelerometers, due to the mentioned characteristics are used in vibration and mechanical shock application where occur dynamic accelerations. As results of this, two units of Triaxial ICP® Accelerometer of PCB PIEZOTRONICS has been select to implement test bench system and to acquire acceleration specimen information.



Figure 43: PCB TLB356A12 Accelerometer

The most important technical specifications of PCB TLB356A12 accelerometer are summarized in the next table:

Performance	
Sensitivity ($\pm 10\%$)	100mV/g
Measurement Range	± 50 g pk
Frequency Range	
($\pm 5\%$)	0.5 to 5000 Hz
($\pm 10\%$)	0.4 to 6000 Hz
Resonant Frequency	≥ 25 kHz
Broadband Resolution (1 to 10,000 Hz)	0.0002 g
Non-Linearity	$\leq 1\%$
Transverse Sensitivity	$\leq 5\%$

Environmental	
Overload Limit (Shock)	± 5000 g pk
Temperature Range (Operating)	-54 to +77°C
Base Strain Sensitivity	0.001 g/μ ϵ
Electrical	
Excitation Voltage	18 to 30VDC
Constant current excitation	2 to 2mA
Output Impedance	≤ 200 Ohm
Output Bias Voltage	8 to 12 VDC
Discharge Time Constant	1 to 3 sec
Settling Time (within 10% of bias)	<10 sec
Spectral Noise	
(1 Hz)	50μg/√Hz
(10 Hz)	20μg/√Hz
(100 Hz)	5μg/√Hz
(1 kHz)	2μg/√Hz
(10 kHz)	1μg/√Hz

STRAIN GAUGE

A strain gauge is a device used to measure strain on an object subjected to deformations. Usually, it is made of an insulating flexible backing which supports a metallic foil pattern. The gauge is attached to the object through a suitable adhesive, such as cyanoacrylate. Object deformation causes a change in electric resistance, measured using a Wheatstone bridge and considering the known quantity called gauge factor.

The gauge factor GF is defined as:

$$GF = \frac{\Delta R/R_G}{\epsilon}$$

In the previous relationship ΔR is the change in resistance caused by strain, R_G is the resistance of the undeformed gauge and ϵ is strain. Commonly gauge factor for most popular metallic foil gauges is about 2.

Conceptually, the functionality principle is the following: an excitation voltage is applied to input leads of the gauge network, and a voltage reading is provided from the output

leads. Typical input voltages are 5V or 12V to which the outputs in the order of millivolts correspond. The value of sensor output voltage is measured considering the balanced Wheatstone bridge configuration, in relation to the excitation voltage providing to the sensor.

Two strain gauge, in half Wheatstone bridge configuration, has been chosen to study deformation phenomenon of the specimen. The following picture shown the typical circuit in half bridge configuration:

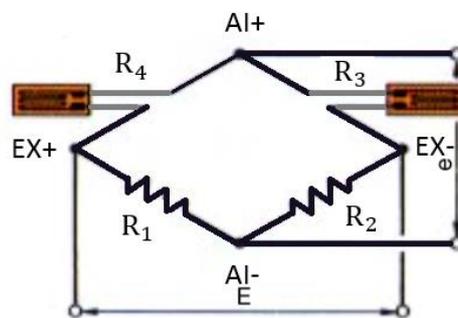


Figure 44: Half bridge strain gauge configuration

The output voltage of the configuration represent above is:

$$AI = \frac{1}{4} \left(\frac{\Delta R_3}{R_3} - \frac{\Delta R_4}{R_4} \right) EX = \frac{1}{4} (\epsilon_3 - \epsilon_4)$$

The two SR4 strain gauge have been installed in half Wheatstone bridge configuration, mounted on opposite face of the specimen as shown in the Figure (). The gauge resistance is $120,0 \pm 0,2$ Ohms for each strain gauge.

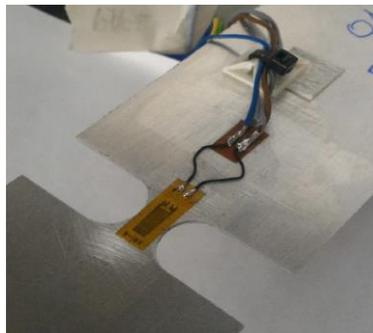


Figure 45: Strain gauge placed on specimen

LOAD CELL

A load cell is a type of transducer, specifically a force transducer. It converts compression, pressure, or torque into a measurable electrical signal. Relating to the force applied to load cell the electrical signal change proportionally. The most common types of load cell used are hydraulic, pneumatic, and strain gauge. According the requirements, piezoelectric load cell type has been chosen. In piezoelectric load cells, the output voltage is generated by the basic piezoelectric material mounted on it and its value is proportional to the deformation of load cell. The piezoelectric effect is dynamic; indeed, the electrical output of a gauge is an impulse function and is not static. As result of this, this load cell type is commonly used in dynamic loading conditions.

An ICP® force sensor model 208C03 of PCB Piezotronics is the model chosen to measure the force exchanged between modal shaker and clamping structure where the specimen is attached.



Figure 46: PCB Model 208C03

ICP® force sensors incorporate a built-in MOSFET microelectronic amplifier. Thanks to this, the high impedance charge output is converted into a low impedance voltage signal for analysis or recording. The mentioned sensors typology, are powered from a separate constant current source, operate over long ordinary coaxial or ribbon cable without signal degradation. The low impedance voltage signal is not affected by triboelectric cable noise or environmental contaminants. Series 200 Impact Sensors are designed to measure

compression and impact forces from a fraction of one lb(N) to 50,000 lbs (to 22.4 kN). The flat sensing surface located on the top of the sensor is designed to measure a dynamic force quickly applied axially to the sensor. Applications include drop testing, machinery studies, punching and forming operations, tensile testing, fatigue testing, fracture analysis, and materials testing.

The PCB 208C03 sensor has the following technical specifications:

Performance	
Sensitivity ($\pm 15\%$)	2248 mV/kN
Measurement Range (Compression)	2.224 kN
Measurement Range (Tension)	2.224 kN
Maximum Static Force (Compression)	13.5 kN
Maximum Static Force (Tension)	2.224 kN
Broadband Resolution (1 to 10,000 Hz)	0.02 N-rms
Low Frequency Response (-5%)	0.0003 Hz
Upper Frequency Limit	36,000 Hz
Non-Linearity	$\leq 1\%$
Environmental	
Temperature Range	-54 to +121°C
Temperature Coefficient of Sensitivity	$\leq 0.09\%/^{\circ}\text{C}$
Electrical	
Discharge Time Constant (at room temp)	≥ 2000 sec
Excitation Voltage	20 to 30 VDC
Constant Current Excitation	2 to 20 mA
Output Impedance	≤ 100 Ohm
Output Bias Voltage	8 to 14 VDC
Spectral Noise (1 Hz)	0.000942 N/ $\sqrt{\text{Hz}}$
Spectral Noise (10 Hz)	0.000488 N/ $\sqrt{\text{Hz}}$
Spectral Noise (100 Hz)	0.000173 N/ $\sqrt{\text{Hz}}$
Spectral Noise (1000 Hz)	0.0000385 N/ $\sqrt{\text{Hz}}$
Output Polarity (Compression)	Positive
Physical	
Stiffness	1.05 kN/ μm
Size (Hex \times Height \times Sensing Surface)	15.88 mm \times 15.88 mm \times 12.7 mm
Weight	22.7 gr
Housing Material	Stainless Steel
Sealing	Hermetic
Electrical Connector	10-32 Coaxial Jack
Electrical Connection Position	Side
Mounting Thread	10-32 Female

All the illustrated hardware component has been placed on the specimen under test according suitable order, to obtain desired information and data from sensors. The following figure shows devices distribution:

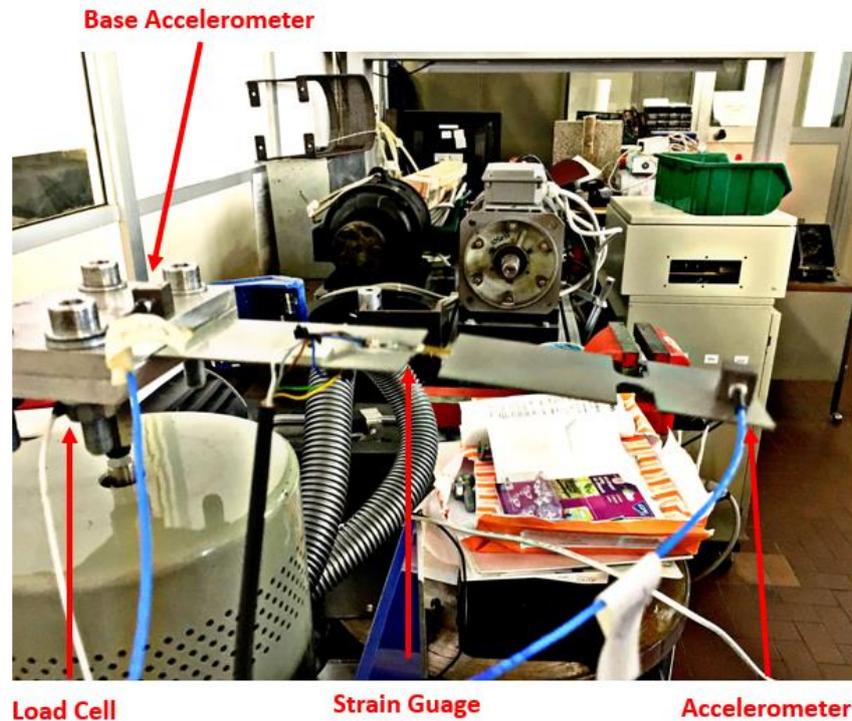


Figure 47: Devices distribution on the specimen

3.3 Data acquisition software

The hardware test bench system previously described has been management using data acquisition and control software developing entirely in NI LabVIEW environment. The reason for this choice is justified by the huge possibility of configuration of the hardware provided by official National Instrument devices. Thanks to this, has been possible to manage all hardware regarding to acquisition of all data coming from sensors.

LabVIEW general overview

Laboratory Virtual Instrument Engineering Workbench (LabVIEW) is a system-design platform and development environment for a visual programming language elaborated from National Instruments, called language “G”. LabVIEW is commonly used for data

acquisition, instrument control and generally all the application regarding industrial automation.

It is possible to define software running operation in LabVIEW, such as Dataflow programming based on data availability. In other words, if there is enough data available to a subVI or function, that sub VI or function will execute. The execution flow depends on the structure of graphical block diagram, representing the LabVIEW-source code. Connecting various function-nodes by wires let to propagate variables to any node and each one can be executed as soon as all its input data become available. According to this, many nodes are processed simultaneously giving the peculiar LabVIEW characteristic: the parallel execution of functions and instructions composing the programming. The software is equipped by a built-in scheduler, useful to manage efficiently and automatically exploited the multi-processing and multi-threading hardware.

LabVIEW graphical programming is based on exploiting elements called virtual instruments (VIs), representing programs-subroutines. Each VI is composed by three parts: a front panel, a block diagram and a connector pane (used to represent the VI in the block diagram of other, calling Vis).

- The front panel is composed of controls and indicators. The controls represent inputs to the software allowing to the user to supply information to the VI. The outputs are represented by indicators; they give a response based on the inputs provided to VI.
- The block diagram is placed in the back panel, where it is contained the graphical source code and the structures and functions operations on controls and supply data to indicators. The objects added at the front panel corresponds to terminals on the back panel.
- Connector pane is used to represent the VI in the block diagram of other, calling VIs.

Controls, indicators, structures, and functions are referred to as nodes. Nodes are connected to one another using wires. As consequence of this, a virtual instrument can be run as either a program, with the front panel serving as a user interface, or, when dropped

as a node onto the block diagram, the front panel defines the inputs and outputs for the node through the connector pane. This implies each VI can be easily tested before being embedded as a subroutine into a larger program. In the LabVIEW programming environment, are included examples and documentation, making creation of small applications quite simple. On the contrary, to realize a high-quality G programming it necessary to acquire a good expertise. Indeed, for complex algorithms or large-scale code is fundamental to have the knowledge of the special LabVIEW syntax and the topology of its memory management. Thanks to this, it is possible to create advanced LabVIEW development systems that can implement stand-alone applications.

Usually, specific architectures, known as design patterns, are used in LabVIEW to design applications. The design patterns commonly available for graphical application are:

- Functional Global Variable to exchange information without using global variable, reducing the number of wiring in the block diagram. Negative aspect is that all VIs interested are kept in memory.
- State machine is used to control execution of the block diagram depending on past events. State machine is implemented by a case structure inside a while loop pass an enumerated variable to a shift register, representing the next state. This type of design pattern realizes user interface; on the other hands all possible state must be known in advance.
- Event-driven user machine is characterized by lossless processing of user actions. It is based on capturing GUI events in an event structure queue, inside a while loop. This one is suspended by the event structure and resumed only when the desired events are captured.
- Master-slave implements the possibility to run independently more than one processes simultaneously. This design pattern is composed of many while loops, put in communication with each other. Some run with the function of the “master” controlling the “slave loop”. Basic aspect is to prevent race condition of data used in the loops.
- Producer-consumer commonly used to data sampling and visualization. This mechanism has the purpose to control asynchronous of multithreaded execution

of loops. A master loop controls the execution of two slave loops, that communicate using notifies, queues and semaphores. While, data-independent loops are automatically executed in separate threads.

- Queue state machine with event-driven producer consumer is composed of an event-driven user interface placed inside the producer loop and a state machine inserted in the consumer loop. The two loops communicate themselves and to others parallel VIs by mean of queues.

LabVIEW is equipped with a compiler producing a native code for the CPU platform, translating the graphical code into executable machine code. The executable code runs exploiting LabVIEW run-time engine, that reduce the compiling time and provides a consistent interface to various operating system and hardware components. LabVIEW run-time engine let to have a portable code across different platform.

A significant number of libraries allowing to have many functions regarding data acquisition, signal generation, signal conditioning, mathematical operations and other capabilities used to implement data capture from hardware sensors. In LabVIEW environment, is present a text-based programming component called MathScript, compatible generally with MATLAB®, to implement custom functions for signal processing or analysis.

LabVIEW is compatible with many add-on software to implement various operations. Those of interest for this thesis work are LabVIEW FPGA module and LabVIEW Real-time module.

LabVIEW FPGA module can be used to create FPGA based system with more efficiency and more effectiveness. This is realizable thanks to development environment well integrated, IP libraries, a high fidelity and debug functionality simulator. It is possible to implement VI FPGA with direct access to I/O, to personalize the hardware for various application, i.e., digital protocol communication. Another feature available, is the presence of HIL (Hardware-In-The-Loop) simulation and the rapid control prototyping. LabVIEW FPGA module includes, also, many signal elaborations integrated routine allowing to integrate third part HDL (Hardware Description Language) code and IP.

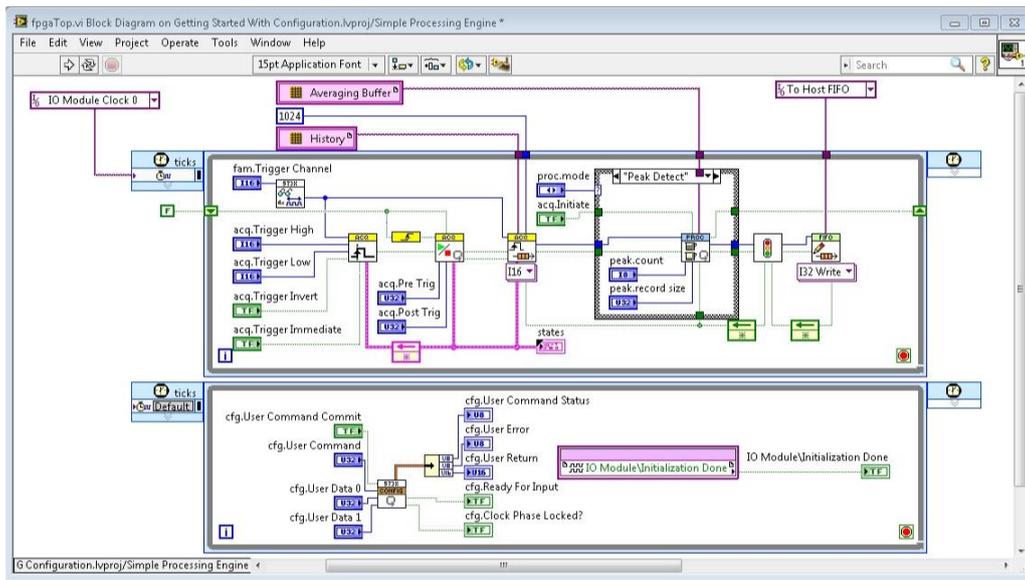


Figure 48: FPGA block diagram example

LabVIEW Real-time module is an add-on software used to implement real-time application to perform test, monitoring and control operations. The LabVIEW real-time module allows to create a complete graphical design to implement stand-alone application for embedded system. It includes integrated instruments for the scheduling of real-time thread and multithreading to write more efficiently deterministic and robust code. The real-time operating system gives the possibility to execute critical application satisfying strict temporization specifications with high degree of reliability.

The realization of a real-time application is composed of two main block diagrams (or tasks) running in two devices that communicate themselves through IP address. The real-time task runs on RT-target, such as *CompactRIO*, to perform acquisition data operations according the defined time dead-line.

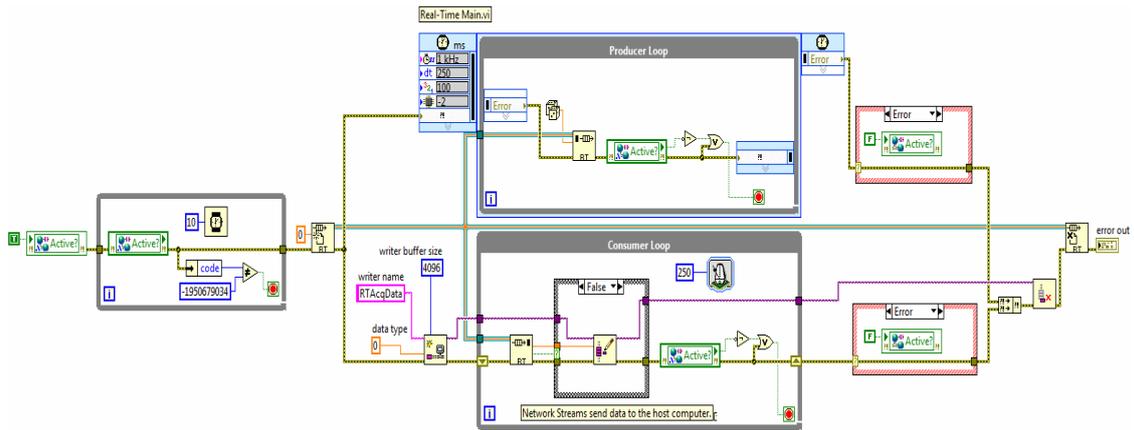


Figure 49: Real-Time task example

As can be possible to see, there is VI with a producer-consumer structure that generates data from the RT target and streams the data to the VI on the host computer.

On the other hands, the acquired data can be managed through the task running on computer host, where the VI reads the data and displays it in a waveform graph. Furthermore, it is possible to visualize them using many functions available in LabVIEW libraries. In the following figure, is shown a block diagram example to implement what previously explained:

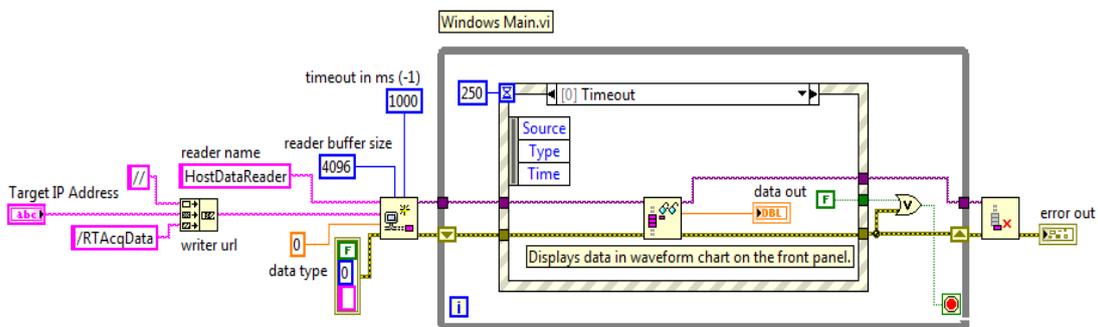


Figure 50: Displaying Real-Time data task

The front panel related to the previous block diagram can be appear in the following general form:

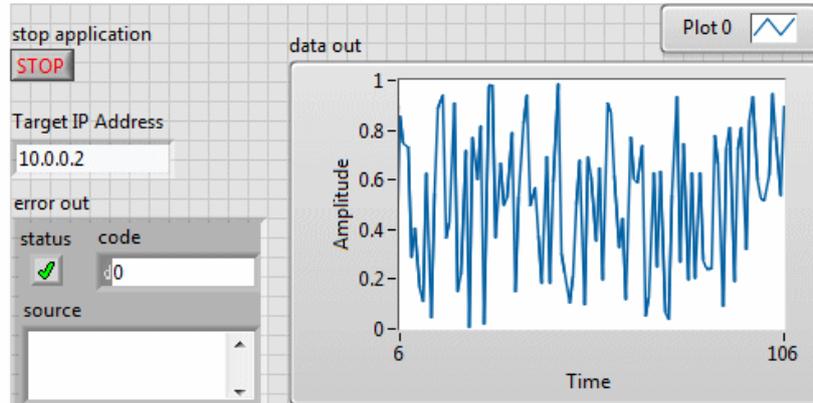


Figure 51: Computer Host Front Panel

Data acquisition software structure

The data acquisition software structure is composed of two main part running in their suitable devices: the first part in the host computer and the second one in Real-Time target device equipped with the FPGA (Field programmable gate array) circuitry, necessary to well manage the sensors placed on the test bench system. The project structure developed for the control of the test system and for data acquisition feature, can be resumed in the following picture:

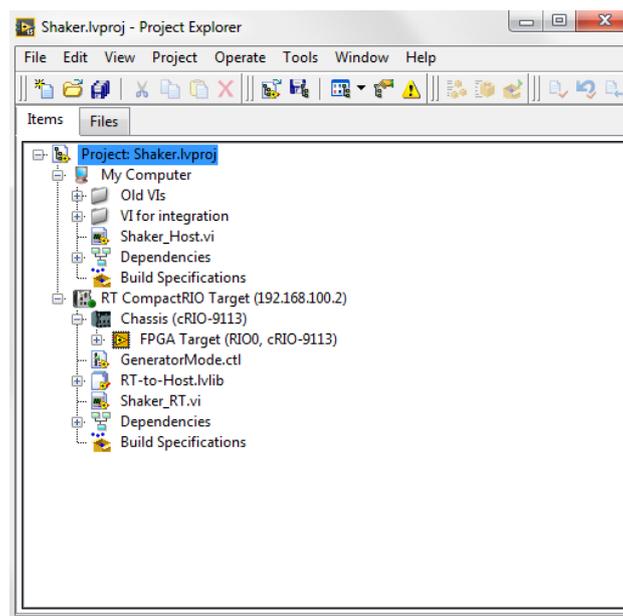


Figure 52: Test bench system LabVIEW project structure

When a new application must develop, LabVIEW create the “project” folder. It contains all the items composing the entire control system software, in this case called “Shaker”. Exploding the “Shaker.lvproj” the project explorer shows all other elements, in which it is possible to see the two main devices interested and where are developed the Vis. The item “My Computer” includes all the Vis running on the host computer, in which is installed LabVIEW software. While “RT CompactRIO Target (192.168.100.2)” is composed by all VIs running in real-time hardware, that is kept in communication with host computer by mean of IP address specified into the brackets. The main properties of VIs and their structure will be explained in the next pages. Following, for simplicity, the visualization order shown in the project explorer.

Inside the “My computer” folder the most important VI is that one called “Shaker_Host”. In this virtual instrument, has been implemented the front panel to display all information coming from real-time target. The following figure shows the structure of the front panel:

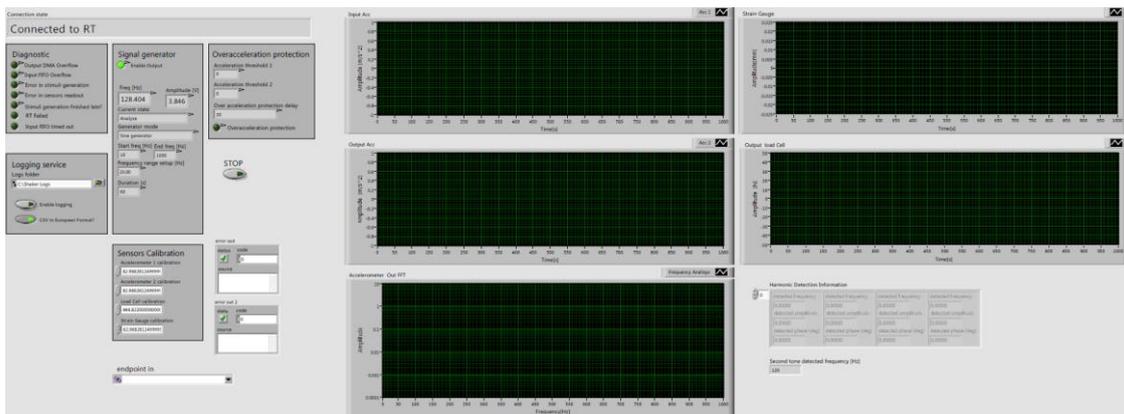


Figure 53: Shaker Host Front Panel

Analyzing the front panel from left to right it is possible to observe:

- Connection State in which is visualized the state of connection with real-time target
- Diagnostic area has been implemented to have information about the status of data exchanged among actuator (Modal Shaker), sensors, real-time target and host computer, activating a green led in case of errors. In details, the led is activated when data overflow occurs in both Output data DMA and Input data. Signal error

for the stimuli generation indicates the satisfaction of the real-time specification, regarding the signal generation data for the desired stimuli to Modal Shaker.

- Logging service is a control input, allowing to user to keep in memory the data acquired. LabVIEW creates an Excel file in .CSV format in the folder settable by user.
- Signal generator box provides the all the setting regarding the signal generation. When the software is running, it displayed the actual state mode (Sweep or Analyze mode), frequency and amplitude of signal and all the information related to Sweep and Analyze mode.
- The over-acceleration protection visualizes if the recorded acceleration of Modal Shaker is above a configurable value, because the software is designed to stop when high acceleration occurs that may damage both Modal Shaker and the under-test specimen.
- Sensors calibration are control inputs to software allowing to user to have information about data acquired from sensors. Indeed, the user can change the numerical value to obtain different representation of data in terms of measure unit and its scale factor. On the other hands, if in the future will use different sensor type characterized by other calibration rules the parameter can easily set improving the software integrability and reusability for other test application.
- The charts show, as signal waveform, the data provided by the corresponding sensor.
- Accelerometer Output FFT graph displays the Fast-Fourier Transformation of waveform coming from accelerometer placed on the specimen extremity. It is useful to find second tone resonance frequency, essential to produce the right stress test signal to specimen.

Regarding the “Shaker_Host” VI block diagram a general overview is shown in the next figure:

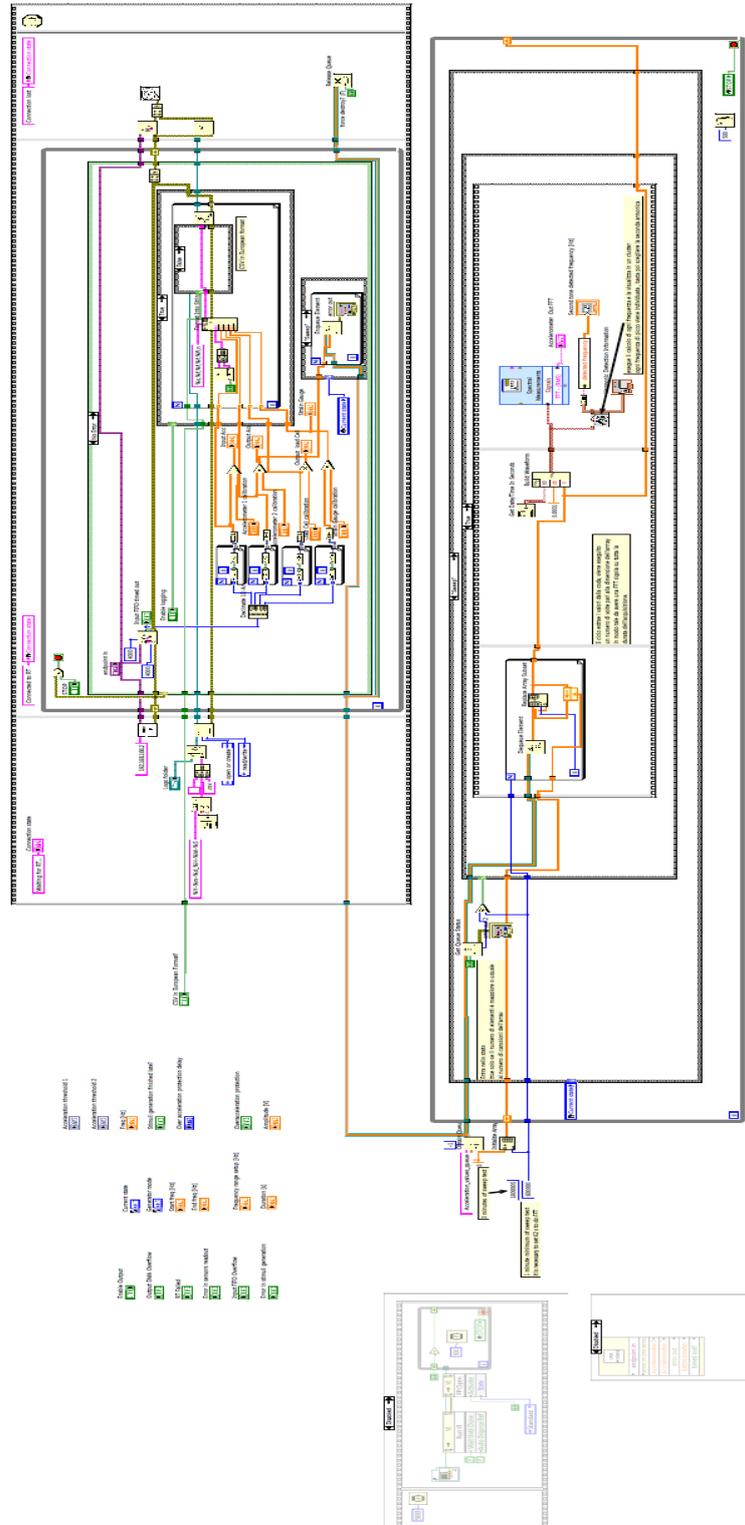


Figure 54: Shaker Host block diagram overview

In the upper block diagram position, it is possible to see the structure used to manage all data coming from real-time target. In details, it is possible to notice the execution is regulated by “Flat sequence structure” to manage the priority of code execution: the operations are performed from left to right according the sub diagrams composing the structure. Data of the next frame are available only when the previous sequence operations are completed, as result of this the code is executed in sequential manner. In the lower part of block diagram is present the structure linked to actual execution state of test process, according to the actual control system state different instructions is performed.

The Flat sequence structure is composed of four frame: the first contains data saving code allowing the user to select the folder and the format in which data have to stored; the second part manages chart visualization of data coming from the all sensors, data provided by accelerometer positioned on the specimen is managed by a queue structure able to link the upper with the lower block diagram; the third frame performs formatting instructions related to previous structure in the flat sequence; the last part contains stop instruction.

The lower code section implements a state machine linked to the four possible state of the control system. A timed while loop implements the storing data operation related the queue created before. The length of the queue is set according a suitable number of element acquired from the accelerometer, considering the acquisition rating is set to *10k samples/second*. In detail, the maximum time for the sweep analysis has been chosen in three minutes ensuring a sufficiently high number of samples to perform the signal elaboration operation. The queue provides acquired data to block instruction implementing the FFT (Fast Fourier Transformation) of the signal, with a sufficiently number of samples to guarantee a good resolution of FFT computation. As shown in the previous paragraph, the resolution depends on sampling frequency and the number of samples generated. The following relation is useful to compute the FFT resolution:

$$\Delta f = \frac{\text{Frequency}}{\text{Samples}}$$

Where Δf is the FFT resolution, frequency data acquisition is *10k samples/second* and samples are *600000 samples* related to 1 minute of sweep test data acquisition:

$$\Delta f = \frac{10 \times 10^3}{600000} \text{Hz} \cong 0.016 \text{Hz}$$

This means that it is possible to detect the resonance frequency with high precision. After FFT computation, the second tone resonance frequency is detected and then it is used to set centered frequency for random white noise generation.

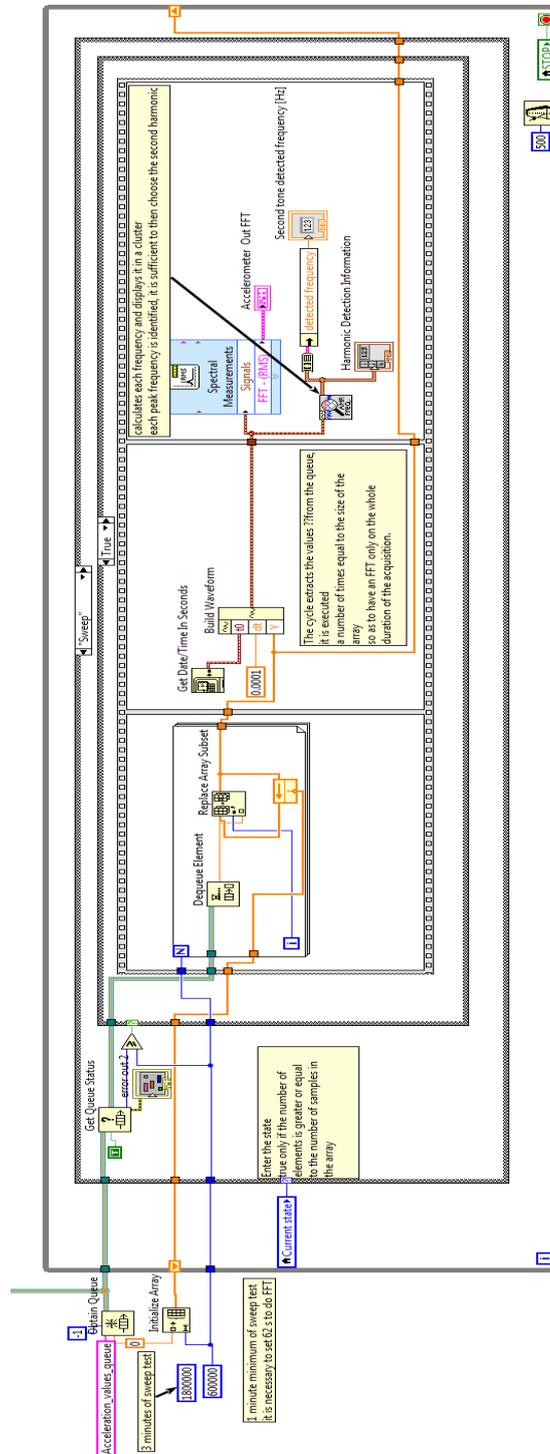


Figure 56: Shaker host lower block diagram

All the previous described operations are executed only when the current state of the test is in the sweep analysis, allowing to have a good managed and usage of memory and CPU resources to have a low impact on system used as host computer.

The second main section of control system project is represented by Shaker_RT block diagram, in which are managed all the stimuli generation and synchronization operation between the variable in output and data input from sensors and their transmission to Shaker_host interface. In the following picture is shown the front panel of Shaker_RT useful to explain the settable parameter by the user to perform different operation according the test procedure chosen.

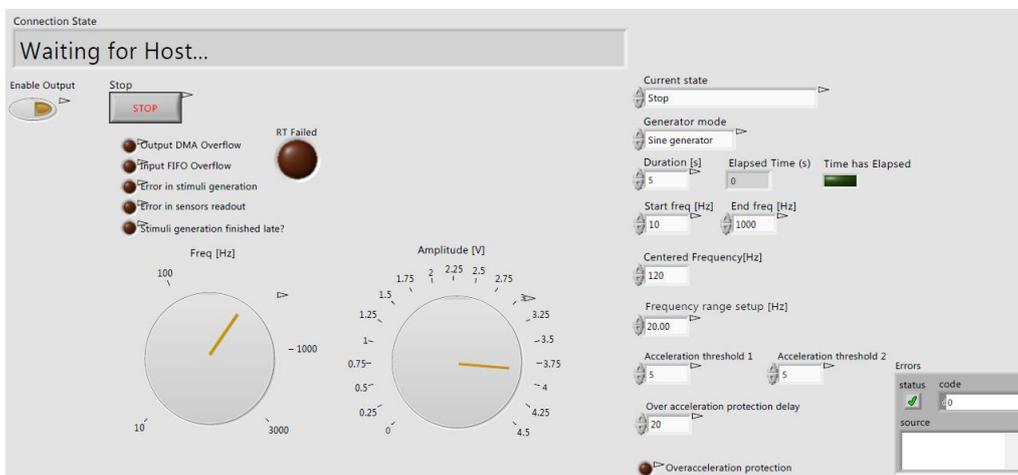


Figure 57: Shaker_RT front panel

The real time module control panel shows the following elements:

- Current state represents the control variable regulating the system operation. It is possible to choose among four states: Stop, Manual, Sweep and Analyse. When running in manual state the stimulus is completely configurable by the user selecting signal type (refers to generator mode control variable), frequency, amplitude and duration. Selecting Sweep state, the system starts to generate a sinusoidal signal with variable frequency during the time necessary to stimulate all resonance frequency of the specimen. Analyse state set the real time module output a white noise signal characterised by frequency and range defined by the user.

- Generator mode is a control input allows to select signal type provided to specimen sine wave and white noise generation mode. The code structure allows to change signal generation mode in when the system is set to manual state, instead in Sweep and Analyse mode the generation mode is set automatically respectively in sine waveform and white noise waveform.
- Duration control variable to set how much time, expressed in seconds, the control system runs in different state. It is necessary to set this parameter before selecting current state machine variable to make the change effective. Near to duration control variable, is placed two indicators showing the time elapsed from beginning of the state (useful when the time of step is set to high value of time) and a led blinking when operation is finished.
- Start and End frequency control allows to set respectively the start and final frequency of time varying sinusoidal signal when the system is in Sweep mode.
- Centred frequency and Frequency range set up are frequency control variable characterizing the white noise signal output generated in Analyse mode able to stress specimen under test according the desired frequency.
- Acceleration threshold is used to set parameter related to protect the Modal shaker from over acceleration that can occur when the configured signal amplitude is too high. These thresholds protect the actuator damage.
- Red led placed on front panel left side indicate some possible errors that can occur during the program execution. They refer to all element regulating real-time operation, in details: DMA data overflow; input FIFO overflow when data input coming from sensors is not synchronized with output variable provided to host computer; “stimuli generation finish late?” indicate if the elements elaborated by the code respect the task time dedicated to it.
- Stop button allows to abort code execution.
- Enable output push button is a control signal allowing to active the signal in output when the control system is set to manual, otherwise the functionality is activated automatically.

The block diagram related to Shaker_RT is composed of multiple section performing various operation, as it is possible to see by the description of front panel. To explain

the most important code related the program section some pictures is used. The first section consider, is related the signal generator manager task: it performs by means of state machine the operational mode of control system. Indeed, it is composed of an external while loop structure able to capture the control variables values coming from user setting and a case structure related the inner part that performs the operation according the operational mode chosen. The next figure illustrates the behaviour of the software when it is set in sweep mode (*current state* control variable) and how signal is generated in terms of time frequency variability related to sweep test duration set.

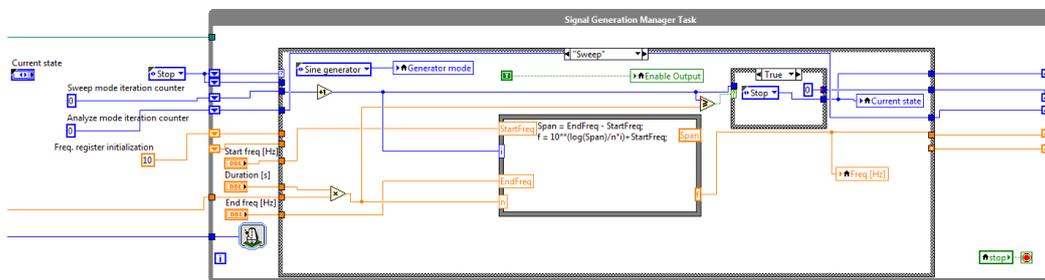


Figure 58: Signal Generator manager sweep mode

When in Sweep state, the case structure shows the operation for the sinusoidal signal elaboration in time. It is implemented using the *Formula node* programming structure, allowing to perform complex mathematical operation and other operation (e.g. while loop, for loop etc.) using text base programming. The signal frequency increases during the time following the logarithmic function, as shown in the following figure:

In details, the relationship regulated the sampled for sinusoidal sweep signal is:

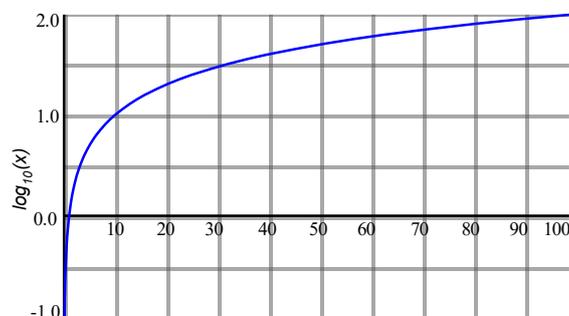


Figure 59: Logarithmic function distribution

$$f_i = i \cdot \frac{\log_{10}(f_{end} + f_{start})}{N}$$

Where i is the actual point frequency, N is the total number point requested by the span that is equal to the difference between start frequency (f_{start}) and (f_{end}). The relationship is used to compute the actual frequency to produce the sinusoidal signal, indeed the local variable ***Freq*** is created and its value is continuously update inside thanks to the while loop. The while loop, is used to manage, also, another control variable called ***Generator mode*** setting it according the actual state of ***Current state***. The local variables is then available and usable in all code sections.

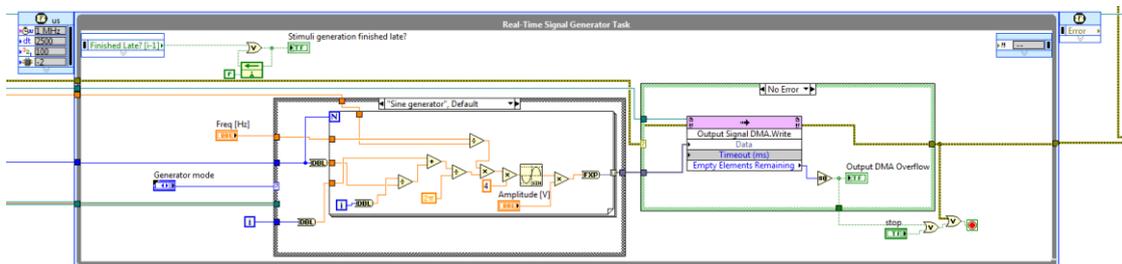


Figure 60: Sine real-time generator task

The previous figure, shows the real-time signal generator task to implement the sine waveform: the ***Generator mode*** control variable is set to Sine generator, the ***Freq*** variable is used to produce the sine waveform with the actual frequency with a given module defined by the ***Amplitude*** variable chosen by the user in the front panel.

In the Analyze case state of Signal Generation, the code instruction implements the timing regulation of the white noise signal according the Duration control variable set by the user. In details the signal is synchronized with the actual time instant in which the signal start producing by the real-time module, setting ***Generatore mode*** variable to Filter white noise generator. The program compute through the function “Elapsed time” the time elapsed from the initial time instant, providing the value for boolean variable “Time has Elapsed”. When its value became true, the actual state of the real-time module is set to Stop and the signal white noise generation is run out.

frequency and the Butterworth then filter is used to implement band pass filter with cut frequency equal to:

$$f_{cut} = f_{centred} \pm f_{std}$$

Where f_{std} is the value frequency set in **standard deviation** variable by the user. The Butterworth filter has been selected, because it guarantee a flat response in band-pass and a good attenuation of the signal outside the cutting frequency. The behavior of this filter type is similar to ideal filter with a marginal reduction of signal amplitude output respect to the input.

A for loop function is used to load filtered white noise array elements to the RT FIFO, writing sequentially the element inside it. Then the element is sent to Real-Time Signal Generation Task, following the rule of first in first output element to guarantee the element order trasmission correctness. As result of this, the signal constructed in real time task is accurately produced to obtain the signal output as user expected. In the next figure, it is possible to see the structure implementing the filter white noise generation instruction:

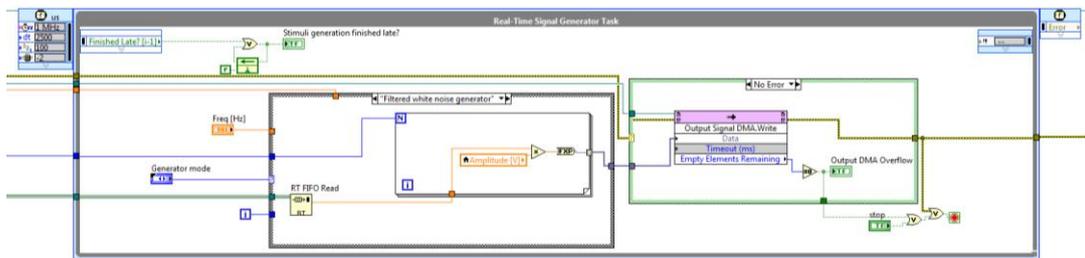


Figure 62: Real-Time Signal Generator Task for filtered white noise

The Real-Time Signal Generator Task for both signal generation is implemented using a Time Loop structure to guarantee real-time characteristic of the stimuli provided to actuator. The main parameters to define a Time Loop, as used in this case, are:

- Source name, indicating the timing source used to control the structure
- Period, used to specify a period for while loop according the time unit of Source name
- Priority, representing the priority of the execution of Time Loop respect to the other objects included in the block diagram
- Processor, a parameter useful to define which processor must be handled. The value set of it is -2, in this case LabVIEW automatically manage processor assignment
- Finish late? [i-1], Boolean variable returning true in the actual iteration of the Time Loop if the previous is not completed before the specified deadline

The RT FIFO read instruction provides to the for loop the element coming from Filtered white noise Task, then the **Amplitude** variable is used to regulate the signal gain output from the real-time module.

3.4 Data acquisition software validation

The development of control system ends with the verification of all functionalities required to implement test process. Therefore, some check on the software characteristic has been done before executing test operation on specimens. In details, has been verified the output signal generation provided by real-time module and data input acquisition coming from sensors. Both operations are realized using additional hardware and different connection of sensors and real-time signal module generator. In next figure is shown the hardware configuration:

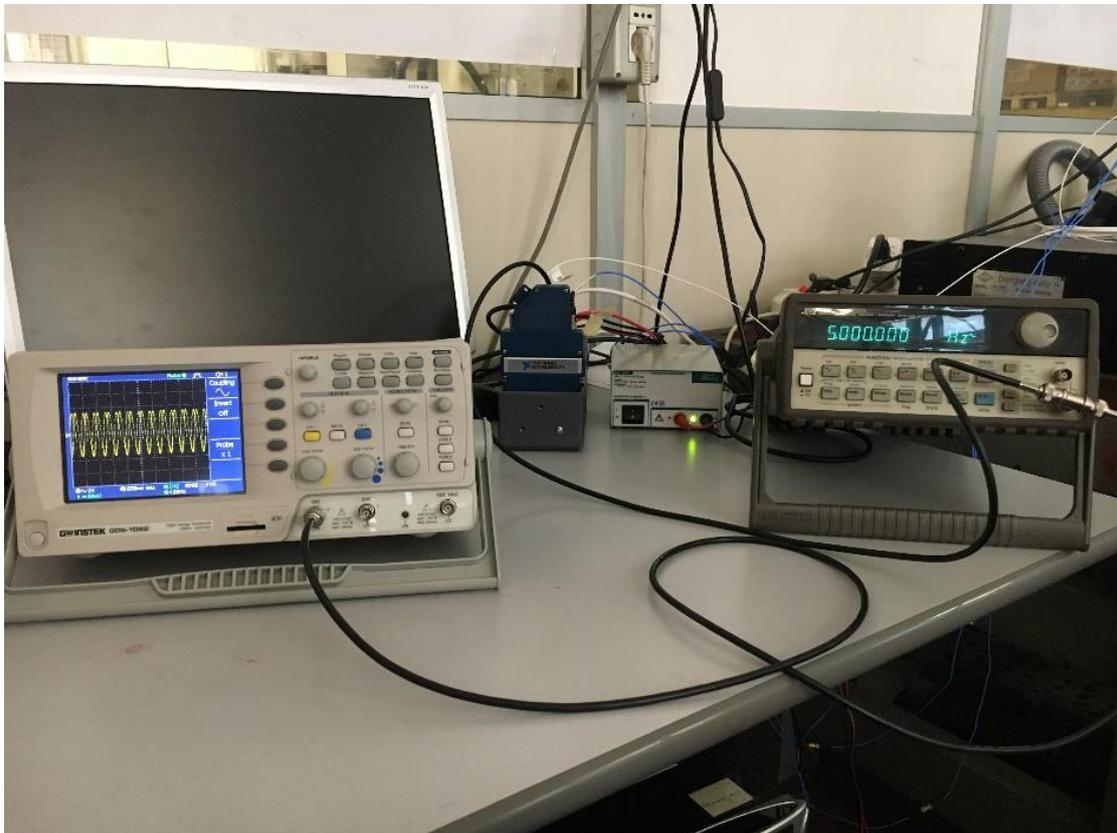


Figure 63: Validation hardware composition

The validation hardware is composed of additional digital storage oscilloscope (GwINSTEK GDS-1022) and function waveform generation (HP3312a), NI modules and computer host. The first step is the verification of output generation of filtered white noise signal. The output signal provided by NI real-time module is connected to oscilloscope to verify the correctness of the signal produced. A series of frequency is set, and the output

is evaluated on the oscilloscope screen. The behaviour of the system at different frequencies is the same, the filtered signal is produced without interruptions in all test time and the noise frequency remain in a limited range according parameter set. In the following pictures it is possible to see the relationship previously described.

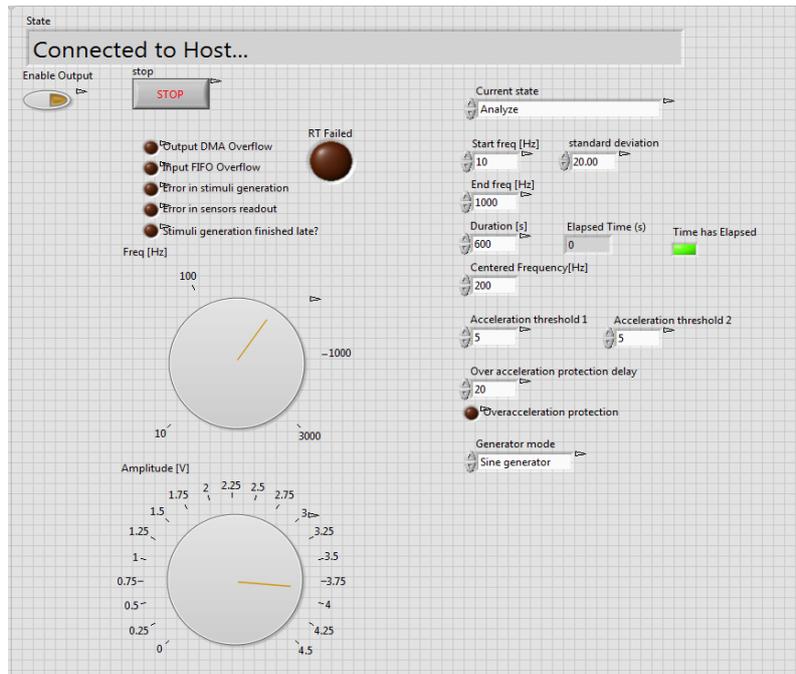


Figure 64: Filtered white noise parameter set

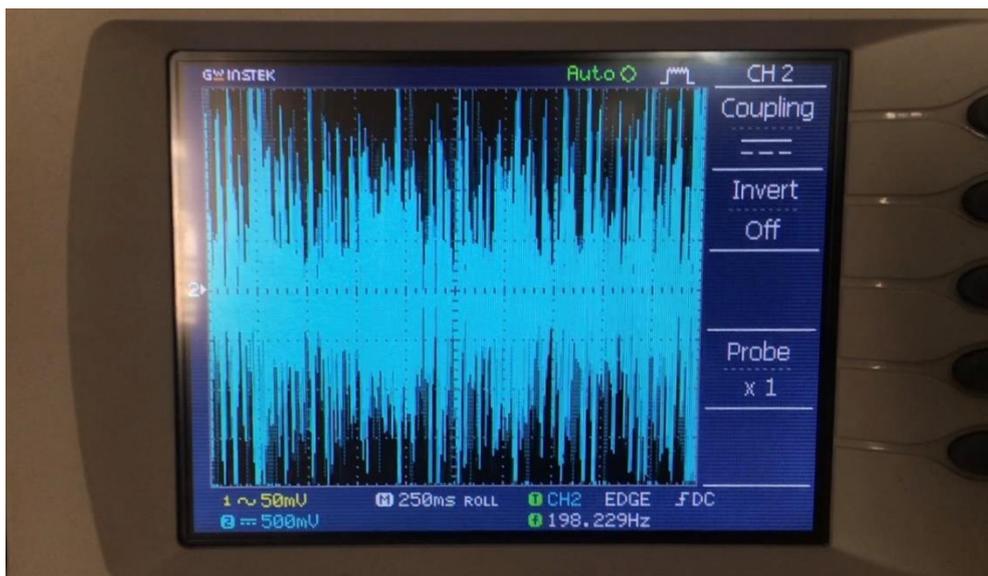


Figure 65: Filtered white noise oscilloscope verification

As shown, the frequency set is 200Hz and the cut of frequency related to it is ± 20 Hz. The oscilloscope screen capture visualizes the correctness of signal produced by National Instrument real-time module.

After the first step the other validation operation regards to entire acquisition process of the control system a different level. The first is related to show in sensor dedicated chart the acquired signal waveform from sensor on computer host, the second regard the code instruction implementing data saving in the export Excel file format. The function waveform generator is set to generate sinusoidal signal; it is linked to input accelerometer channel and to oscilloscope through coaxial cable. In the following image, the connection setting is shown:

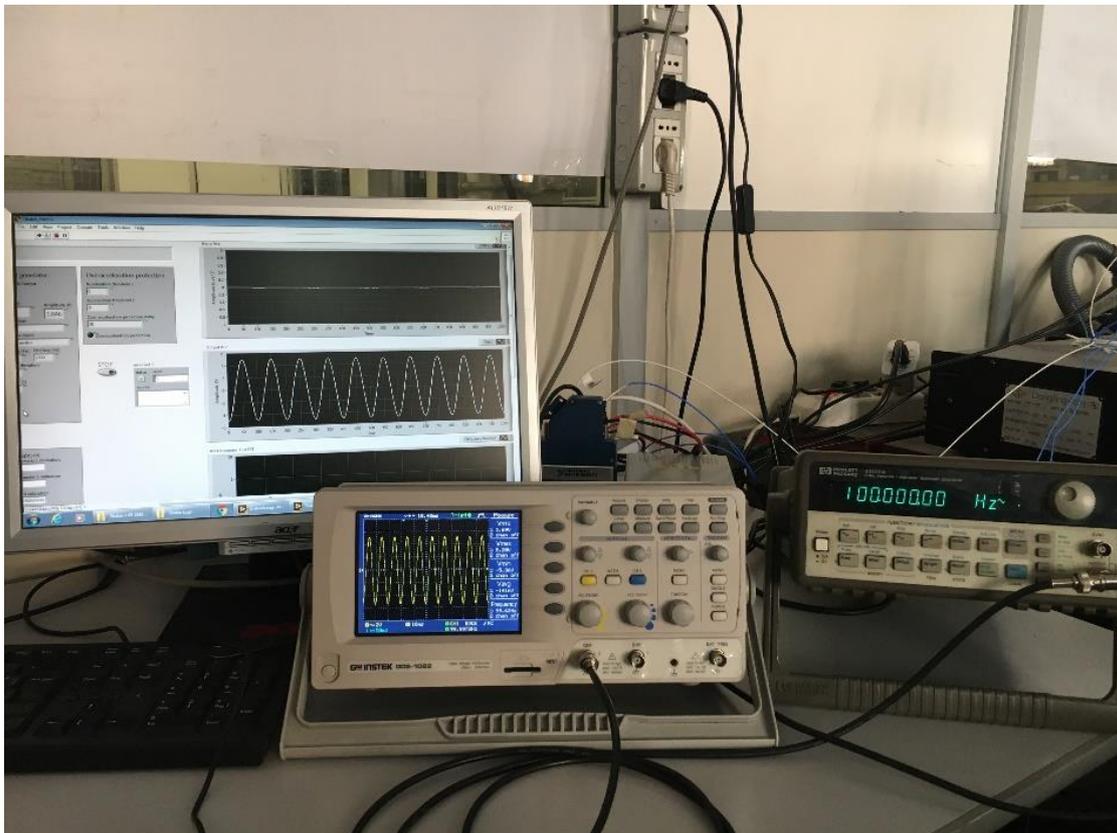


Figure 66: Accelerometer data acquisition validation

A series of frequency variable sinusoidal signal from 10Hz to 1kHz is provided to accelerometer channel, testing the visualizing acquired signal on the chart of the computer host end verify the relationship with the oscilloscope. Among all frequency set, the

control system acquisition performance at different levels remain the same. For example, a 100 Hz sinusoidal signal is generated by function waveform generator evaluating the system characteristic. The system acquires 10000 *samples/s* and the signal frequency is 100Hz, as result of this a observing time of 100ms is chose to make verification simpler. According to this relationship, ten sinusoidal waveforms is contained in that time range and this parameter is used to verify the good performance of the system. The previous picture shows the same signal representation on the all device screen, the oscilloscope and the chart implemented in the host computer have the same sinusoidal waveform number.

The control system software data stored functionality is tested using the same connection of the previous step. Indeed, the possibility of data loss is more common considering the data storing operation. It is usually the last part of software structure; in this case it is not real-time task and it is important to verify the correct order of the data saved in different extension. In the following figure, it is possible to notice how 100Hz signal is stored in terms of good signal reconstruction.

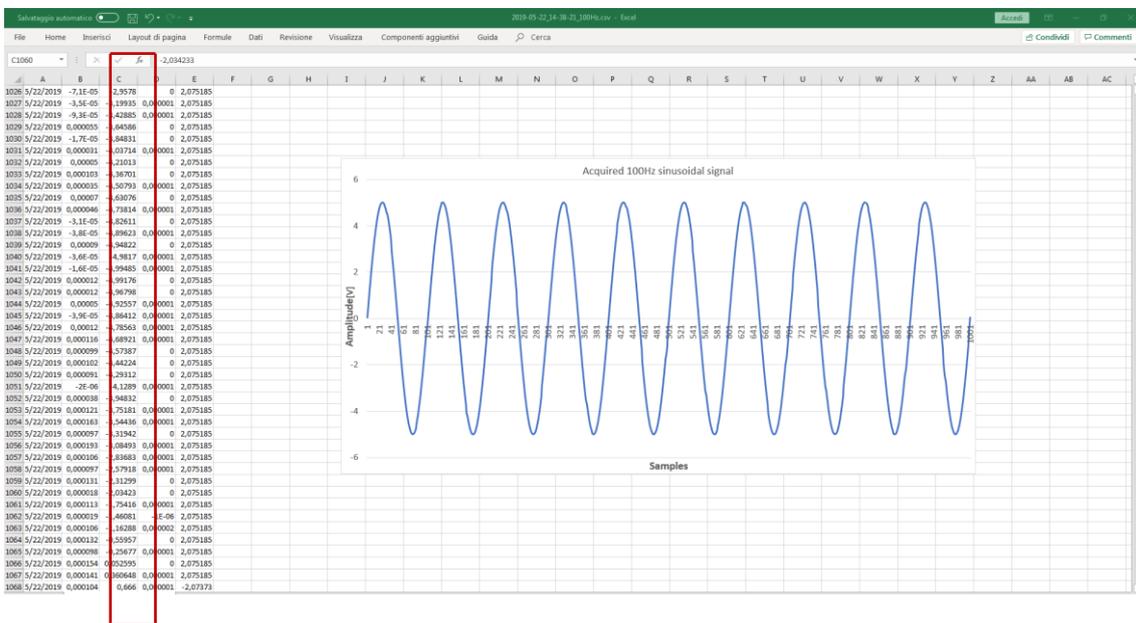


Figure 67: Acquired 100Hz sinusoidal samples

The highlighted red column is composed of all samples acquired by the accelerometer sensor channel, related to that the chart is elaborated and the signal is accurately reconstructed: 1000 samples is selected, and 10 sinusoidal waveforms is visualized. The same operations have been reproduced for the all frequency range related to experimental

test execution. The control system acquisition section has the same performance respect to the all frequency provided as input to it. As result of this, the control system developed for analysis of stress fatigue on the metallic material is considered sufficiently reliable to perform stress test procedure on the specimen.

4 Experimental results

The testing procedure, as previously anticipated, start with analyzing the characteristics of a new specimen. The operations correspond to this first step are setting the parameter related to signal sweep provided to specimen, the frequency range used is between 10Hz to 1000Hz and the duration of the test is about 60 sec . These parameters are sufficiently to acquire a good samples number respect to all used sensors, to perform all the analysis about the specimen characteristics starting from elaboration and evaluation of saved data. On the other hands, sweep duration test represents a stress on the specimen and its contribution on the fatigue of material could influence the ageing and correlated variable.

Among all data acquired, the main attention regards to analysis of modal frequency value of the specimen under test. Thanks to software structure and functionalities the modal frequency computation can be execute considering both extensimeter and accelerometer data. The Fast Fourier Transformation is performed on the acquired data to obtain information about the second modal frequency value. The availability of data stored in suitable Excel file, allowing to compute the frequency response of the specimen with any elaboration data software. In this case a MathWorks® MATLAB script is developed, implementing the possibility of select the desired file among stored by various test executed and then execute the Fast Fourier transformation. As result of this, it makes simpler and faster any comparative operation and evaluation of the specimen ageing state.

The script has the functionality of chose the sensors in which apply the Fast Fourier transformation, because according the specimen conditions is better to execute data elaboration on one sensor respect to the other. Indeed, observing experimentally the

behavior of the specimen when it is near to breakdown, the extensimeter position gives inconsistent data value because it is positioned on the break point. In this situation accelerometer placed on the specimen given a sufficiently accurate result of FFT computation to detect second modal frequency.

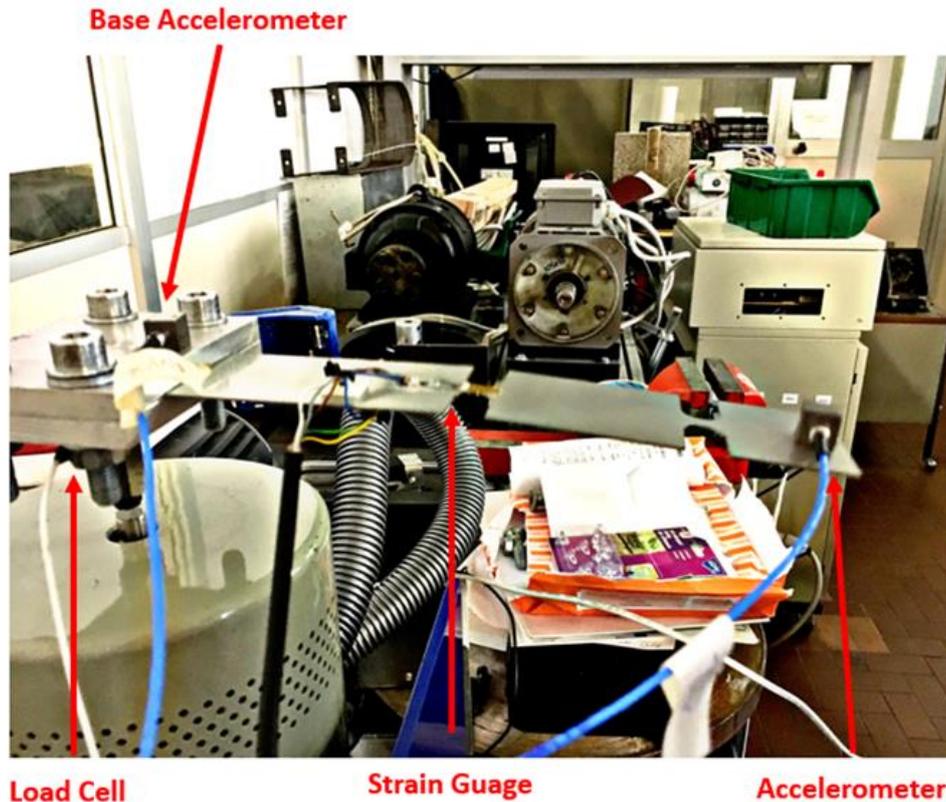


Figure 68: Sensors distribution on the specimen

As shows in the previous figure, the specimen presents high fatigue stress condition. Indeed, it is possible to notice visually the deformation in the middle of it respect to the fatigueless one. The strain gauge is positioned in the “Notch 1” point corresponding to fatigue break down point, when it is near to permanently damage the strain gauge gives not accurate value. Consequently, accelerometer placed on the specimen extremity provides data allowing to obtain information about material properties modification.

In the following pictures is illustrated the result of computation considering different level of specimen fatigue, executing FFT computation to data coming strain gauge or accelerometer.

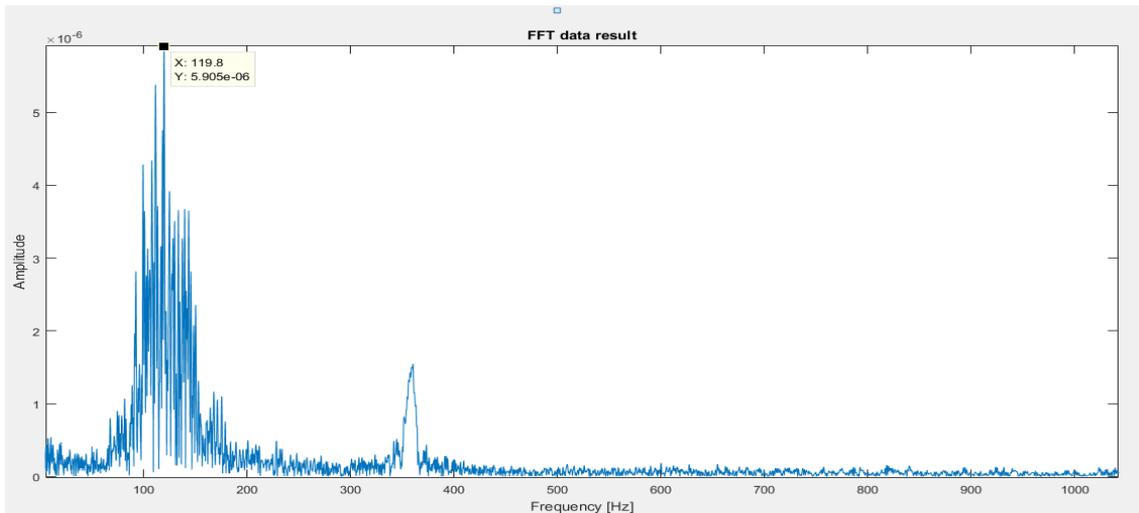


Figure 69: Sensors data FFT computation result 119Hz

The image shows the result of about 119Hz for second modal frequency of the specimen, this is easily identifiable on the chart by a higher amplitude value in a bounded frequency range respect to all sweep frequency interval set. The Fast Fourier computation result indicates medium fatigue accumulation on the metallic object under test.

The same specimen, after a stress cycle of limited duration, presents a variation of second modal frequency. In details, it has been performed a sweep test when the specimen is near to break down and the save data has been analyzed following the same procedure. The result of this in shows in the following picture:

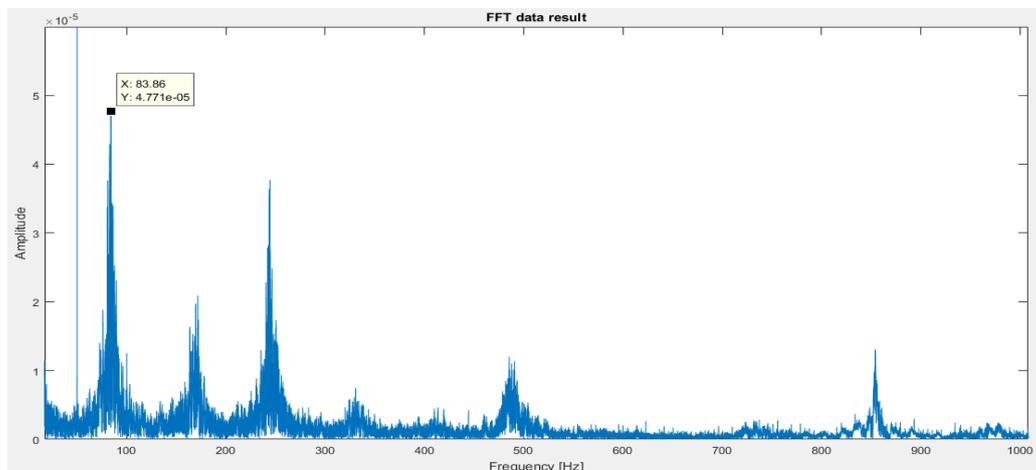


Figure 70: Specimen second modal frequency near to break down conditions

Considering the fatigue accumulation on the specimen metallic structure, the Fast Fourier transformation of the data coming from sensor gives a significantly variation of second modal frequency. Indeed, the result of frequency is approximately of 83 Hz as expected in considered fatigue condition.

The test operations, previously described, have been applied to various specimen characterized of a well-known metallic composition. Five specimens have been analyzed and in all of them the same behavior of second modal frequency has been observed. Indeed, the value of the considered modal frequency decreases with the increasing fatigue level of the specimen under test. Regarding the break down conditions, the experimentally result shows that it happens in an interval of time relatively similar among all specimen under test. The next figure gives graphical representation of second modal frequency trend of five specimens under test:

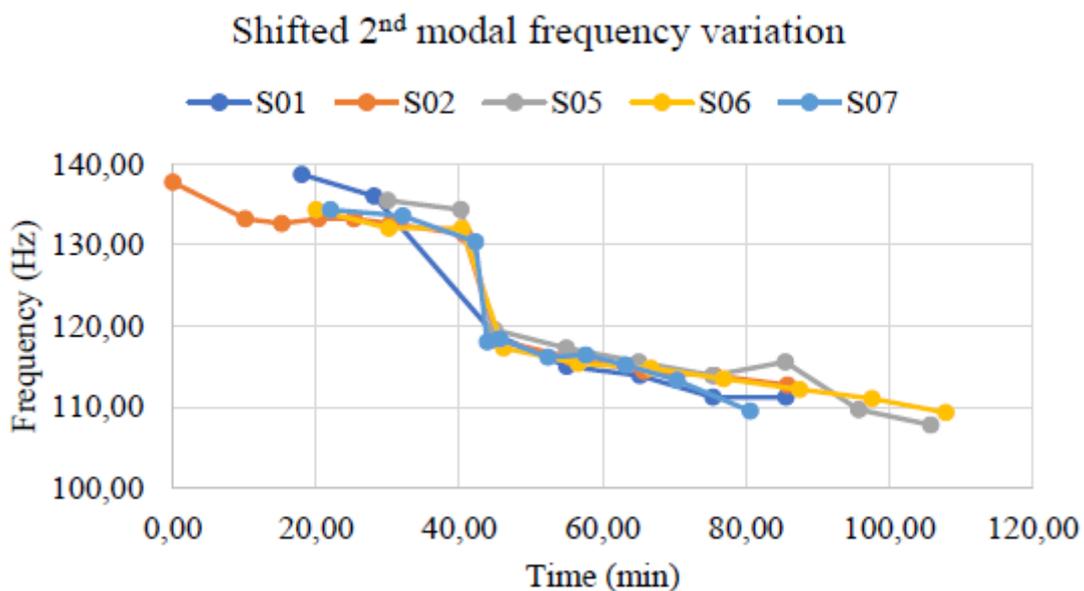


Figure 71: Second modal frequency trend of stressed specimens

The absolute final value of second modal frequency near to critical damage is not considered, because not relevant respect to statistical data analysis. Other considerations related to all data acquired are not objective of this thesis work and they will be analysed in others academics works.

5 Conclusions

In this thesis, an experimental set up to study a fatigue phenomenon has been performed. It is possible to evolve it adding more functionalities and more control operations on the actuators. The laboratory bench structure presents a reusable software for different modal electromechanical actuator guaranteeing a good integration with them. The real-time instruction, implemented with the usage of external NI Module, is characterized by possibility of easily manage deadline and signal generation type maintaining the same structure. Thanks to high integration level of LabVIEW with National Instrument module, it is possible to change software instructions to adapt it to additional or different module that can be used in the next control system developments. On the other hands, it is possible to add other sensors to improve data acquisition elements obtaining more information about the specimen characteristics using the defined manager task already implemented in the dedicated software section.

The main objective of the software development is to implement a laboratory bench using a distributed computational platform composed user interface, real-time computer and FPGA. The user can easily manage the parameter of signal generation like frequency, type and duration thanks to developed user interface running on any computer in which running LabVIEW software. The control system represents an evolution respect to previous software used to study and stress metallic specimen of different alloy. Indeed, the generation of filtered white noise signal gives to researches an appropriate instrument to simulate fatigue conditions and different ageing state of material under test. The researches or academic student can analyse all data coming from sensors at any moment, allowing to perform different data processing operations without the necessity to run the software. All these main control system characteristics can help the next academic study on structural mechanics to acquire, save and elaborate data about alloy material.

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