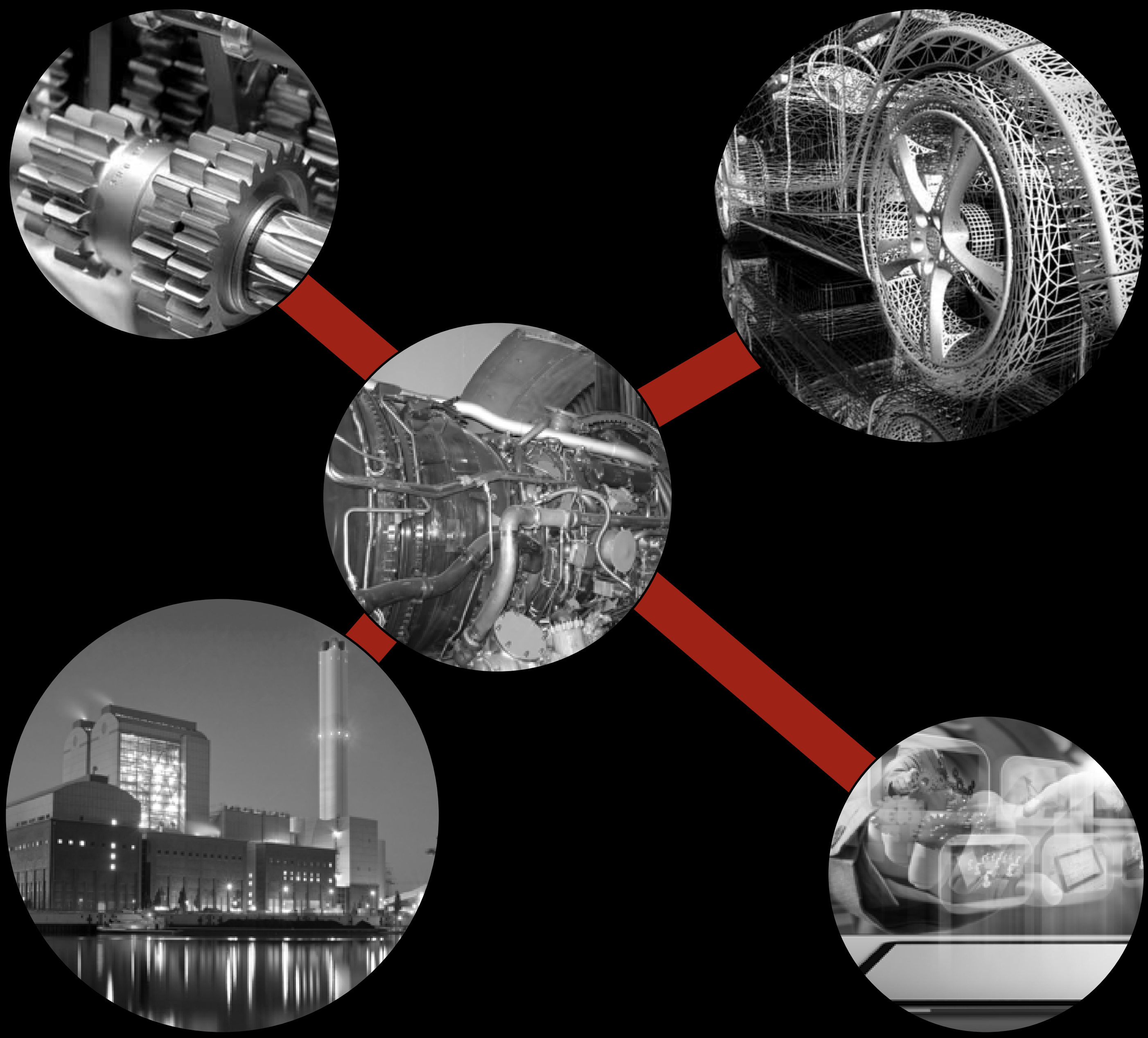
Sixth Edition

Noise and Vibration
Measurement Handbook



Measurement & Sensors

Key information on which sensors to use and how to use them

Algorithms & Analysis

The basic and advanced methods of noise & vibration analysis

Condition Monitoring

Learn why and how to monitor vibration in large rotating plant

About Prosig

Prosig was established in 1977 by members of the Institute of Sound & Vibration Research (ISVR) at the University of Southampton. The company's goal was, and remains, to create the best available tools for noise and vibration measurement.

The company focuses on producing reliable, high quality, integrated measurement systems that allow their customers to achieve best practice with the latest tools. Prosig sells hardware and software solutions and can offer consultancy, rental and training services around the world.

Our systems are designed using the knowledge that our experts have gained over 40 years of solving sound and vibration problems for the likes of NASA, Jaguar Land Rover, RWE, Ford, British Aerospace, Curtiss-Wright, BMW and Airbus as well as F1 teams, air forces, power generators, universities, researchers and defence organizations around the world.

Prosig is a part of Condition Monitoring Group (CMG) Limited. The combined knowledge and expertise of Prosig and the other CMG companies allows us to provide engineering solutions large and small across a range of disciplines and market sectors.

Prosig maintains a network of partners, distributors and agents to support business across the globe.

We are passionate about sound and vibration measurement and signal processing. We needed tools with a high degree of precision and integrity to measure and analyze sound and vibration. Nothing we found was good enough; so we made our own.

About CMG

Condition Monitoring Group (CMG) Ltd. specializes in measuring, analyzing, trending and interpreting vibration, acoustic and associated data for condition monitoring and testing across a broad range of industries.

The group has over 40 years' experience in data acquisition, engineering knowledge, system design and safety critical software development.

CMG is made up of:

- **Helitune Ltd.** – Helicopter Rotor Track & Balance, Component Vibration Analysis and Health & Usage Monitoring Systems (HUMS).
- **Prosig Ltd.** – Acoustic & Vibration Measurement and Testing, Vibration Condition Monitoring.
- **Beran Instruments Ltd.** – Vibration Condition Monitoring, Electronic Systems for Vibration Analysis of Rotating Machines.
- **Systems & Electronics, Inc. (SEI)** – Structural Health Monitoring Systems for Fixed-wing Aircraft.

With its headquarters in the UK, Condition Monitoring Group Ltd. has subsidiaries in three of its major markets – USA, Germany and Italy - and a global network of partners and distributors.

Chris Mason

General Manager, Prosig

"Prosig was created by engineers and academics who were committed to measuring and analyzing sound & vibration using the highest quality tools. We continue that tradition today by making measurement and monitoring systems that are precise, robust and reliable. Through our products, blogs, support services & training we continue to share our many decades of knowledge with our partners and clients.

Prosig Noise & Vibration Blog

The Noise & Vibration Measurement Blog contains many articles on all aspects of noise & vibration testing, measurement and analysis. To read some previous articles and to sign up for the newsletter visit

http://blog.prosig.com

Article Contents

Measurement & Sensors

Algorithms & Analysis

Condition Monitoring & Measurement

About The Authors...

Find out about the authors that you will find in this handbook

Don Davies

Don Davies graduated from the Institute of Sound and Vibration Research (ISVR) at Southampton University in 1979. Don specialises in the capture and analysis of vibration data from rotating machines such as power station turbine generators. He developed the original Prosig PROTOR product and is Applications Group Manager at Prosig. Don is a member of the British Computer Society.

Adrian Lincoln

Adrian Lincoln is Principal Analyst at Prosig Ltd and has responsibilities for signal processing applications, training and consultancy. He was formerly a Research Fellow at the Institute of Sound & Vibration Research (ISVR) at Southampton University. He is a Chartered Engineer and member of the British Computer Society and Institute of Mechanical Engineers.

Dr Mike Donegan

Mike graduated from the University of Southampton in 1979 and then went on to complete a PhD in Seismic Refraction Studies in 1982. Mike joined Prosig as a special applications engineer. He now researches & develops new algorithms for Prosig’s DATS software and assists customers with data analysis issues.

Chris Mason

Chris’ passion for engineering, technology & innovation began early with Sinclair ZX80’s, Commodore PETs & Apple][’s. This led to a career in product development, team leadership, web development and marketing. Chris has an MBA from the University of Winchester and is General Manager at Prosig. Away from the hustle, his passions include bicycles, technology, coffee, walking, reading, cooking and V8s.

James Wren

James Wren was Sales & Marketing Manager for Prosig Ltd until 2018. James graduated from Portsmouth University in 2001, with a Masters degree in Electronic Engineering. He is a Chartered Engineer and a registered Eur Ing. James left Prosig in December 2018 to pursure new opportunities.

Dr Colin Mercer

Dr Colin Mercer founded Prosig in 1977. He was formerly at the Institute of Sound and Vibration Research (ISVR), University of Southampton where he founded the Data Analysis Centre. Colin retired as Chief Signal Processing Analyst at Prosig in December 2016. He is a Chartered Engineer and a Fellow of the British Computer Society.

John Mathey

John Mathey graduated with a MS degree from the University of Toledo in 1972. John has over 35 years of experience with instrumentation, measurement, and analysis. Twenty-five of those years were spent at Ford Motor Company solving and providing training for vehicle noise, vibration, and harshness (NVH) issues. He was a technical specialist at Prosig USA, Inc. until his retirment in 2017.

Visit the Prosig Noise & Vibration Measurement blog at http://blog.prosig.com to read the latest posts and articles. There you can also sign up to our mailing list and make sure you never miss a new article.

SYSTEM PACKAGES

The system packages on the following pages are examples of the complete solutions that Prosig can provide. Alternative packages of hardware, software and sensors can easily be supplied to suit your individual needs. Please see the Hardware and Software sections of this catalog for details of our full range of products.

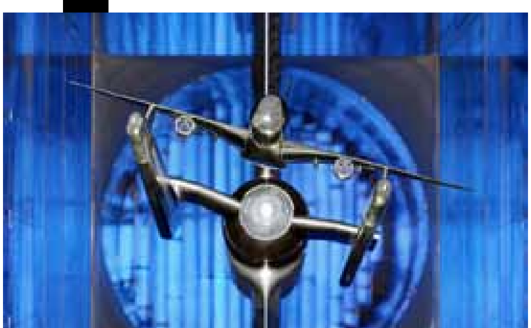
In the Real World

Brake Squeal Evaluation on High Performance Vehicle

The customer uses microphones, accelerometers, thermocouples & pressure transducers attached to a vehicle to measure brake squeal events. A **Prosig measurement system** measures, stores and analyzes all significant events during road tests lasteing several hours. Sophisticated pre- and post-trigger capture along with data visualisation in **DATS-toolbox** helps to achieve a better understanding of brake squeal. The Prosig system was selected after similar, competitive systems were unable to cope with the environmental and capture/analysis requirements.

Assessment of Human Exposure to Vibration

A **4-channel Prosig system** and the **DATS Human Biodynamics Analysis Suite** is used to make assessments on the exposure of the human body to vibration data. The data is captured on a moving train. Health and comfort criteria are calculated according to various ISO standards and provided to the user in the form of standard reports. The end-user has used many of the results in expert testimony work in legal cases.



Testing in Low Temperature Transonic Wind Tunnel

A high channel-count **DATS-hyper12 system** is used to capture the vibration signals from an aircraft model sited in a wind tunnel that can operate at temperatures as low as -261C and flow speeds as high as Mach 1.3. Strain and acceleration measurements at various locations on the body of the model are taken over a preset range of tunnel conditions. An additional **8-channel Prosig system** monitors the real-time forces and moments experienced by the balance gauge mounted inside the model.

Monitoring Flow in New Domestic Water Meter

Prosig supplied a turnkey industrial monitoring system that measures the accuracy of an innovative new design of domestic water meter. A **Prosig P8020** system is used to capture pre-conditioned fluid pressure signals together with other test rig control parameters such as temperatures and pressures. Advanced pulse analysis software in **DATS** processes the captured signals and produces detailed reports that compare meter performance at different flow rates under various test conditions.

Pre-build Assessment of Vibration in Tower Block

The customer needs to check whether noise and vibration from an underground train line is going to cause a nuisance in a proposed multi- storey housing block. A sophisticated measuring system based on a triaxial accelerometer is connected to a **Prosig P8000**, which is used to capture the data. The results of further analysis are used to determine if the noise and vibration of the trains will fall within prescribed limits.

DATS-easyRecord

Data Capture with Triggering
Realtime Frequency
Realtime Time Histories
Triggering
Data Export

The **DATS-easyRecord (DATS-ER)** package provides the quickest, easiest way to capture noise & vibration data. The system comprises a **DATS-solo** 4-channel, 24-bit data acquisition system and data capture software for your PC.

The **DATS-solo** is a pocket sized, USB powered, high precision, 24-bit data acquisition system.

The Solo's data capture software provides a simple, easy-to-use interface and displays real-time data while capturing so you can be sure that your measurements are of the highest quality.

If you need to quickly capture data to analyse later, either in **Prosig DATS** or **DATS-EM software**, the Easy Record package is what you need.

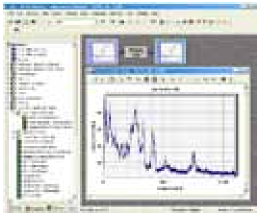
System includes...
- DATS-solo hardware
- DATS-solo capture software
- All leads & cables

DATS-easyRecord System

03-66-1109	DATS-ER (DATS-easyRecord) System including 4-channel DATS-solo and integrated DATS-solo data capture software.

DATS-easyMeasurement

Vibration Analysis
Acoustic Analysis
Data Acquisition
Time Domain Analysis
Frequency Analysis
Digital Filtering
Signal Arithmetic
Data Import / Export



DATS-easyMeasurement is a cost effective hardware and software solution for data capture and analysis. **DATS-easyMeasurement** comprises a **DATS-solo** 4-channel, 24-bit data acquisition system and **DATS-EM software**.

The **DATS-solo** is a pocket sized, USB powered, high precision, 24-bit data acquisition system.

The **DATS-EM software** is an intuitive, highly interactive & configurable package for the capture, analysis and reporting of noise & vibration and associated data.

System includes...
- DATS-solo hardware
- DATS-EM software
- All leads & cables

DATS-easyMeasurement System

03-33-0834	DATS-easyMeasurement System including 4-channel DATS-solo and integrated DATS-EM data acquisition & signal processing software.

DATS-HITS

- Powerful & Intuitive UI
- Small, Rugged Hardware
- Modal Testing
- FEA Validation
- FRF Studies
- Optional Sensors

Source: Kistler AG

The Prosig Hammer Impact Test System (HITS) is a complete hardware and software bundle that provides a test engineer with everything needed to capture and analyze frequency response data. The **DATS-solo** unit has four 24-bit analog inputs and capture speeds of up to 100k samples per second per channel are available in 24-bit precision.

The DATS **Hammer Impact Software** includes software tools to capture hammer impact data using the **DATS-solo** system.

Prosig's solution is not only the most simple to use and understand, but it is 100% reliable, giving perfect results every time. The complex mathematics of windowing, transfer function type, frequency range is all taken care of for you automatically.

Additionally, the automated peak picking algorithm will find the modes automatically for you. Need to export test results to Word or Excel for FEA validation? No problem. It's all included.

System includes...
- DATS-solo hardware
- DATS Hammer Impact software
- All leads & cables
- Optional sensors

DATS-HITS System

03-33-1073	DATS-HITS (Hammer Impact Test System) including 4-channel DATS-solo, DATS Hammer Impact Software (sensors can be purchased seperately)

DATS-listen

- Acoustic Studies
- Transportation Studies
- Traffic Noise
- Noise Level Measurement
- Hall Acoustics
- Studio Design

The DATS-listen System combines a high quality **DATS-solo** measurement system with the rich functionality of the **DATS Acoustic Analysis software** package.

The **DATS-solo** is an ultra-portable, high quality, 24-bit data acquisition system. It is compact, rugged and has 4 high speed analog inputs. Industry standard SMA or BNC sockets are used for input connections.

The **DATS Acoustic Analysis software** package has a range of time domain and frequency domain functions taken from the **DATS-toolbox package**. In addition it has functions specific to acoustic measurement such as a Sound Level Meter, 1/N Filters, a Room Acoustics suite, Reverberation Time T60, Total Absorption and so on. For Pyschoacoustic analysis, it adds Loudness, Sharpness, Roughness, Fluctuation Strength, Prominence Ratio and much more.

System includes...
- DATS-solo hardware
- DATS Acoustic Analysis software package
- All leads & cables
- Optional sensors

DATS-listen System

03-66-1134	DATS-listen System including 4-channel DATS-solo, DATS Acoustic Analysis software package, all necessary cables and leads.

DATS-easyRefine

NVH Studies
- Engine
- Transmission
- Pump Noise
- Muffler / Exhaust
- Sub-system & Component Testing

Sound Quality Studies
- Vehicle Cabin Noise
- Loudness, AI, Harshness etc.

Noise Source Identification

Sound Power Measurement

The **DATS-easyRefine System** is a complete hardware and software bundle that provides an NVH test engineer with everything needed to capture and analyze NVH data. The DATS-solo unit has four 24-bit analog inputs. Capture speeds of up to 100k samples per second per channel are available.

The **DATS NVH analysis software** includes data acquisition software to control the **DATS-solo system** and a full analysis & reporting package.

Analysis functions are provided for waterfall analysis, order extraction, sound quality metrics, frequency domain processing (FFT, power spectra, etc), digital filtering and much more. Complex multi-channel analysis applications can be easily created using Prosig's unique Visual Scripting environment.

System includes...
- DATS-solo hardware
- DATS-toolbox software
- DATS NVH Analysis Suite
- All leads & cables
- Optional sensors

DATS-easyRefine	
03-33-1008	DATS-easyRefine System including 4-channel DATS-solo, DATS-toolbox software, DATS NVH Analysis suite, all necessary cables & leads.

DATS-proRefine

Noise & Vibration/Refinement Studies

Sound Quality Studies

Animation / ODS

Psychoacoustic Metrics

Noise Source Identification

Sound Power Measurement

Chassis Dynamics

The **DATS-proRefine System** provides everything that the DATS-easyRefine System contains, but adds extra input channels, CAN bus capability and more specialist application software. The **DATS-tetrad** unit is configured with sixteen 24-bit analog inputs, dual CAN bus inputs and two dedicated tacho inputs.

As well as the **DATS NVH** analysis software the **proRefine** system adds the **Hammer Impact**, **Psychoacoustics** and **Structural Animation** packages.

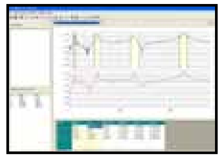
System includes...
- DATS-tetrad 16-channel data capture system
- CAN bus input
- GPS Tracking
- DATS-toolbox software
- DATS NVH analysis suite
- DATS Hammer Impact software
- DATS Psychoacoustic analysis suite
- DATS Structural Animation software
- All leads & cables
- Optional sensors

DATS-proRefine	
03-33-1009	DATS-proRefine System including DATS-tetrad with 16 analog channels & CAN bus input, DATS-toolbox software, DATS NVH Analysis suite, DATS Hammer Impact software, DATS Structural Animation software, DATS Psychoacoustic software, all necessary cables & leads.

DATS-easyModal

Hammer Impact Testing

- Frequency response functions
- Coherence measurement
- Force and response windowing



3D Structural Animation

- Model editor
- Operating deflection shape
- Wireframe or solid animation
- Time or frequency animation

Modal Analysis

- Curve-fitting to frequency response functions
- Identification of modal frequencies and damping factors
- Identification of mode shapes for animation

The **DATS-easyModal System** is a complete hardware and software bundle that provides a test engineer with everything needed to capture and analyze frequency response data. The **DATS-solo** unit has four 24-bit, analog inputs and capture speeds of up to 100k samples per second per channel.

The DATS **Hammer Impact Software** includes software tools to capture hammer impact data using the **DATS-solo** system. Identification of modal parameters from FRF's is provided within the **Modal Analysis Software**. The **Structural Animation** package provides all of the facilities to build and animate models of the test piece.

System includes...
- DATS-solo hardware
- DATS-toolbox software
- DATS Modal Analysis software suite
- DATS Hammer Impact software
- DATS Structural Animation software
- All leads & cables
- Optional sensors

DATS-easyHumanVib

ISO2631 Whole Body (Parts 1,4 & 5)

Motion Sickness ISO2631

ISO5349 Hand Arm (including Multi-Tool)

DIN45669 Building Vibration

ISO6954 Ship Vibration

ISO8041 Weightings

SEAT Vibration (ISO10326-1 & EEC78/764)

VDV, RMQ, RMS, MSDV, MTVV

Vibration Quality Measure

Many aspects of our lives including work, travel and leisure expose our bodies to vibration. Many of these vibration phenomena are described and limited by legislation, and many can be accurately measured according to specific ISO, DIN and EEC standards.

The **DATS-easyHumanVib System** provides all the hardware and software required to carry out human vibration studies. The **DATS-solo** unit has four 24-bit analog inputs for accurate measurement of vibration signals from the supplied transducers.

DATS Human Biodynamics Suite contains all of the necessary functions to analyze vibration data and produce results for the standards shown above.

System includes...
- DATS-solo hardware
- DATS-toolbox software
- DATS Human Response Biodynamics Suite
- All leads & cables
- Optional sensors

DATS-easyModal	
03-33-1010	DATS-easyModal System including 4-channel DATS-solo, DATS-toolbox software, DATS Modal Analysis software, DATS Hammer Impact software, DATS Structural Animation software, all necessary cables & leads.

DATS-easyHumanVib	
03-33-1007	DATS-easyHumanVib System including 4-channel DATS-solo, DATS-toolbox software, DATS Human Response Biodynamics Suite, all necessary cables & leads.

HARDWARE PRODUCTS

DATS-solo

4-Channel, Pocket Sized, USB Powered, 24-bit Data Acquisition System

- Small, light, ultra portable
- 24-bit precision
- Sample at up to 100k samples/second/channel
- 4 analog channels
- 105dB dynamic range
- -130dB noise floor
- USB powered

The DATS-solo is a pocket-sized, ultra portable, high quality, 24-bit data acquisition system. It is compact, rugged and has 4 inputs selectable as IEPE, voltage or tachometer inputs. Industry standard SMA connectors are used for input with optional adapters for other connectors. The size, tough packaging and power from USB make it ideal for mobile use. Internally there is comprehensive signal conditioning for IEPE or voltage with anti-alias filters, all controlled by Prosig's DATS software.

System

Inputs	4 x Analog (IEPE or voltage)
Maximum sampling rate	100k samples/sec per channel
Resolution	24 bit
Communications	USB
Noise floor	-130dB
Overall accuracy	± 0.10% FSD
Input voltage range	9 selectable ranges
Input impedance	1MΩ
IEPE power	20V@4mA switchable

Analog Inputs

Input mode - Analog	IEPE (single ended) Voltage (differential)
Anti-alias protection	>150dB
Dynamic range	100dB @ 10k samples/second
Maximum Input Range	± 24V

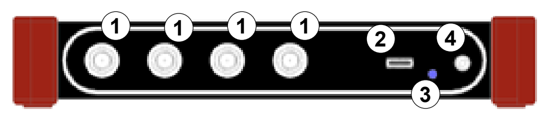
Tachometer Input (any input channel)

Input mode - Tachometer	IEPE, Voltage
Tacho range	Up to 1,500,000rpm @ 1ppr
Tacho input range	± 24V

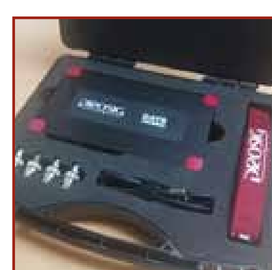
Environment & General

Shock and vibration	MIL-STD-810G
Operating temperature	0°C to +65°C (32°F to +149°F)
Environmental protection	IP54
Humidity	95% RH, non-condensing
Power usage	<2W
Supply voltage	from USB
Connectors	4 x SMA (Analog inputs) USB-C (Power & data)
Dimensions (H x W x D)	30mm x 167mm x 97mm 1.2" x 6.6" x 3.8"
Weight	365g (0.80 lbs) 823g (1.81 lbs) with case and accessories

Connectors



1. Analogue Inputs
2. USB-C (Data & power)
3. Power LED
4. Electrical Ground



DATS-tetrad

Multi-channel, standalone, mobile, 24-bit Data Acquisition System

- 24-bit precision
- Ethernet/USB/WiFi
- Internal battery
- Standalone operation
- Rugged & portable
- Easy-to-use software

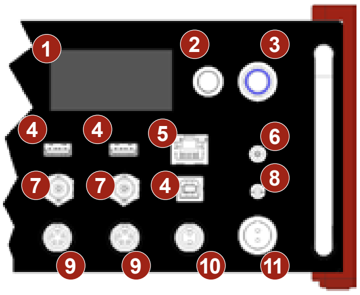
The DATS-tetrad is a rugged, portable data acquisition system that supports up to 32 high speed analog inputs and can be stacked for higher channel counts. Many sensor input options are available including microphones, accelerometers, strain gauges, thermocouples, load/pressure/force sensors, tachometers, rotary encoders, digital I/O and CAN-bus. Analog output and digital output are available with options for open and closed loop control. The system has an internal GPS receiver.

The unit has hassle free, zero configuration Ethernet, WiFi & USB communication interfaces. It can be powered from mains, vehicle or its internal battery power. A built-in processor and solid state storage provide standalone capability. The system can also be connected to a laptop or PC for real time monitoring and data capture via any of the communication interfaces.

System

Inputs	Up to 32 channels (plus tachos)
Maximum sampling rate	Up to 400k samples/sec/channel
Resolution	24 bit
Communications	Ethernet (Gigabit) USB WiFi
Multi Rate Sampling	Multiple sampling rates can run concurrently on separate cards
GPS (Optional)	Built-in with external aerial socket
I/O Options	See p. 13-15
Battery (Optional)	up to 4 hours (dependent on configuration)

Front Panel



1. Display screen
2. Change display page
3. Power on/off
4. USB connection
5. Ethernet connection
6. GPS antenna connection
7. Tachometer inputs
8. Ground connection
9. Multi-system interconnect
10. Fused output power
11. Input power

Environment & General

Operating temperature	-10°C to 45°C (14°F to 113°F) (Cold start) -20°C to 45°C (-4°F to 113°F) (Warm start)
Humidity	95% RH, non-condensing
Power usage	<80W (worst case)
Shock and vibration	MIL-STD-810G
IP Rating	IP31
Supply voltage	Internal battery (≤100Wh meeting IATA & PHMSA guidelines) External 10-36V DC (e.g. vehicle battery) AC mains (adapter supplied)
Expansion	Interconnectable by Ethernet & USB
Dimensions (H x W x D)	115mm x 360mm x 225mm 4.5" x 14.2" x 8.9"
Weight	~5kg/11lbs (dependent on configuration)

Measurement Sensors

The DATS-tetrad can measure the following:

Vibration	Acoustics
Displacement	Strain
Tachometers	Force
Load	Shock
Torque	Position
Temperature	Pressure
CAN-Bus	

Or any suitable IEPE, charge, temperature, bridge, or powered sensor.

DATS-hyper12

Rack Mount/Portable. High Channel Count System

- High channel count
- Standalone or rack mount
- 24-bit precision
- Up to 100k samples/sec/channel (24-bit)
- Up to 400k samples/sec/channel (16-bit)
- Up to 1024 channels
- Configurable channel options

The DATS-hyper12 is the high channel count version of the DATS-tetrad 24-bit data acquisition system. It is ideal for rack mounting in a laboratory, but can also be used as a portable, standalone system. It has all the same signal conditioning as the DATS-tetrad. It can also be configured with all the same measurement modules.

System	
Analog inputs	**Up to 1024 channels plus tachos**
Expansion	**Flexible packaging options**
Split rate sampling	**Multiple sampling rates can run concurrently in separate cards**
Programmability	**All features under software control**
Communications	**USB**
Environmental	
Shock and vibration	**Suitable for mobile use (5g rms)**
Operating temperature	**0°C to +40°C 32°F to +104°F**
Humidity	**80% RH, non-condensing**
Weight	**Dependent on configuration, channel count & chassis**
General	
Supply voltage	**Choice of 10-36V DC (e.g. vehicle battery) or AC mains (adapter supplied)**
Dimensions† (H x W x D)	**185mm x 450mm x 400mm (7.3″ x 17.7″ x 15.7″)**

† Dimensions are measured exclusive of any handles or other attachments

Remote Control Unit

03-33-0892

03-33-0893

- Simple LED display for ‘heads up’ operation
- Large tactile buttons allows ‘eyes on the road’
- USB connection
- Button box available separately
- Optional LED display

The button box / LED indicator unit is designed to aid tests where it is necessary to drive a vehicle and control data capture. The LED indicator box can be attached in the vehicle to provide a ‘heads up’ display. The button box has large, easy to select buttons that do not require an operator to break eye contact with vehicle controls or the road / track.

Note that DATS-tetrad systems offer alternative wireless solutions for remote control of the unit.

DATS-tetrad/hyper12 wired remote control

03-33-0892	Remote control button box
03-33-0893	Remote control LED indicator box

sales@prosig.com +1 847 228 0985 (USA) +44 (0)1329 239925 (UK) http://prosig.com

DATS-tetrad/hyper12 Measurement Modules

All of the cards in this section are available for the DATS-tetrad and DATS-hyper12 systems. The DATS-tetrad can be configured with a maximum of four (4) cards. The DATS-hyper12 can hold up to twelve (12) cards.

8X02

4ch ADC + Tacho, IEPE, Direct

- 4 analog channels and 1 tacho input
- DC, AC and IEPE† inputs
- 400k samples/second/channel
- Tacho input sampled at up to 800k samples/second/channel
- TEDS with connection detection

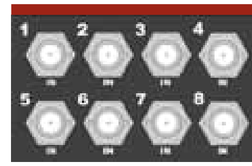
The 8X02 is a flexible general purpose acquisition card, with built-in signal conditioning for almost any type of transducer. It has the capability of high sample rates and synchronous parallel sampling with an additional dedicated tachometer input. It also offers a choice of AC or DC coupling to direct voltage inputs and support for IEPE† transducers, including those with TEDS. Importantly, it has a large number of analogue amplifier steps to maximize resolution. This card offers the flexibility of capturing data in 24-bit or 16-bit resolution.

8X04

4ch ADC + Tacho, IEPE, Direct, Bridge

- 4 analog channels and 1 tacho input
- DC, AC and IEPE† inputs
- 400k samples/second/channel
- Tacho input sampled at up to 800k samples/second/channel
- TEDS with connection detection
- Programmable excitation
- Programmable ¼, ½, full bridge input
- Input nulling & excitation sensing

The 8X04 is an ultra-flexible general purpose acquisition card. It encapsulates Prosig's 30 years of test and measurement experience and is the only card you'll ever need! The 8X04 has all the functionally and full specification of the 8X02 card. But additionally each channel includes bridge completion configurations of ¼, ½ and full bridge, internal calibration shunt resistors and selectable bridge resistance configurations of 120, 350 or 1000Ω. Further, each channel provides program selectable supply voltage of 5V & 10V for transducer excitation.



8X12

8ch ADC + Tacho, IEPE, Direct, TEDS

- 8 analog channels and 1 tacho input
- DC, AC and IEPE† inputs
- 100k samples/second/channel (24 bits)
- Tacho input sampled at up to 800k samples/second/channel
- TEDS with connection detection

This card is ideal for situations where higher sampling rates are not required, but high quality, repeatable, high resolution data captures are desired. Although the 8X12 has a slightly lower specification than the 8X02 it provides twice the channel density. This allows for example a DATS-tetrad chassis to support a total of analog channels with two tacho channels. This card is used primarily in situations where high channel counts are required, the flexible, mutlipole connector makes the complex wiring tasks associated with high channel counts systems both manageable and tidy.

8X02

Description	**4ch ADC + Tacho, IEPE, Direct, TEDS**
Input channels	**4**
Output channels	**n/a**
16-bit sample rate *	**400k**
24-bit sample rate *	**100k**
Effective bandwidth	**0.4 x sample rate**
Anti-aliasing attenuation	**> 100dB**
AC coupling high pass filter	**20dB/dec -3dB at 0.3 or 1Hz**
DC Input	✓
AC Input	✓
IEPE Input	✓
Charge Input	×
Programmable excitation	×
24-bit Dynamic range	**105dB at 10Ks/s**
24-bit Noise floor	**-120dB at 10Ks/s**
16-bit Dynamic range	**92dB at 10Ks/s**
16-bit Noise floor	**-110dB at 10Ks/s**
Non-linearity	**< 1 bit**
Accuracy	**±0.1% FSD**
DC Offset control	**±50% FS in 32768 steps**
Tacho channels	**1**
Tacho input range	**±28V**
Supports TEDS	✓
Autozero	✓
Input range	**±10mV to ±10V FSD**
Output range	**n/a**
Gain Steps	**13**
Input common mode range	**±10V**
Absolute max input range	**±24V**
Prog. bridge completion	×
Connector	**BNC**
Power usage (worst case)	**6W**

8X04

Description	**4ch ADC + Tacho, IEPE, Direct, Bridge, TEDS**
Input channels	**4**
Output channels	**n/a**
16-bit sample rate *	**400k**
24-bit sample rate *	**100k**
Effective bandwidth	**0.4 x sample rate**
Anti-aliasing attenuation	**> 100dB**
AC coupling high pass filter	**20dB/dec -3dB at 0.3 or 1Hz**
DC Input	✓
AC Input	✓
IEPE Input	✓
Charge Input	×
Programmable excitation	✓
24-bit Dynamic range	**105dB at 10Ks/s**
24-bit Noise floor	**-120dB at 10Ks/s**
16-bit Dynamic range	**92dB at 10Ks/s**
16-bit Noise floor	**-110dB at 10Ks/s**
Non-linearity	**< 1 bit**
Accuracy	**±0.1% FSD**
DC Offset control	**±50% FS in 32768 steps**
Tacho channels	**1**
Tacho input range	**±28V**
Supports TEDS	✓
Autozero	✓
Input range	**±10mV to ±10V FSD**
Output range	**n/a**
Gain Steps	**13**
Input common mode range	**±10V**
Absolute max input range	**±24V**
Prog. bridge completion	✓
Connector	**Lemo**
Power usage (worst case)	**8W**

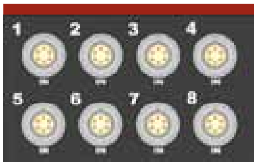
8X12

Description	**8ch ADC + Tacho, IEPE, Direct, TEDS**
Input channels	**8**
Output channels	**n/a**
16-bit sample rate *	**n/a**
24-bit sample rate *	**100k**
Effective bandwidth	**0.4 x sample rate**
Anti-aliasing attenuation	**> 100dB**
AC coupling high pass filter	**20dB/dec -3dB at 0.3 or 1Hz**
DC Input	✓
AC Input	✓
IEPE Input	✓
Charge Input	×
Programmable excitation	×
24-bit Dynamic range	**102dB at 10Ks/s**
24-bit Noise floor	**-120dB at 10Ks/s**
16-bit Dynamic range	**n/a**
16-bit Noise floor	**n/a**
Non-linearity	**< 1 bit**
Accuracy	**±0.1% FSD**
DC Offset control	**±FSD in 32768 steps**
Tacho channels	**1**
Tacho input range	**±28V**
Supports TEDS	✓
Autozero	✓
Input range	**±10mV to ±10V FSD**
Output range	**n/a**
Gain Steps	**4**
Input common mode range	**±10V**
Absolute max input range	**±24V**
Prog. bridge completion	×
Connector	**BNC**
Power usage (worst case)	**6W**

† IEPE (Integral Electronic PiezoElectric) type transducers are often known by trade names such as Piezotron®, Isotron®, DeltaTron®, LIVM™, ICP®, CCLD, ACOtron™ and others.
* All sample rates are specified in number of samples per second per channel

DATS-tetrad/hyper12 Measurement Modules

All of the cards in this section are available for the DATS-tetrad and DATS-hyper12 systems. The DATS-tetrad can be configured with a maximum of four (4) cards. The DATS-hyper12 can hold up to twelve (12) cards.

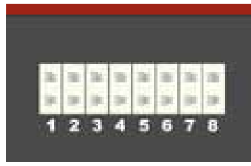


8X14

8ch ADC + Tacho, Direct, Bridge, IEPE, TEDS

- 8 analog channels and 1 tacho input
- IEPE, DC, AC inputs
- 100k samples/second/channel (24 bits)
- Tacho input sampled at up to 800k samples/second/channel
- Programmable excitation
- Programmable ¼, ½, full bridge input
- Input nulling & excitation sensing
- TEDS with connection detection

This card has the main features of the 8X12 and includes bridge completion and transducer excitation. Each channel provides bridge completion configurations of ¼, ½ and full bridge, internal calibration shunt resistors and selectable bridge resistance of 120, 350 or 1000Ω. This card allows a DATS-tetrad chassis to support up to 32 analog channels and two tacho channels. The flexible mutlipole connector gives these systems manageable wiring and offers the option of fast connection external boxes if desired. Each channel also provides program selectable supply voltage of 5V & 10V for transducer excitation.



8X08

8ch Thermocouple

- Eight channels of thermocouple inputs
- Universal input connector supports all popular thermocouple types
- Smallest step change 0.075 degrees (assuming 1 degree = 40µV)
- Integral cold junction reference
- Typical accuracy : 0.5°C

This is the universal thermocouple card suitable for use with industry standard connector types. The 8408 provides up to eight thermocouple inputs and supports all popular thermocouple types. This card gives the option for temperature data to be integrated and synchronised with noise and vibration data.

8X20

4ch Advanced Tacho

- Programmable signal conditioning to de-bounce inputs
- 240MHz resolution
- Pulse counting
- Noise Offset' & 'Hold Off' setting
- Programmable threshold & slope
- Pulse time stamping

The 8X20 card is intended as a solution for situations with rotating machines where positional information and time relative to position information are required. This would classically be a very high speed shaft encoder with a fine resolution. This card is used in applications where there is a requirement to accurately measure rotational speed at several points in a drivetrain. The high speed and resolution of this card mean it is suitable for in-depth rotational machine analysis such as torsional and angular vibration. The 8420 card measures the time between pulses with a 4ns resolution.

8X14

Description	8ch ADC + Tacho, Direct, Bridge
Input channels	8
Output channels	n/a
16-bit sample rate *	n/a
24-bit sample rate *	100k
Effective bandwidth	0.4 x sample rate
Anti-aliasing attenuation	> 100dB
AC coupling high pass filter	20dB/dec -3dB at 0.3 or 1Hz
DC Input	✓
AC Input	✓
IEPE Input	✓
Charge Input	×
Programmable excitation	✓
24-bit Dynamic range	102dB at 10Ks/s
24-bit Noise floor	-120dB at 10Ks/s
16-bit Dynamic range	n/a
16-bit Noise floor	n/a
Non-linearity	< 1 bit
Accuracy	±0.1% FSD
DC Offset control	±FSD in 32768 steps
Tacho channels	1
Tacho input range	±28V
Supports TEDS	✓
Autozero	✓
Input range	±10mV to ±10V
Output range	n/a
Gain steps	4
Input common mode range	±10V
Absolute max input range	±24V
Prog. bridge completion	✓
Connector	Lemo
Power usage (worse case)	12W

8X08

Description	8ch Thermocouple
Input channels	8
Output channels	n/a
16-bit sample rate *	n/a
24-bit sample rate *	500
Effective bandwidth	n/a
Anti-aliasing attenuation	n/a
DC Input	✓
AC Input	×
IEPE Input	×
Charge Input	×
Programmable excitation	×
Non-linearity	< 1 bit
Accuracy	±0.1% FSD
Tacho channels	n/a
Tacho input range	n/a
Supports TEDS	×
Autozero	×
Input range	Thermocouple
Output range	n/a
Input common mode range	n/a
Absolute max input range	n/a
Prog. bridge completion	×
Connector	IsoThermal Block
Power usage (worst case)	6.2W

8X20

Description	Advanced Tacho
Tacho input channels	4
Tacho input range	±28V
Absolute max input range	±50V
Slope selection	+ve, -ve
Dynamic noise rejection	✓
Resolution	4ns
Connector	BNC
Power usage (worst case)	1.3W

* All sample rates are specified in number of samples per second per channel

DATS-tetrad/hyper12 Measurement Modules

All of the cards in this section are available for the DATS-tetrad and DATS-hyper12 systems. The DATS-tetrad can be configured with a maximum of four (4) cards. The DATS-hyper12 can hold up to twelve (12) cards.

8X24

2ch/4ch DAC, Digital IO

- Four analog DAC output channels
- 288k samples/second/channel maximum output
- Digital interpolating filter
- Optional integral digital IO with 8 inputs & 8 outputs

The 8X24 DAC card, often known as an analog output card, is ideal for situations where analog replay of signals is required. Traditionally, it is used in applications such as modal analysis or general noise and vibration analysis. Analogue output is most often used where single or multi-point shaker excitation is required. Captured or various generated signals can be replayed as analog voltages at optimal sample rates.

A selection of optional front panel configuration offers either four DAC outputs or digital IO only. These options offer greater flexibility and integration with other systems.

8X40

CAN, GPS

- CAN bus input
- Passive and active CAN modes
- Time stamping: time or sample number
- GPS Data

The 8X40 card supports both simple monitoring, where messages are read and logged from the bus, and PID mode, where automatic PIDs can be requested under user control. CAN bus gives the flexibility of access to the tens or hundreds of parameters that are already present on an automotive vehicle or even modern aircraft communication bus.

The 8440 card supports two separate, independent CAN bus inputs for dual system monitoring and capture.

For DATS-hyper12 systems a GPS option is available on this card so that position, velocity or accurate wall clock time can be recorded with the data. Further there are GPS options that have different specification depending on the customer's requirements.

For DATS-tetrad systems there is an optional internal GPS module.

8X24

Description	2ch/4ch DAC, Digital IO
Option 1 - 4ch DAC	
Analogue output channels	4
Digital input channels	0
Digital output channels	0
24-bit sample rate *	288k
Analog output range	±4V
Digital output range	n/a
Connector	4 x BNC
Power usage (worst case)	1.8W
Option 2 - Digital IO only	
Analogue output channels	0
Digital input channels	8 TTL
Digital output channels	8
24-bit sample rate *	n/a
Digital output range	TTL compatible
Connector	2 x 9-way D-type
Power usage (worst case)	1.8W

8X40

Description	CAN
Link interface	ISO11898
Bus rates	125kbits/sec 250kbits/sec 500kbits/sec 1Mbits/sec
Operating modes	Broadcast, PID, DMR
Power usage (worst case)	1.3W
CAN Bus inputs	2
GPS Option 1	
Receiver type	50 channels, GPS L1
Update rate	10Hz
Velocity accuracy	0.1 m/sec
Position accuracy	2.5m
Time accuracy	30ns RMS
GPS Option 2	
Receiver type	GPS L1
Update rate	20Hz
Velocity accuracy	0.03m/s
Position accuracy	1.8m
Time accuracy	20ns RMS

* All sample rates are specified in number of samples per second per channel

DATS-solo

03-33-1109 DATS-solo 4-channel system. Includes
- DATS-solo with 4 analog input channels with SMA connectors
- DATS-solo Data Capture Software for Microsoft Windows
- PC to DATS-solo USB communications cable (02-66-1135)
- 4 x SMA-to-BNC adaptors
- Ground cable
- Carry case

DATS-tetrad

03-33-1112 DATS-tetrad 4 card chassis. Includes
- Chassis (capable of holding a maximum of 4 cards. Chassis can be linked for higher channel counts)
- PC to DATS-tetrad USB communications cable (02-33-0852)
- PC to DATS-tetrad Ethernet communications cable (02-34-1136)
- Mains power supply/charger for DATS-tetrad (02-34-1137)
- In-vehicle power cable for DATS-tetrad (02-34-1138)
- GPS aerial (optional)
- Ground cable

Select any combination of the following cards up to a maximum of 4 cards

03-33-8602	4ch ADC + Tacho, IEPE, Direct (BNC connectors) *
03-33-8604	4ch ADC + Tacho, IEPE, Direct, Bridge (6-pin Lemo connectors) *
03-33-8608	8ch Thermocouple
03-33-8612	8ch ADC + Tacho, IEPE, Direct
03-33-8614	8ch ADC + Tacho, Direct, Bridge
03-33-8620	4ch Advanced Tacho
03-33-8624	4ch DAC, Digital IO
03-33-8640	CAN bus

* The DATS-tetrad chassis has two tacho inputs (T1 & T2). To have a tacho input available on T1 either an 8602, 8604, 8612 or 8614 card needs to be fitted in slot 1. Similarly, to have a tacho input available on T2 either an 8602, 8604, 8612 or 8614 card needs to be fitted in slot 2.

DATS-hyper12

03-33-8048 96-channel (12 card) chassis. Includes
- Chassis (capable of holding a maximum of 12 cards. Chassis can be linked for higher channel counts)
- Chassis is available in rack mountable or portable form
- PC to DATS-hyper12 USB 2.0 communications cable (02-33-852)
- Mains power supply for DATS-hyper12 (02-33-867)
- In-vehicle power cable for DATS-hyper12 (02-33-866)

Select any combination of the following cards up to a maximum of 12 cards

03-33-8502	4ch ADC + Tacho, IEPE, Direct (BNC connectors) *
03-33-8504	4ch ADC + Tacho, IEPE, Direct, Bridge (6-pin Lemo connectors) *
03-33-8508	8ch Thermocouple
03-33-8512	8ch ADC + Tacho, IEPE, Direct
03-33-8514	8ch ADC + Tacho, Direct, Bridge
03-33-8520	4ch Advanced Tacho
03-33-8524	4ch DAC, Digital IO
03-33-8540	CAN, GPS

* The DATS-hyper12 chassis has two tacho inputs (T1 & T2). To have a tacho input available on T1 either an 8502, 8504, 8512 or 8514 card needs to be fitted in slot 1. Similarly, to have a tacho input available on T2 either an 8502, 8504, 8512 or 8514 card needs to be fitted in slot 2.

Made To Measure

James Wren

What types of measurements can I make with a Prosig system?

The purpose of this article is to introduce the different types of transducers that can be used with the Prosig DATS data acquisition systems such as the DATS-solo and DATS-tetrad. The article deals with the design and function of the different types of transducer and the applications they are normally associated with.

Accelerometer

An accelerometer is a device that measures acceleration. It is normally attached directly to the surface of the test specimen. As the object moves, the accelerometer generates an electric current that is proportional to the acceleration. Acceleration and vibration are similar but not the same. If a material or structure has a vibration then it will be subject to certain accelerations. The frequency content and the magnitude of these accelerations are directly proportional to the vibration.

The main type of accelerometer is a piezo electric type. The official name for this type is IEPE, which stands for Integrated Electronic Piezo Electric. This is often referred to by product names such as Piezotron®, Isotron®, DeltaTron®, LIVM™, ICP®, CCLD, ACOtron™ and others, which are all trademarks of their respective owners. The acquisition equipment must include suitable power supplies and signal conditioning to work with the internal electronics of these transducers.

IEPE transducers are usually based on quartz crystals. These transducers normally have the crystal mounted on a mass. When the mass is subjected to an accelerative force a small voltage is induced across the crystal. This voltage is proportional to the acceleration.

There are also many other types of accelerometer in the market: capacitive types, piezo resistive, hall effect, magneto resistive and heat transfer types.

Accelerometers normally come in two distinct packages, side and top mounted. The side mounted package has the interconnecting cable or connector on the side and the top mounted package has them on the top. Different circumstances require different mounting methods. Accelerometers can come in one of two different types: single axis and triaxis. The single axis accelerometer measures acceleration in one direction, where as the triaxial accelerometer measures the acceleration in

Source: Kistler AG

the classical 3 dimensional planes.

Mounting methods are very important for an accelerometer. It is normally best not to rely on just one attachment method from a reliability point of view. The mounting method for an accelerometer is effectively an undamped spring and the frequency response effects of this spring on the frequency related magnitude of the measured vibration must be considered. Consequently, there is no single attachment method that is best for all cases. The most widely used attachment is a sticky wax, although super glue is also very popular. Both have different frequency transmission characteristics as well as other advantages and disadvantages.

Accelerometers are probably the most widely used transducers in any investigative work on a structure of any kind. NVH (Noise, Vibration & Harshness) testing is just one of many fields that make use of them. If there is a requirement to find the frequency of a vibration then an accelerometer would be the obvious choice. It is important to make sure the frequency that is being investigated is within the usable range of the accelerometer and that the maximum amplitude capability of the accelerometer is not exceeded at any point during the test.

Microphones

Microphones are used to measure variations in atmospheric pressure. Variations in pressure that can be detected by the human ear are considered to be sounds.

Acoustics is the science behind the study of sound. Sound can be perceived as pleasing to the ear or to be undesirable.

The human auditory system normally has a maximum range of 20 Hz (or cycles per second) to 20 kHz, although this range generally decreases with age. Sound pressure variations in that frequency range are considered to be detectable by the human ear. A microphone must be able to detect all of these frequencies and in some cases more. Sometimes sound pressure variations outside that frequency range can be important to design engineers as well.

The main types of microphone are the condenser microphone, carbon microphone, magnetic microphone and the piezoelectric microphone. The condenser microphone is the most widely used in situations where quality and accuracy are required. It is capacitive in its design and it operates on the transduction principle in which a diaphragm that is exposed to pressure changes moves in relation to the pressure fluctuations. Behind the diaphragm there is a metal plate, usually called a back plate. This back plate has a voltage applied across it and is effectively a capacitor. As the diaphragm moves closer or further away from the back plate the charge on the plate changes. These changes in charge are then converted to voltages.

A vibration can be considered to be a rapid motion of a particle or a fluctuation of a pressure level. NVH studies are concerned with the study of vibration and audible sounds. These studies focus on reducing the excitations and thus the amplitudes of the sounds, and reducing the transitions between large changes of frequency or shocks. Microphones are used in much wider fields, however, and in some cases they can be used to acquire a sound so that it can be amplified later. Microphones are generally used in any area of testing where the precise input to the human auditory system needs to be measured, for example environmental noise or inside an aircraft cabin.

Strain Gauges

A strain gauge is a transducer used for determining the amount of strain, or change in dimensions in a material when a stress is applied. When the transducer is stretched or strained its resistance changes.

The most common type of strain gauge is the foil gauge type. This is effectively an insulated flexible backing that has a foil pattern upon it. This foil pattern usually forms a 2 wire resistor. The gauge is attached to the structure under test by way of an adhesive. As the structure under test is deformed the foil is also deformed. This deformation causes the length of the foil pattern lines to change. This change in turn affects the gauges resistance.

In order to measure this very small change the gauge is normally configured in a Wheatstone bridge.

Strain gauges are very sensitive to changes in temperature. To reduce the effect of this potential corruption a Wheatstone bridge is used with voltage supply sensing. This reduces the effect of temperature changes.

Strain gauges are also available as thin film types and as semiconductor types. The thin film types are used in higher temperature applications where they are applied directly on to the surface of the structure. This has the additional advantage of not disrupting airflow in aerodynamic design. The semiconductor types of gauges are referred to as piezo resistors. These gauges are used in preference to the resistive foil types when the strain is small. The piezo resistors are the most sensitive to temperature change and are the most fragile strain gauge type.

Strain gauges are used in many disciplines of science and engineering, the classical use for strain gauges is in material and structural fatigue prediction, but strain gauges are also used in areas as diverse as medicine, biology, aircraft structures and even bridge design.

Torque Sensors

Torque measurement is the measurement of the angular turning moment at a particular location and instantaneous time in a shaft.

Torque is usually measured in one of two ways: either by sensing the actual shaft deflection caused by the shaft twisting or by measuring the effects of this deflection. From the measured deflections it is possible to calculate the torque in the shaft.

Classical torque sensors, normally based on strain gauges, are used to measure the deflection in a shaft. These gauges are mounted at 90 degrees to each other. One is mounted in parallel to the main axis of the shaft the other perpendicular to this.

Torque transducers based on strain gauges are often foil types, but can also be diffused semiconductor and thin film types. However, any strain gauge based device will be subject to temperature variations. Unless the change due to temperature is sensed and accounted for then corruption of results can occur. In order to correct for this effect bridge supply sensing is required.

There are also more modern types of torque transducer; these include inductive, capacitive or optical types. These types use a slightly different method to measure the torque in which the angular displacement between two positions on a shaft is measured and the torque deduced from the amount of twist.

Torque transducers are used to measure many parameters such as the amount of power from an automotive engine, an electric motor, a turbine or any other rotating shaft.

Recently torque sensors have been used increasingly as part of hand tools on a production line. This enables the measurement of torque as screws or bolts are tightened, which can be used to improve quality control.

Impact Hammer

The function of an impact hammer is to deliver an impulsive force into a test specimen or material. The force gauge that is built into the hammer measures the magnitude and frequency content of this excitation.

An impact hammer is usually used in conjunction with at least one accelerometer. These accelerometers would normally be single axis, however triaxial accelerometers can also be used. These accelerometers measure the response of the impulsive force in the material. From the combined measurements of the excitation and response a frequency function can be calculated. It is possible to change the frequency characteristics of the impulse by changing the type of 'tip' on the hammer head. A harder tip will generally produce a shorter impact time, and will often be used for situations where higher frequency analysis is of interest.

Impact hammers, sometimes called Modally Tuned® impact hammers, are normally applied manually. These devices often resemble common everyday hammers. They would be used in classical structural or modal analysis situations, although they can also be used for acoustic testing. To use an impact hammer it is necessary to 'hit' the structure with the hammer. The weight and the type of hammer head must be correctly selected for the amount of force required to excite the structure. The bigger and heavier the structure the larger the mass of the hammer head must be and greater the force used.

The reason the impact hammer is sometimes referred to as a Modally Tuned® hammer is that the structural design of the hammer has been optimized so that the structural resonances of the hammer during the 'hit' will not affect the frequencies of interest in the structure and will therefore not corrupt the test data.

Impact hammers are used in many fields from automotive and aerospace design and development through to bridge health assessment.

Pressure Transducers

Most pressure transducers as used in industry today are of a similar appearance; they normally take the form of a cylindrical steel body with a pipe fitting on one end and a cable or connector at the other.

A pressure transducer is a transducer that converts pressure into an electrical signal. There are various types of these transducers on the market, but the most widely used is the strain gauge based transducer. The conversion from pressure into an electrical signal is achieved by the deformation of a diaphragm inside the transducer that has been instrumented with a strain gauge. It is then possible, knowing the relationship of applied pressure to diaphragm deformation, that is the sensitivity, to deduce the pressure.

Pressure transducers are used in any application where the pressure in a gas or fluid is being monitored. Sometimes this monitoring is over long periods, water pressure for example; sometimes it is for very short durations, for example an automotive airbag.

Source: Kistler AG

Pressure transducers normally provide a direct voltage output. However, the IEPE constant current types are increasingly becoming standard. These types of transducers are used in situations where electromagnetic noise is present or when long cabling is required.

Force Transducer

A force transducer is a device for measuring the force exerted upon a particular structure. The force transducer usually measures the deflection in the structure that has been caused by the force, not the force itself.

However, some materials used in the construction of force transducers, when compressed, actually change in electrical characteristics. These materials measure the force directly; they are active sensing elements and have a high frequency response. They can however only normally withstand small amounts of force before damage occurs. The material in question is a certain quartz based crystalline material. These transducers are sometimes called piezoelectric crystal transducers. Electronic charges are formed in the crystal surface in proportion to the rate of change of that force being applied to the crystal. There are other types of force transducer available, for example the ceramic capacitive type.

Typically force transducers are used for bench testing and for monitoring quality during reshaping or bonding operations on a production line.

Force transducers are heavily used in the aerospace industry, they are often used as part of a pilot's controls. It is important in these situations not just to have positional information on the pilots' controls, but also the force being applied to the control system.

Load Cell

A load cell is basically a transducer that converts a load into an electronic signal. The majority of load cells in the market are strain gauge based, however there are some other alternatives. In almost any modern application that involves weighing, a load cell comprised of strain gauges in a Wheatstone bridge configuration is used.

Source: Kistler AG

Load cells based on strain gauges consist normally of four strain gauges bonded on to beam structure that deforms as weight or mass is applied to the load cell. In most cases strain gauges are used in a Wheatstone bridge configuration as this offers the maximum sensitivity and temperature compensation. Two of the four gauges are usually used for temperature compensation.

There are several ways in which load cells can operate internally: bending, shear, compression or tension. All of these types are based on strain gauges.

Less common are the hydraulic load cell and the pneumatic load cell. The hydraulic load cell, as the name implies is based on a fluid under pressure. As the pressure or weight on the cell changes a diaphragm is moved. This type of load cell is more commonly used in situations where temperature can be highly dynamic. Hydraulic load cells can be almost as accurate as the strain gauge based type. Pneumatic load cells operate on the force balance principle; they use multiple dampener chambers to provide higher accuracy than the hydraulic load cell. They would normally be used in situations where the mass being weighed is small, and as they are not based on fluid they can be used in clean room environments. Additionally, they have a very high tolerance of variation in temperature.

Load cells are often used in automotive brake testing and development; the force applied to the pedal is compared to the force generated at the disk or drum, for example. The assist braking system can then be optimized for the expected use. It is possible to adjust the system so that for a given force on the pedal an appropriate force is generated at the brake pad to optimize the vehicle deceleration. Load cells are also used in such diverse areas as engine dynamometry, suspension spring testing, production line batch weighing and production line connector insertion force monitoring.

Thermocouples

A thermocouple is a sensor for measuring temperature. It normally consists of two dissimilar materials joined at one end that produce a small voltage at a given temperature proportional to the difference in temperature of the two materials.

Thermocouples are among the easiest temperature sensors to use, and are used heavily in industry. They are based on the

Seebeck effect that occurs in electrical conductors that undergo a temperature change along their length.

Thermocouples are available in different combinations of metals. The four most common types are the J, K, T and E types. Each has a different temperature range and is intended for use in a different environment.

Thermocouples are often used for temperature measurement of corrosive liquids or gasses, usually at high pressures. Thermocouples are used extensively in the steel and iron production industries; they are used to monitor temperatures through the manufacturing process. Because of their low cost they are suitable for extreme environments where they often have to be replaced. They are a versatile transducer type and are probably the one used in most fields, from aerospace to cryogenic applications.

TEDS - Transducer Electronic Data Sheets

Transducer Electronic Data Sheet or TEDS for short, is a set of electronic data in a standardized format defined within the IEEE-P1451.4 standard. This data specifies what type of sensor is present, describes its interface, and gives technical information such as sensitivity, reference frequency, polarity and so on.

TEDS offers large benefits in that it simplifies troubleshooting, greatly reduces costs and removes the need for re-calibration when changing or replacing sensors.

From the users point of view, when using a TEDS transducer, upon connection certain important fields are automatically uploaded from the transducer microprocessor and can then be used in the test setup matrix. This can be very useful in a test environment. Furthermore, when used in conjunction with a transducer database, all the fields in the test setup matrix will be automatically filled in upon transducer connection.

CAN bus

Whilst not strictly a measurement transducer in its own right the CAN bus and other buses (Flexray, CAN-FD, Ethernet) are having an increasing impact on engineering measurement across a wide range of applications.

The Controller Area Network or CAN bus for short, is a multicast shared serial bus standard, originally developed in the 1980s by Robert Bosch GmbH. The bus is designed for use when connecting electronic control units. CAN was specifically designed to be robust in noisy electromagnetic environments and can utilize a differential balanced line like RS-485. It can be even more robust against noise if twisted pair wire is used.

Although initially created for the automotive market, it is now used in many embedded control applications that may be subject to high levels of external noise. The use of the CAN bus continues to grow in all automotive sectors, and even in the aerospace sector.

Conclusion

All of the transducers and measurement systems in this article are supported by the DATS hardware. However, there are many other types of transducers not discussed here that can be used with the DATS systems. If you have other requirements please contact Prosig to discuss.

What are dB, Noise Floor & Dynamic Range?

James Wren

What are they and why are they important?

Most engineers are probably familiar with or have come across the decibel or dB as a unit of measurement. Its most common use is in the field of acoustics where it is used to quantify sound levels. However, as will be explained in this article, it is also useful for a wide variety of measurements in other fields such as electronics and communications.

One particular use of dB is to quantify the dynamic range and accuracy of an analogue to digital conversion system. This applies to Prosig's DATS range of data acquisition hardware where the noise floor, dynamic range and resolution are all specified in terms of dB.

Decibel (dB)

The decibel is a logarithmic unit of measurement that expresses the magnitude of a physical quantity relative to a **reference** level. Since it expresses a ratio of two quantities having the same units, it is a dimensionless unit.

Definition

A decibel is used for the measurement of power or intensity. The mathematical definition is the ratio (L) of a power value (P_1) to a reference power level (P_0) and in decibels is given by:

$L_{db} = 10Log_{10}(P_1/P_0)$

When considering amplitude levels, A, the power is usually estimated to be proportional to the square of the amplitude and so the following can be used:

$L_{db} = 10Log_{10}(A_1^2/A_0^2)$ or

$L_{db} = 20Log_{10}(A_1/A_0)$

Since the decibel is a logarithmic quantity it is especially good at representing values that range from very small to very large numbers. The logarithmic scale approximately matches the human perception of both sound and light.

Like all logarithmic quantities it is possible to multiply or divide dB values by simple addition or subtraction.

Decibel measurements are always relative to given reference levels and can therefore be treated as absolute measurements. That is, if a particular reference value is known then the exact measurement value can be recovered from one of the equations shown above.

The dB unit is often qualified by a suffix which indicates the reference quantity used, some examples are provided in the following section.

Applications

The decibel is commonly used in acoustics to quantify sound levels relative to a reference. This may be to compare two sound sources or to quantify the sound level perceived by the human ear. The decibel is particularly useful for acoustic measurements since for humans the ratio of the loudest sound pressure level to the quietest level that can be detected is of the order of 1 million. Furthermore, since sound power is proportional to the pressure squared then this ratio is approximately 1 trillion.

For sound pressure levels, the reference level is usually chosen as 20 micro-pascals (20 μPa), or $2x10^{-5}$ Pa. This is about the limit of sensitivity of the human ear.

Note that since the most common usage of the decibel unit is for sound pressure level measurements it is often abbreviated to just dB rather than the full dB(SPL).

The common decibel units used in acoustics are:

dB(SPL)	Sound Pressure Level. Measurements relative to $2x10^{-5}$ Pa.
dB(SIL)	Sound Intensity Level. Measurements relative to 10^{-12} W/m^2 which is approximately the level of human hearing in air.
dB(SWL)	dB Sound Power level. Measurements relative to 10^{-12} W.

The human ear does not respond equally to all frequencies (it is more sensitive to sounds in the frequency range from 1 kHz to 4 kHz than it is to low or high frequency sounds). For this reason sound measurements often have a weighting filter applied to them whose frequency response approximates that of the human ear (A-weighting). A number of filters exist for different measurements and applications, these are given the names A,B,C and D weighting. The resultant measurements are expressed, for example, as dBA or dB(A) to indicate that they have been weighted.

In **electronics** and **telecommunication**, the decibel is often used to express power or amplitude ratios in order to quantify the gains or losses of individual circuits or components. One advantage of the decibel for these types of measurements is that, due to its logarithmic characteristic, the total gain in dB of a circuit is simply the summation of each of the individual gain stages in dB.

In electronics the decibel can also be combined with a suffix to indicate the reference level used. For example, dBm indicates power measurement relative to 1 milliwatt. The following are some common decibel units used in electronics and telecommunications.

dBm	Power measurements relative to 1mW
dBW	Power measurements relative to 1W. Note that $L_{dBm} = L_{dBW} + 30$
dBk	Power measurements relative to 1kW. Note that $L_{dBm} = L_{dBk} + 60$
dBV	Voltage measurement relative to 1V – regardless of impedance
dBu or dBv	Voltage relative to 0.775V and is derived from a 600 Ohm dissipating 0dBm (1mW)
dBμ	Electric field strength relative to 1μV per meter
dBJ	Energy relative to 1 joule. Used for spectral densities where 1 joule = 1 W/Hz

Examples

If the numerical value of the reference is undefined then the decibel may be used as a simple measure of relative amplitudes. As an example, assume there are two loudspeakers, one emitting a sound with a power P_1 and a second one emitting the same sound at a higher power P_2. Assuming all other conditions are the same then the difference in decibels between the two sounds is given by:

$10 \log (P_2/P_1)$

If the second speaker produces twice as much power than the first, the difference in dB is

$10 \log (P_2/P_1) = 10 \log 2 = 3$ dB.

If the second had 10 times the power of the first, the difference in dB would be

$10 \log (P_2/P_1)= 10 \log 10 = 10$ dB.

If the second had a million times the power of the first, the difference in dB would be

$10 \log (P_2/P_1) = 10 \log 1000000 = 60$ dB.

Note that if both speakers produce the same power then the difference in dB would be

$10 \log (P_2/P_1) = 10 \log 1 = 0$ dB.

This illustrates some common features of the dB scale irrespective of the measurement type:

- A doubling of power is represented approximately by 3dB and a doubling of amplitude by 6dB.
- A halving of power is given by -3dB and a halving of amplitude by -6dB
- 0dB means that the measured value is the same as the reference. Note that this does **not** mean there is no power or signal.

Noise floor

Any practical measurement will be subject to some form of noise or unwanted signal. In acoustics this may be background noise. In electronics there is often thermal noise, radiated noise or any other interfering signals. In a data acquisition measurement system the system itself will actually add noise to the signals it is measuring. The general rule of thumb is: the more electronics in the system the more noise imposed by the system.

In data acquisition and signal processing the noise floor is a measure of the summation of all the noise sources and

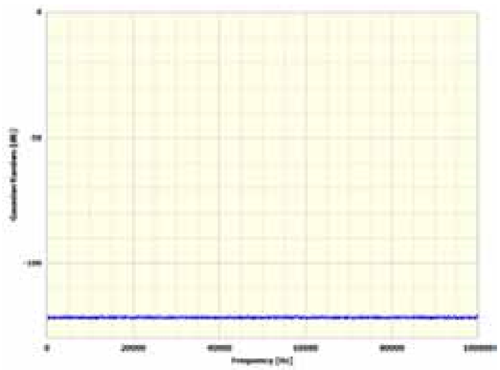


Figure 1

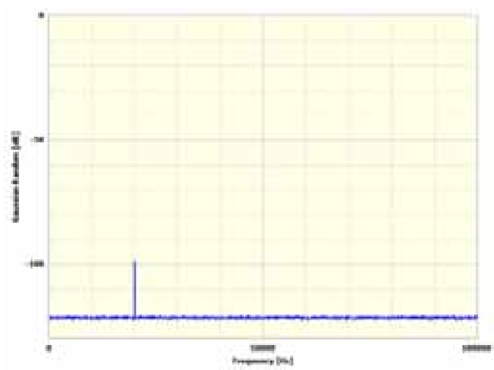


Figure 2

unwanted signals generated within the entire data acquisition and signal processing system. The noise floor limits the smallest measurement that can be taken with certainty since any measured amplitude cannot on average be less than the noise floor.

In summary, the noise floor is the level of background noise in a signal, or the level of noise introduced by the system, below which the signal being captured cannot be isolated from the noise.

As shown in Figure 1 the noise floor is better than -120 dB.

Figure 2 shows that only signals above the noise floor can be measured with any degree of certainty. In this case the signal level of -100dB at 20KHz could be measured. If however, the noise floor increased above the -120dB level then it would become more difficult to measure this signal.

For example, it is possible for the human ear to hear a very low sound such as a pin drop or a whisper. However, this is only possible if the noise floor or background noise of the particular environment is very low such as in a soundproof or quiet room. It would not be possible to hear or discriminate such low levels in a noisy room.

Various techniques are employed by the DATS data acquisition system in order to ensure that the noise floor of the equipment is kept as low as possible. These include signal-processing functions as well as practical features such as the ability to disable cooling fans during acquisition scans.

Dynamic range and resolution

Dynamic range is a term used to describe the ratio between the smallest and largest signals that can be measured by a system.

The dynamic range of a data acquisition system is defined as the ratio between the minimum and maximum amplitudes that a data acquisition system can capture.

In practice most Analogue to Digital Converters (ADC) have a voltage range of ± 10V. Sometimes amplification may be applied to signals before they are input to an ADC in order to maximize the input voltages within the available ADC range.

The resolution of a measurement system is determined by the number of bits that the ADC uses to digitise an input signal. Most ADCs have either 16-bit or 24-bit resolution. For a 16-bit device the total voltage range is represented by 2^{16} (65536) discrete digital values. Therefore the absolute minimum level that a system can measure is represented by 1 bit or 1/65536th of the ADC voltage range. For a system with a voltage range of ±10V then the smallest voltage that the system can distinguish will be:

$$20 / 65536 = 0.3 \text{ mV}$$

In decibels this dynamic range is therefore expressed as:

$$20 \text{ Log}_{10} (1 / 65536) = 96\text{dB}$$

Therefore for a 16-bit ADC the dynamic range is 96dB. Using the same calculations the dynamic range of a 24-bit ADC is 144dB.

The noise floor of a measurement system is also limited by the resolution of the ADC system. For example, the noise floor of a 16-bit measurement system can never be better than -96dB and for a 24-bit system the lower limit is limited to -144 dB. In practice, however, the noise floor will always be higher than this due to electronic noise within the measurement system.

Modern data acquisition systems, such as the DATS-solo and DATS-tetrad, employ a number of sophisticated digital signal processing techniques to improve the amplitude resolution and thereby allow low amplitude data, such as noise floor signals, to be measured with greater precision and with greater accuracy.

Strain Gauges Explained

What is a strain gauge and how does it work?

James Wren

A strain gauge is an electrical sensor which is used to accurately measure strain in a test piece. Strain gauges are usually based on a metallic foil pattern. The gauge is attached to the test piece with a special adhesive. As the test piece is deformed, so the adhesive deforms equally and thus the strain gauge deforms at the same rate and amount as the test piece. It is for this reason that the adhesive must be carefully chosen. If the adhesive cracks or becomes detached from the test piece any test results will be useless.

Strain gauges are used not just for metals; they have been connected to the retina of the human eye, insects, plastics, concrete and indeed any material where strain is under investigation. Modern composite materials like carbon fibre

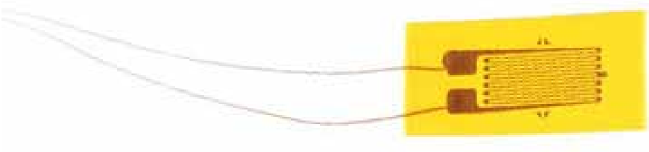


Figure 1: A strain gauge

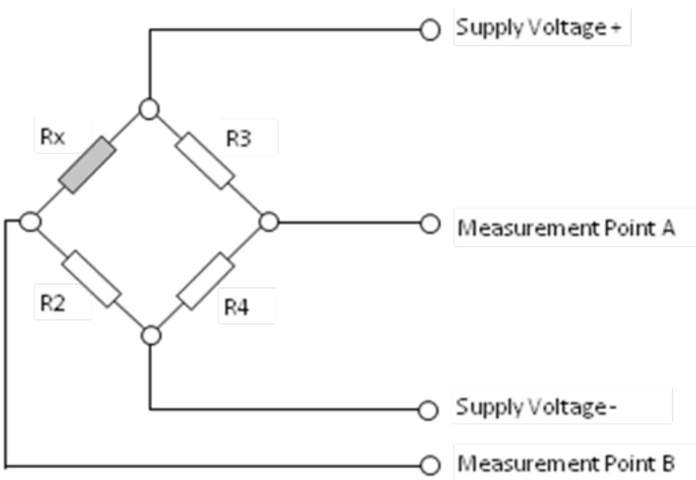


Figure 2: A Wheatstone bridge

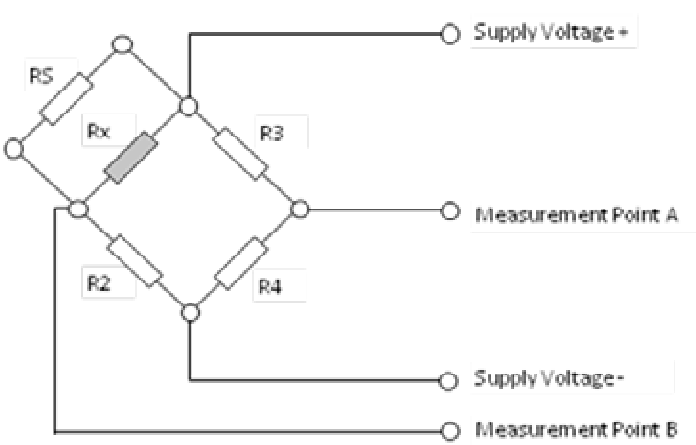


Figure 3: With shunt resistor

when under development are often constructed with strain gauges between the layers of the material.

The strain gauge is effectively a resistor. As the strain increases so the resistance increases.

In a basic sense a strain gauge is simply a long piece of wire. Gauges are mostly made from copper or aluminium (Figure 1). As the wire in the strain gauge is mostly laid from end to end, the strain gauge is only sensitive in that direction.

When an electrical conductor is stretched within the limits of its elasticity it will become thinner and longer. It is important to understand that strain gauges actually deform only a very small amount, the wire is not stretched anywhere near its breaking point. As it becomes thinner and longer it's electrical characteristics change. This is because resistance is a function of both cable length and cable diameter.

The formula for resistance in a wire is

$$\text{Resistance in ohms (R)} = \frac{\rho \times L}{\alpha}$$

ρ = Resistivity (ohms per meter)
L = length in meters
α = cross section (m^2)

For example, the resistance of a copper wire which has a resistivity of $1.8 \times 10^{-8}\ \Omega/m$, is 1 meter long and has a cross sectional area of $2mm^2$ would be

$$R = \frac{1.8\times10^{-8} \times 1}{0.002^2} = \frac{0.000000018}{0.000004} = 0.0045\Omega$$

Resistivity is provided by the manufacturer of the material in question and is a measurement of how strongly the material opposes the flow of current. It is measured in ohm's per meter (Ω/m).

If in our example the cable was then put under appropriate strain its length would increase and its cross sectional area would decrease. Suppose the cable extended to 2 meters in length and the cross sectional area decreased to $0.5mm^2$, the resistance now would be

$$R = \frac{1.8\times10^{-8} \times 2}{0.005^2} = \frac{0.000000018}{0.00000025} = 0.072\Omega$$

As can clearly be seen the resistance is now different, but the resistances in question are very small. This example shows only the difference when the characteristics of the copper wire have changed. It is not practically possible to stretch and extend a piece of copper wire by such amounts. The example merely shows how resistance changes with respect to length and cross sectional area and demonstrates that strain gauges, by their very nature, exhibit small resistance changes with respect to strain upon them.

These small resistance changes are very difficult to measure. So, in a practical sense, it is difficult to measure a strain gauge, which is just a long wire. Whatever device is used to measure the strain gauge's resistance will itself have its own resistance. The resistance of the measuring device would almost certainly obscure the resistance change of the strain gauge.

The solution to this problem is to use a Wheatstone bridge to measure the resistance change. A Wheatstone bridge is a device used to measure an unknown electrical resistance. It works by balancing two halves of a circuit, where one half of the circuit includes the unknown resistance. Figure 2 shows a classical Wheatstone bridge, Rx represents the strain gauge.

Resistors R2, R3 and R4 are known resistances. Typically, 120Ω, 350Ω or 1000Ω resistors are used depending on the application. Knowing the supply voltage and the returned signal voltage it is possible to calculate the resistance of Rx very accurately.

For example if R2, R3 and R4 are 1000Ω and if the measured signal voltage between measurement points A and B was 0 Volts then the resistance of Rx is

$$\frac{R3}{R4} = \frac{Rx}{R2} \quad \text{or} \quad Rx = \frac{R3}{R4} \times R2$$

For our example we get

$$Rx = \frac{1000\Omega}{1000\Omega} \times 1000\Omega = 1000\Omega$$

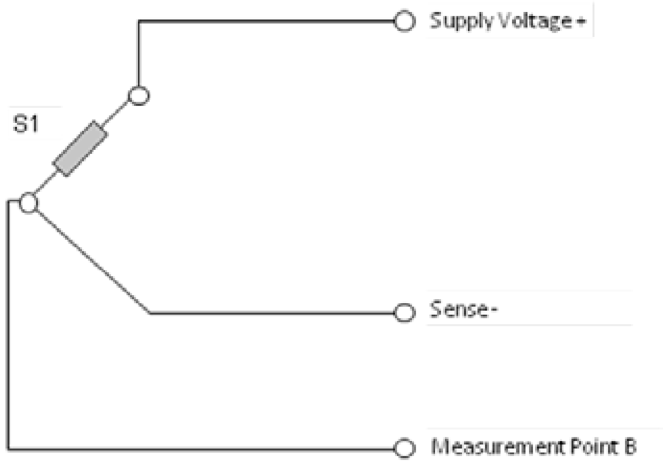


Figure 4: Quarter bridge

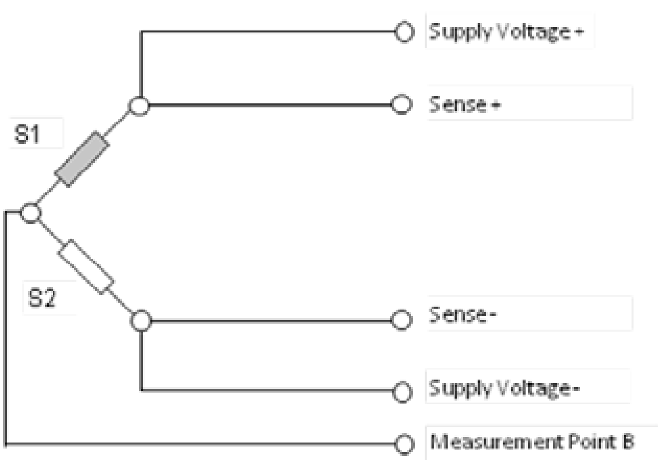


Figure 5: Half bridge

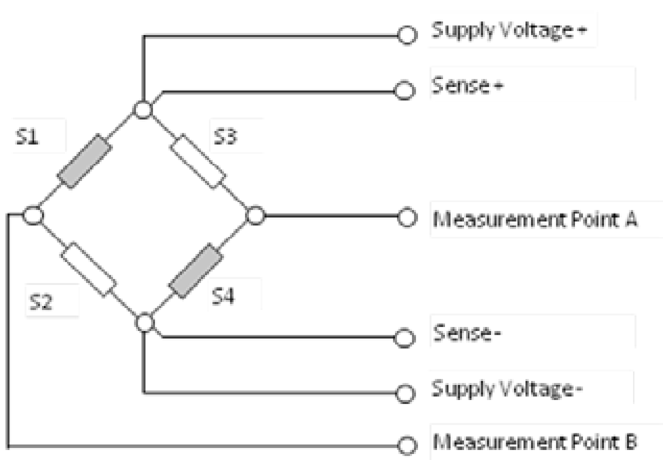


Figure 6: Full bridge

This implies a perfectly balanced bridge. In practice, because the strain gauge goes through different strain levels its resistance changes and the measured signal level between measurement points A and B is not zero. This is not a problem when using a system like the DATS-tetrad as it can accurately measure the voltage between measurement points A and B.

It is necessary to know the relationship between resistance and voltage. Only then can the measured voltage be related to a resistance and, hence, a strain value.

Figure 3 shows the addition of another resistor RS, called the shunt resistor. The shunt resistor is a known fixed value, normally 126,000Ω.

The Shunt resistor is added for calibration purposes and is a very high precision resistor. By measuring the voltage between measurement points A and B with the known shunt resistor across Rx and also with the shunt resistor removed it is possible to relate the measured voltage change to a known resistance change. Therefore the volt per ohm value is known for this particular bridge and this particular Rx.

In order to go one step further and calculate the strain from the resistance, the gauge factor must be known. This is a calibrated number provided by the manufacturer of the strain gauge. With this information the sensitivity of the whole sensor may be calculated in terms of volts per strain.

Inside the DATS-tetrad the resistors used to complete the bridge are very high precision. This allows the Tetrad to calculate the resistance, and therefore, strain with a high degree of accuracy.

Strain gauge readings can be affected by variations in the temperature of the strain gauge or test piece. The wire in the strain gauge will expand or contract as an effect of thermal expansion. This will be detected as a change in strain levels by the measuring system as it will manifest itself as a resistance change. In order to address this most strain gauges are made from Constantan or Karma alloys. These are designed so that temperature effects on the resistance of the strain gauge cancel out the resistance change of the strain gauge due to the thermal expansion of the test piece. Different materials have different thermal properties and hence differing amounts of thermal expansion.

So, where temperature change during the test is an issue, temperature compensating strain gauges can be used. However this requires correctly matching the strain gauge alloy with the thermal expansion properties of the test piece and the temperature variation during the test. In certain circumstances temperature compensating strain gauges are either not practical nor cost effective. Another more commonly used option is to make use of the Wheatstone bridge for temperature compensation.

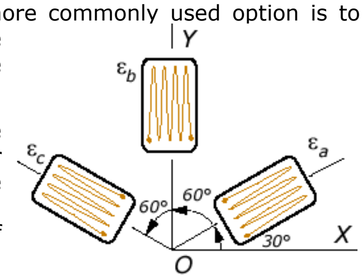


Figure 7: Strain gauge rosette

When using a Wheatstone bridge constructed of four strain gauges, it is possible to attach the four gauges in a fashion to remove the effect of changes in resistance caused by temperature variation. This requires attaching the strain gauge Rx in the direction of interest and then attaching the remaining strain gauges, R2, R3 and R4, perpendicular to this. The piece under test however must only exhibit strain in the direction across Rx and not in the perpendicular direction.

It's important to understand that the R2, R3 and R4 strain gauges should not be under strain, hence their direction. This means the whole bridge is subject to the same temperature variations and therefore stays balanced from a thermal expansion point of view. As the resistance changes due to temperature, all the resistances in all four gauges change by the same amount. So the voltage at measurement points A and B due to temperature fluctuations stays constant. Only the strain in the desired direction, across Rx, in the test piece affects the measured voltage readings.

The DATS-tetrad system has multi-pin inputs, these allow for the connection of strain gauges in all the various different bridge configurations.

The strain gauge configurations are,

Quarter Bridge is the most common strain gauge configuration. As can be seen in Figure 4 it is actually a three wire configuration. The rest of the bridge as shown in Figure 2 is completed inside the DATS-tetrad system. Quarter Bridge uses three wires to allow for accurate measurement of the actual voltage across S1.

Half Bridge is not an often used strain gauge configuration. As can be seen in Figure 5 it is actually a five wire configuration. The rest of the bridge as shown in Figure 2 is completed inside the DATS-tetrad system. The main advantage of the Half Bridge configuration is that both the strain gauges S1 and S2 can be attached to the test piece, but perpendicular to each other. Which as previously discussed allows for temperature compensation.

Full bridge is used for situations where the fullest degree of accuracy is required. The Full Bridge configuration is a six wire system, as shown in Figure 6. The Full Bridge configuration is the most accurate in terms of temperature variation because it can have two active gauges, S1 and S4. The gauges can be configured with S1 and S4 in the direction of interest on the test piece and S2 and S3 perpendicular to this. Further the voltage sense lines have no effective current flow and therefore have no voltage drop, therefore the voltage measured by the DATS-tetrad system is the actual voltage that is exciting the bridge. The reason for this requirement is that strain gauges are often on long wires and all wires have their own resistance. The DATS-tetrad system could be exciting the gauge with 5 Volts for example, but the voltage at the active part of the bridge might be 4.95 Volts because of the resistance of the wires carrying the supply voltage. This small change once measured using the sense lines can be allowed for automatically in the strain calculations inside the data acquisition system.

Strain gauge measurements with direction.

Strain Gauges can be configured in a particular pattern that allows for the calculation of the overall strain component, this is often referred to as a strain gauge rosette. As shown in Figure 7, three strain gauges are placed either very close together or in some cases on top of each other. These may be used to measure a complex strain, the strain is complex because it has both amplitude and a direction. Using the Prosig DATS software it is possible to calculate the principal component of the strain, the amplitude over time and to calculate the direction as an angle from the reference X axis over time.

Mounting Accelerometers

A summary of modern methods and best practice

Adrian Lincoln

This article is an overview of information gathered from discussions with automotive testing engineers in the USA and UK. It reflects modern practice in testing in the area of design and refinement and also in end-of-line production applications where not only accuracy but speed of measurement is a primary consideration.

Introduction

In many vehicle testing situations, the mounting of a transducer needs to be treated with as much importance as the selection of the transducer itself. If the motion of the test structure is not accurately transmitted to the transducer, it cannot be accurately measured. The choice of transducer mounting method, the surface flatness and surface preparation can significantly affect the amplitude-frequency response of the measurement particularly at high frequencies. When attaching a transducer to a structure using adhesives, the stiffness and strength of the glue will affect the usable frequency range of the transducer. It is also important that any mounting method that is different from that used for calibration should be characterized for its dynamic characteristics over the intended frequency and amplitude range.

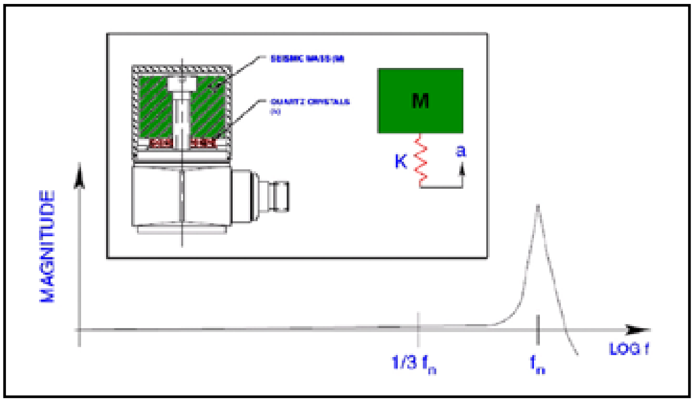


Figure 1: Accelerometer schematic

Theory of Accelerometer Operation

An accelerometer is an inertial measurement device that converts mechanical motion into an electrical signal. It consists of a piezoelectric (quartz) crystal and a seismic mass enclosed

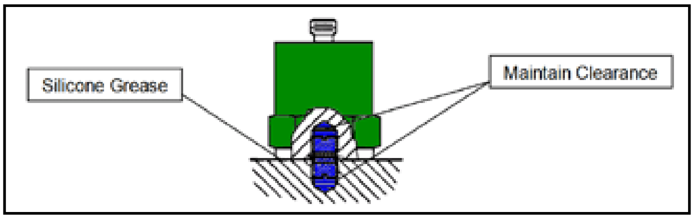


Figure 2: Example of stud mounting

in a protective metal case. Acceleration is transmitted from the surface of a vibrating structure into the case of the accelerometer through the base. As the force is applied to the crystal, the crystal creates a charge proportional to acceleration. The charge output is measured in pico Coulombs per g (pC/g) where g is the force of gravity, or pico Coulombs per metres/second^2 (pC/m/sec^2). Some sensors have an internal charge amplifier, while others have an external charge amplifier. The charge amplifier converts the charge output of the crystal to a proportional voltage (mV/g or mV/m/sec^2). By design, accelerometers have a natural resonance (corresponding to f_n in figure 1) which is 3 to 5 times higher than the advertised high end frequency response. The operating frequency response range is limited to the lower part of the operating range that has a flat frequency response. The advertised range is achievable only by using bolt (threaded stud) mounting. Any other mounting method will lower the frequency of the natural resonance, and consequently reduce the usable frequency response range.

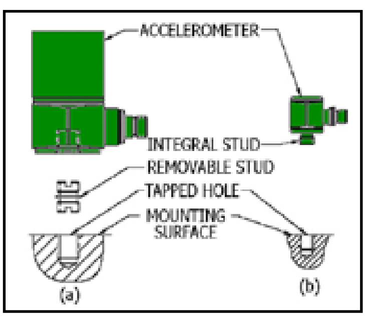


Figure 3: Removable stud v integral stud

Mounting using Threaded Studs

For permanent installations, where a very secure attachment of the accelerometer to the test structure is preferred, stud mounting is recommended. The stud may be integral, i.e., machined as part of the accelerometer or it may be separate (removable). Stud mounting provides higher transmissibility than any other method. The transducer should be mounted using the specified stud or screw, so that the entire base of the transducer is in intimate contact with the surface of the test article. A smooth, flat area at least the size of the sensor base should be ground or machined on the test object according to the manufacturer's specifications. Then a hole must be tapped in accordance with the supplied installation drawing, ensuring that the hole is perpendicular to the mounting surface. When installing accelerometers with the mounting stud it is important that the stud does not reach the bottom of either the mounting surface or accelerometer base (see figure 2). Good mounting studs have depth-limiting shoulders that prevent the stud from bottoming-out into the accelerometer's base. Each base incorporates a counter bore so that the accelerometer does not rest on the shoulder. Any stud bottoming or interfering between the accelerometer base and the structure will inhibit acceleration transmission and affect measurement accuracy. When installing a stud, it is best to first thread the stud into the accelerometer to ensure that the stud fully enters the threaded hole, then to thread the accelerometer into the mounting hole until the surfaces meet and finally screw in place using a torque wrench.

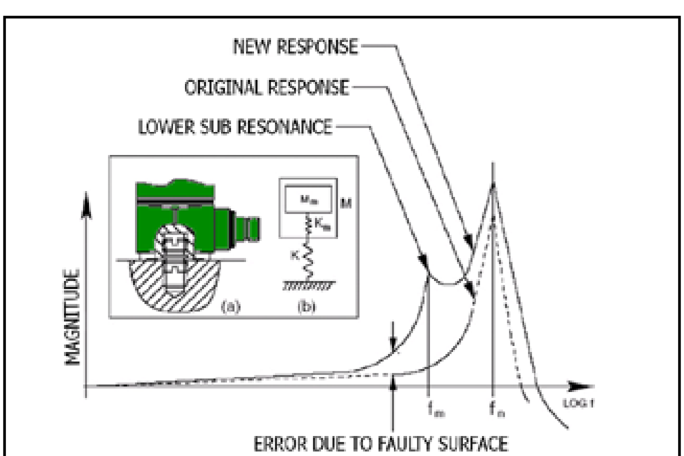


Figure 4: Error due to faulty surface

Any nicks, scratches, or other deformations of the mounting surface or the transducer will affect frequency response. They may also result in damage to the accelerometer. With regards to surface preparation, good machine-shop practices are usually adequate - Surface Flatness 0.076 mm TIR (Total Indicator Runout), Surface Roughness 0.8 µm, Perpendicularity of Hole: 1 degree ± 0.5°, Tap Class 2. (when using studs). Also, a thin application of a light lubricant such silicone grease will improve transmissibility by filling voids with nearly incompressible fluid, thereby increasing the compressive stiffness of the joint. This is particularly important for measurements above 2 kHz, where changes in resonance have a significant effect on measurements. A torque wrench should be used to mount all accelerometers ensuring repeatability in the installation of the transducers and preventing thread damage. A thread-locking compound may be applied to the threads of the mounting stud to safeguard against loosening.

Two stud mount designs are illustrated in figure 3, the removable stud (a) and the integral stud (b).

The removable stud style of accelerometer is the most popular for several reasons:

1. The removable stud allows easy access to the mounting surface of the accelerometer for restoration of surface flatness should this become necessary. Even with normal care, eventually after many installations, the mounting surface of the accelerometer may become worn or damaged to a point where it is no longer flat enough to achieve a satisfactory mount and frequency response will be compromised as shown in figure 4. It is a simple matter to restore flatness if the stud can be removed and the accelerometer base can be applied directly to a lapping plate for restoration of flatness. When the stud is integral and cannot be removed, refurbishment of the mounting surface becomes very difficult and can only be performed at the factory.

2. If the integral stud is broken or the threads become stripped or otherwise damaged, the transducer becomes unusable. On the other hand, the removable stud can be easily replaced.

3. At times, with radial connector style accelerometers it is important during installation, to orient the connector so that nearby obstacles may be avoided. By exchanging mounting studs, the desired orientation may be obtained.

4. The removable stud type accelerometer can also be adhesive mounted without using a mounting adapter.

Mounting using Threaded Screws (Bolts)

When installing accelerometers onto thin-walled structures, a cap screw passing through a hole of sufficient diameter is an

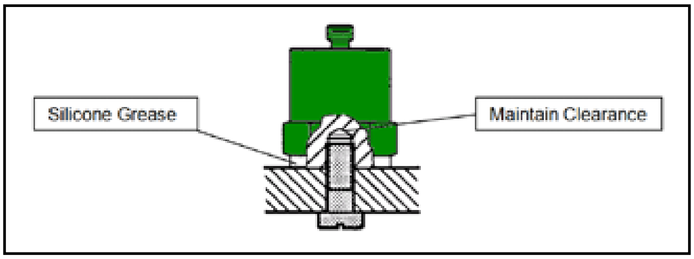


Figure 5: Mounting using threaded screws

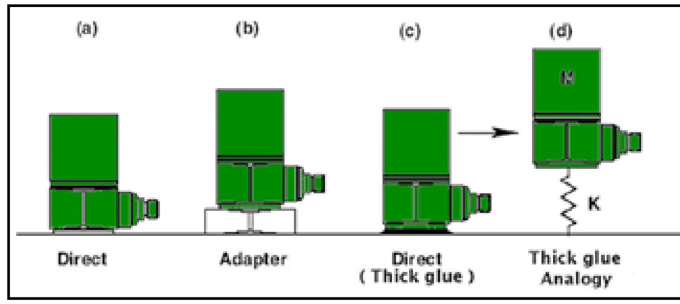


Figure 6: Direct, adapter and different glue mountings

acceptable means of securing the accelerometer to the structure. The screw engagement length as usual should always be checked to ensure that the screw does not bottom into the accelerometer base. A thin layer of silicone grease at the mounting interface ensures high-frequency transmissibility.

Mounting using Adhesives

Occasionally, mounting by stud or screw is impractical. For such cases, adhesive mounting offers an alternative method of attachment without the need for extensive machining. However, this mounting method will typically reduce the operational frequency response range. This reduction is due to the damping and stiffness qualities of the adhesive. Also, removal of the accelerometer is more difficult than any other attachment method. For this reason the use of separate adhesive mounting bases is recommended as it prevents the adhesive from damaging the accelerometer base or from clogging the mounting threads. Some adhesive mounting bases also provide electrical isolation, which eliminates potential noise pick-up and ground loop problems.

In the case of miniature accelerometers stud mounting is not an option. Most of them can only be mounted using an adhesive, which then becomes part of the structure being measured. Often they are provided with the integral stud removed to form a flat base. The stiffness of the cured adhesive is critical to the measurement performance of the total system. No adhesive is as stiff as a normal mounting stud. The more adhesive joints there are between the test structure and the accelerometer, the greater the degradation of transmissibility. Also, surface cleanliness is of prime importance for proper adhesive bonding. Since the manufacturer calibrates his transducer using a specific mounting adhesive, it is critical to follow the manufacturer's recommendation in obtaining the intended performance. Different adhesives should be evaluated over the intended frequency and amplitude range.

Figure 6 illustrates two adhesive mount installations: one a direct mount and the other a mount with an adhesive adapter. Diagram (c) shows a direct mount with an undesirable thick layer of glue and (d) shows the mechanical analogy of the thick glue layer mount. In this situation the layer of adhesive acts like a spring and creates an effect similar to that caused by surface deformation as shown earlier.

To avoid mounting problems caused by thick glue layers it is better to use a cyanoacrylate adhesive, sometimes known as "Super Glue" or "Instant Bond" adhesive. This type of adhesive is widely available and has the following advantages:

1. It sets very quickly.

2. Not much adhesive is required for a strong bond so glue depths will naturally tend to be very thin.

3. It can be removed easily with acetone.

4. It has the best coupling characteristics at room temperature over a wide frequency range.

Dental cement is also worth considering. It is highly rigid which results in acceptable transmissibility characteristics even though a slightly thicker layer of glue has to be used. However, the problem with dental cement lies with its strength and tenacity; there is no suitable solvent available that readily dissolves it, so removal of the accelerometer can result in damage to the transducer.

Hot glue (glue gun) adhesive is the least effective in terms of rigidity and hence transmissibility, but it can be easily applied and removed, and is therefore quite popular with engineers who need to perform quick tests.

A variety of adhesives are available from many manufacturers, who usually provide specification charts and application notes for their adhesives. For example Loctite provides a wide range of adhesives in its "Automotive Aftermarket" division. For applications at extreme temperatures, there are commercially available adhesives that are specifically formulated to handle the hot or cold environments. For cryogenic applications, at room temperature cure, a two component polymer epoxy resin system has been proven to be effective down to -200°C. It is important for a low-temperature adhesive to be able to withstand cryogenic thermal shock without showing signs of cracking. For applications at very high temperature (up to 700°C), ceramic based adhesives are typically used due to their heat resistant properties. But ceramic adhesives also require a high curing temperature, which prevents their use in most transducer mounting applications. At lower temperatures (from a maximum of 200 to 300°C), a few commercial suppliers offer proprietary modified epoxy resins that are room temperature cured, and can operate up to 260°C.

At normal temperatures, anodized aluminium-cementing studs for adhesively mounting a stud mount accelerometer can be used. For higher temperature requirements stainless steel studs may be required.

Alternatively, when higher temperatures are involved, aluminized Mylar tape can be applied to the test structure and the accelerometer mounted with an adhesive base using an appropriate high temperature adhesive. After the test, the tape can be easily removed without damaging the surface finish of the structure.

In general the dismounting of any adhesively-mounted transducer must be carried out with great care. It should not be removed with impacts, but instead with solvents, allowing softening of the bond, supplemented by light shearing torque. All traces of adhesives should be removed using recommended solvents only. Most damages to miniature accelerometers are caused by improper removal techniques

Mounting using Magnetic Adapters

The magnetic mounting method is typically used for temporary measurements with a portable data collector or analyzer. They are popular in industrial vibration monitoring applications where quick point- to-point measurements are to be made periodically. This method is not recommended for permanent monitoring, because the transducer may be inadvertently moved and the multiple surfaces and materials of the magnet may interfere with or increase high frequency signals.

Special attention is required when using a magnetic mounting adapter. During installation, the magnetic force that pulls the adapter/accelerometer assembly towards the mounting structure often induces an unexpectedly high level of shock input to the accelerometer at the time of contact, causing damage in the sensing elements or the internal electronics. Effective use of magnets for mid-level frequencies requires detailed surface preparation, which may extend the overall test timeframe.

Wedged, dual-rail magnetic bases are generally used for installations on curved surfaces, such as motor and compressor housings and pipes. However, dual-rail magnets usually significantly decrease the operational frequency range of an accelerometer. For best results, the magnetic base should be attached to a smooth, flat surface. A thin layer of silicone grease should be applied between the sensor and magnetic base, as well as between the magnetic base and the structure.

When surfaces are uneven or non-magnetic, steel pads can be welded or epoxy-glued in place to accept the magnetic base. Use of such a pad ensures that periodic measurements are taken from the exact same location. This is an important consideration when trending measurement data.

Mounting using Triaxial Blocks and Isolation Adapters

Many installations require the transducer to be mounted on an adapter block for triaxial (three orthogonal axes) measurement, or for electrical ground isolation purposes. The block itself becomes part of the structure being measured, and acts as an additional spring mass system, whose transfer function needs to be defined before use. To maximise transmissibility, a good mounting block or adapter should be as small, lightweight and stiff as possible. The ideal material is beryllium, but it is not commonly used due to safety regulations and cost. Other materials, such as magnesium or aluminium are widely used with some compromise in transmissibility above 10kHz. It is therefore recommended that the accelerometers be calibrated together with the mounting block or adapter. There are triaxial accelerometers on the market that come in a single housing, designed to minimise mounting block related effects. There are also transducers that feature built-in electrical ground isolation, which eliminates the use of an isolation adapter.

Ground Isolation, Ground Noise and Ground Loops

When installing accelerometers onto electrically conductive surfaces, there is always a possibility of ground noise pick-up. Noise from other electrical equipment and machines that are grounded to the structure, such as motors, pumps, and generators, can enter the ground path of the measurement signal through the base of a standard accelerometer. When the sensor is grounded at a different electrical potential than the signal conditioning and readout equipment, ground loops can occur. This phenomenon usually results in current flow at the line power frequency (and harmonics thereof), potential erroneous data and signal drift. Under such conditions, it is advisable to electrically isolate or “float” the accelerometer from the test structure. This can be accomplished in several ways. Most accelerometers can be provided with an integral ground isolation base. Some standard models may already include this feature, while others offer it as an option. The use of insulating adhesive mounting bases, isolation mounting studs, isolation bases, and other insulating materials, such as paper beneath a magnetic base, are effective ground isolation techniques. It is important to note that any additional ground-isolating hardware can reduce the upper frequency limits of the accelerometer.

Automotive Applications

Typically, stud mounting of transducers is not often used as usually transducers have to be attached and removed very frequently. Stud mounting directly to the structure is only used for very special development tests where the transducers are mounted only once and a series of tests is performed without removing the transducer. Clearly, this is not a practical option for testing customer vehicles or testing vehicles from the assembly line. As an alternative, a transducer can be stud mounted to a small, light-weight aluminium or titanium block and the block in turn attached to the structure using an adhesive. The block can be machined to the desired shape so that the transducer can either be mounted at a specified location which may not have a flat surface or mounted on a surface that is parallel or perpendicular to one of the vehicle’s major coordinates.

For many automotive applications hot glue adhesive is used despite the restriction this imposes on the effective frequency range. However, since the frequency range of interest is typically less than 1000 Hz this is not usually a problem. Good hot glue will provide sufficient adhesion for small to moderate sized accelerometers so there are no concerns about the proximity of the resonant frequencies of the mounted transducers.

Bees wax or petroleum wax is not widely used, but if it is used then the temperature must be less than 20°C. However, wax adhesive is sometimes preferred for hammer impact testing or in general Frequency Response Function testing when the response transducer needs to be moved frequently.

Cyanoacrylate glue (super-glue) is sometimes used in calibration laboratories, but this method is rarely used for mounting transducers on vehicles.

For transducer mounting positions which experience elevated temperatures (typically under vehicles or in engine compartments) dental cement is used especially when performing (hot) tests on running vehicles. Mounting blocks are often used to keep adhesive cement away from the threaded holes of the transducer.

Mounting transducers on vehicle steering wheels presents a different sort of problem which is the need to avoid marking or destroying the steering wheel. One solution is to wrap duct tape around the steering wheel rim at the desired position and then use a clamp such as a hose clamp to grip a mounting block to the rim. If desired the mounting block can be attached to the clamp by adhesive such as hot glue. The transducer itself may then be stud mounted to the mounting block.

Magnetically mounted transducers are sometimes used for quick vehicle setup. This technique is not widely used, but can be useful if the test engineer can find a suitable magnetic flat surface to which to attach the mounting magnet.

Recommendations

There is no one “best” mounting method for all applications because of the many different structural and environmental considerations, such as temporary or permanent mount, temperature, type of surface finish, and so forth. Almost any of the mounting methods described earlier when used at low acceleration levels provides the full frequency range of use if the mounting surface is reasonably flat.

As surface irregularities increase or the thickness of the adhesive increases, the usable frequency range decreases. The less-stiff, temporary adhesives reduce an accelerometer’s usable frequency range much more than the more rigid, harder adhesives. Nevertheless, temporary adhesives are quite satisfactory for low-frequency (<1000 Hz) structural testing at room temperatures.

When using adhesives, problems can be expected at high frequencies in proportion to the size of mass of the accelerometer. If possible, an accelerometer should be calibrated using a back-to-back accelerometer system using exactly the same mounting method that will be used in the actual test. In this manner, the precise behavior of the measurement system can be determined at the expected frequencies.

What Is The Difference Between Single-Ended & Differential Inputs?

James Wren

Prosig DATS-tetrad systems use differential inputs, but what are they and why are they so special?

This subject is not always fully understood and, therefore, the focus of this article is to try to make the difference clearer and explain why differential inputs are the obvious choice in a high quality, high precision system like the DATS-tetrad.

First, we need to understand what we mean by single ended and differential inputs. To that end we should start with amplifiers. The difference between single ended and differential ended inputs is the difference between the types of amplifiers used. Most modern data acquisition systems will have some sort of signal amplifiers.

Amplifiers will, quite simply, take a signal and amplify it. For example, in Figure 1 we have a sine wave with an amplitude of ±5 Volts. If this were amplified by a factor of two, the result would be as shown in Figure 2 - a sine wave with an amplitude of ±10Volts.

Therefore, if we consider a simple amplifier, the signal into the amplifier is amplified to the output. In Figure 1 consider that the sine wave was actually carried on two copper wires. One would be connected to earth or ground and the other would carry the AC (alternating current) signal as seen in Figure 1.

There are however problems with this method of cabling and transporting signals. The main problem is that ground is not a constant 0V, but is, in reality, at different levels in different places. The closer these places are together then the more likely the ground levels will be the same. This is often the main cause of errors with single ended inputs. With a connection between two quite different grounds, the difference in these levels can cause large currents to flow, these are commonly known as earth or ground loops.

Additionally, single ended inputs can suffer from noise injection. Noise can be injected into signals because the wire that carries the signals can act as an aerial and thus pick up all manner of electrical background noise. Once this noise has been introduced into the signal this way there is no way to remove it.

Unlike single ended amplifiers, differential amplifiers amplify the voltage difference between two inputs. A single ended amplifier is shown in Figure 3 and a differential amplifier in Figure 4.

Both these types of amplifier are powered in the same way, but the differential amplifier, amplifies the difference between its two inputs, whereas the single ended amplifier, amplifies the difference between its single input and ground.

The differential signal, and the cabling method used, eliminates common mode noise. This is unwanted noise that is injected into both wires, but because only the difference between the inputs is measured it is ignored. This allows for much longer cables without increasing unwanted noise in the measured signal.

However, a common problem when using differential inputs is neglecting any possible connection to ground. For example, in the case of battery powered equipment, the battery and circuits might produce signals that are near the limit of the amplifier. If a ground is introduced into the circuit the levels

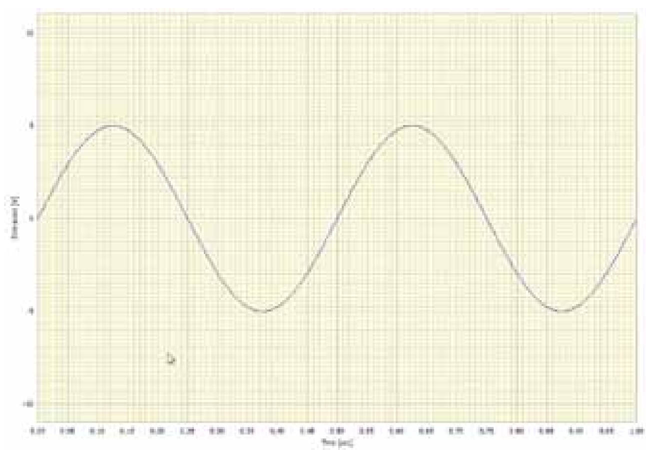


Figure 1: +/- 5 volt sine wave

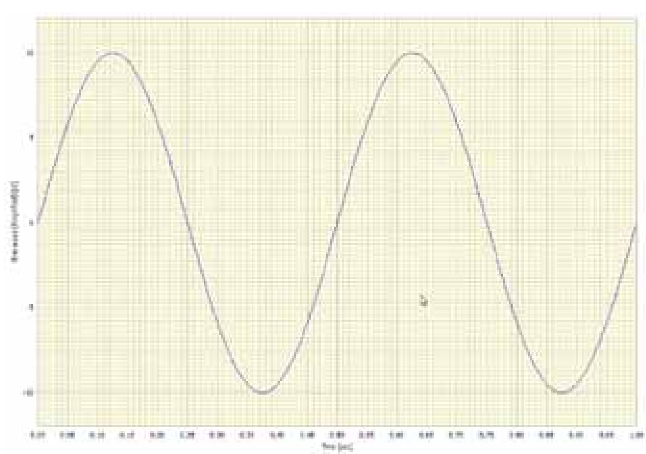


Figure 2: Sine wave amplified by factor of 2

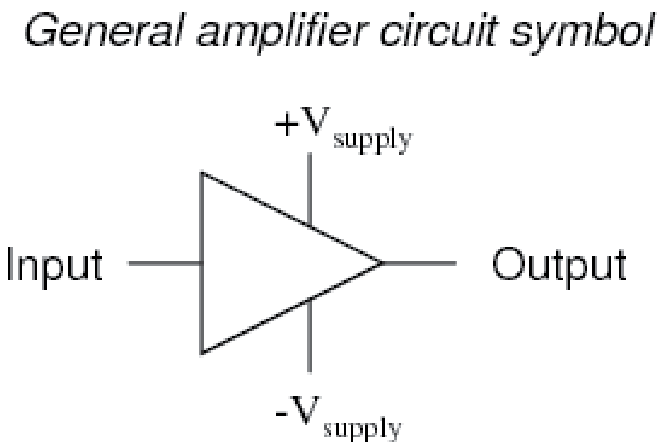


Figure 3: Single ended amplifier

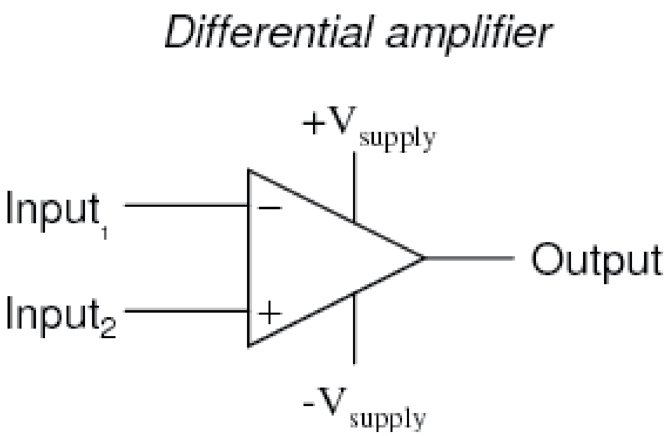


Figure 4: Differential amplifier

could be significantly different to that of the battery voltages. Furthermore, when you consider all three of these voltages, ground, battery – and battery +, the levels could be outside the operational range of the amplifier. These situations are known commonly as floating signals. The signals are not referenced to ground. In these cases it is required to introduce a ground to provide this reference. For this situation the instrumentation, in this case a DATS-tetrad, has a GND connection. A wire would be inserted here and connected to the electrical ground. Thus the differential signals are measured against each other with reference to ground.

In summary, differential inputs reduce noise and allow for potentially longer cabling. They can be short circuited to be used as single ended inputs if required. Differential inputs can be used for floating signals, but in such cases a reference should be provided to the instrumentation.

What is the Difference Between Microphone Types?

James Wren

Learn about the difference between free field, diffuse and pressure microphones

We are often asked what is the difference between free field microphones, diffuse field microphones and pressure microphones.

For a run-of-the-mill ½ inch microphone the short answer is nothing.

However, the long answer is a bit more involved.

Basically, if the frequency range is below 16 kHz and an accuracy of ±2dB is acceptable then there is no real difference between the types.

So, what is the difference between these types of microphone?

Well, first you have to understand the different types of sound field involved.

In theory a free field is a sound field where the sound waves are free to expand outwards forever from the source. That is, we assume that there are no reflections or reverberations. In practice, we would consider an anechoic chamber to be a free field or an outdoor measurement, provided we measured at a sufficient distance from the ground.

A diffuse field or a random incidence field, is a sound field where the sound waves arrive equally from all directions. Another way to think of a diffuse field is that you have sounds coming from different directions in succession with no time in between their arrival. For example, in a diffuse field the sound waves that arrive at a person's ears are so completely different that the brain finds it impossible to work out where the sound came from.

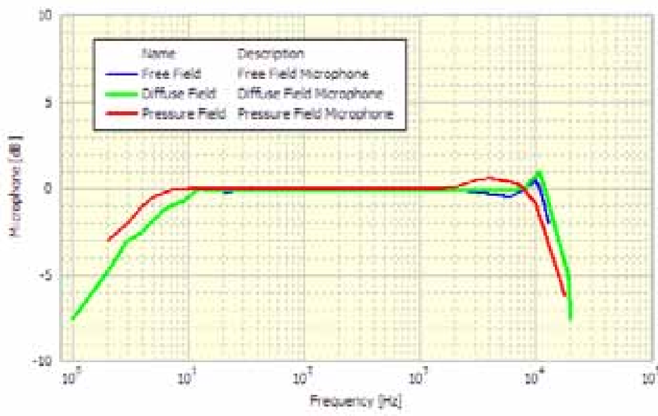
A pressure field is a sound field where the sound pressure has the same magnitude and phase at any position in the sound field.

It is important to understand that all of these microphone types are fundamentally the same. They are transducers that are designed to sense pressure levels in air. The differences between them are in the designs of the microphone heads. Each type of microphone will normally be supplied with a calibration data sheet that will show the frequency response against sound pressure. Comparing the frequency response against sound pressure for the three microphone types will show the differences clearly.

The differences between the three types of microphone generally occur at higher frequencies as we stated previously, below about 16 kHz the response from each will be similar. This is partly to do with the physical size of the microphone with respect to the wavelength of the sound being measured.

Free field microphones are designed to compensate for the effect of the microphone itself in the field. So you measure the sound as though the microphone was not there. They are designed this way as, of course, the presence of the microphone will affect the propagation of the sound wave. A free field microphone should be pointed towards the sound source at a 0° angle of incidence.

Diffuse field microphones are designed to respond in a uniform manner to any signal arriving on its measuring surface



from any angle. Generally, these tend to be oriented at about 75° to the direction of the sound wave propagation in a free field.

Pressure microphones are designed to respond to a uniform frequency response to the sound level itself. When used in a free field a pressure microphone should be mounted at 90° to the direction of the sound wave propagation, effectively the sound passes the measuring face of the microphone.

Avoiding Ground Loops

It's not only a problem with accelerometers

James Wren

When using modern, high technology measurement devices one can often be tripped up by the simplest things. The most common is the ground loop. Time and again this issue rears its head. So, let's talk about how to avoid a ground loop.

What is a ground loop?

Most sensors and their cases will be made from metal. Usually the mechanical structures under test are made of metal. These metals are normally ferrous and, almost always, will conduct electricity.

Generally, accelerometers are of the non-isolated type. That is, the case of the accelerometer is electrically connected to the ground of the cable that carries the sensor signal.

This means the electrical potential of a large metal object being tested could be at a different potential to the measurement system. Perhaps the measurement system is running from mains power or a battery. The result is that a current is induced in the cabling between the measurement system and the metal test object. Thus the cables carrying the signals are corrupted by the current flows. The only solution is to correct the issue and stop the current flow. Another 'easy' solution will not have the desired effect.

How to avoid a ground loop

Normally, one would ensure that the measurement system and any computers using the measurement system are powered from the same source and so they will be at the same potential. Even this is not as easy as it sounds. Low cost power supply units often omit the common ground path from AC/DC input to DC output.

Good quality measurement systems will always have a ground connection. One could attach an accelerometer to an engine block and then ground the measurement system to the same ground as the block. This would be the vehicle 0V. Then, if there is an issue, the ground cable can make the connection and so any current flowing will not adversely affect the signal path. The 0V should be the same 0V that is powering the measurement system and any laptops.

It is always best to ensure that the piece of metal under test is itself grounded.

Are there any other considerations?

What is the catch?

Sometimes the above will not help, but running the laptop from its internal battery rather than from the common ground will. Technically this should not help or even work.

However, experience has shown that although we don't always understand what is happening, sometimes it makes the difference. That is why we call it the black art of ground loops!

What can be done to help this issue?

It is possible to apply electrical tape between the base of the accelerometers and the metal they are physically connected to. This will isolate the accelerometers and they will still respond in a similar fashion, but it is likely that the tape will change the frequency response of the accelerometer. Using a product like Blu Tack would mean the accelerometers may not respond as expected, as it will act like a spring. Also it will melt easily.

Petro wax is the most commonly used method for affixing accelerometers. Unfortunately, that method does not ensure that there are no electrical connections between the accelerometer and the test piece.

An acceptable solution is to purchase isolated accelerometers or isolation bases for the accelerometers.

What about impact hammers?

When using impact hammers a metal tip is often required to excite at the desired frequency range. These hammers are usually not an isolated design and so when an impact is made the signal can become corrupted. In these cases, it is not possible to apply a material (e.g. tape) to the tip or use a non-metallic tip. This would adversely affect the excitation frequency range. The only solution is to ensure the test piece is properly grounded.

My equipment is connected to the mains power socket, so surely it's grounded?

Well, not really. It is likely that the earth connector in the wall socket is not actually connected to anything and therefore the potential of the supply is actually floating relative to another supply. In my experience this is a big problem in the Far East. If that is the case there is no choice, but to ground the equipment directly. Usually, with a large pole buried into the ground.

Calibrating a Hammer

John Mathey

A no-nonsense look at calibrating an instrumented impact hammer

The intention of this article is not to present any new scientific theories or ground breaking techniques, but to introduce new users to hammer impact testing and remind seasoned technicians of a method of how to calibrate an impact hammer and/or verify the calibration of an instrumented impact hammer (excitation transducer) and an accelerometer (response transducer) pair.

The reader's imagination can expand on these ideas to use other excitation techniques to calibrate excitation/response transducers as their needs dictate.

Everyone involved in vibration testing, and indeed every good science student, is aware of Newton's Second Law of motion…

Force = Mass x Acceleration

When performing any testing to measure the frequency response function (FRF) of a structure, this is the basis of all modal testing. Essentially the FRF is the implementation of the second law of motion and is typically expressed as response/excitation. There are several formulations of this which include

- Accelerance (or Inertance) = response acceleration per unit force (A/F)
- Mobility = response velocity per unit force (V/F)
- Compliance (or Receptance) = response displacement per unit force (D/F)

There are other representations, but typically for modal hammer impact calculations the Accelerance (A/F) is used.

What is acceleration divided by force (A/F) equal to? It is the reciprocal of the dynamic mass of the structure between the

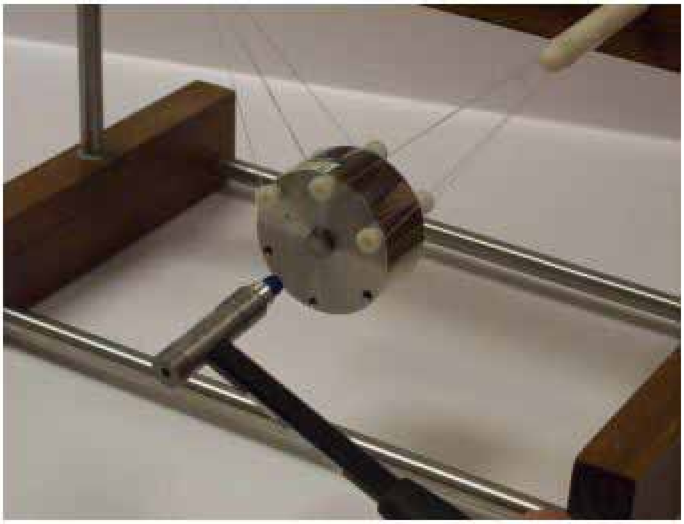


Figure 1: The Modal Shop models 9155D-961 mass calibration system

measured points. If a solid mass structure (block of metal) is impacted at the same point where the response is measured (driving point measurement), any resonances of the structure will typically be higher in frequency than those of interest of most structures. So, below the first structural mode of the test specimen the value of the FRF will be "flat" over a wide frequency range and will be equivalent to the reciprocal of the actual mass of the specimen. There will also be "low" frequency resonances which are the rigid body modes of

Figure 2: Improvised system to calibrate an impact hammer

the structure which will be dependent on how the mass is supported or suspended.

The Modal Shop has a product (model 9155D – 961) that uses this theory to calibrate an impact hammer with a known mass and a calibrated accelerometer (see Figure 1). However, even if the actual sensitivities of the accelerometer and the hammer are not known, the ratio of the sensitivities of these two transducers can be determined to give the actual mass of the block of metal.

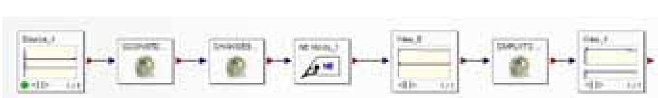


Figure 3: DATS Worksheet to produce results in magnitude and phase

The elaborate suspension system of this known mass minimizes rigid body modes of this suspended mass.

I did not have access to this fancy system, but I wanted to check the sensitivities of my impact hammer and response accelerometer. I performed a very simple test on a known mass of 57 grams I had in the laboratory using an impact hammer (nominal 2.5 mV/N) and an accelerometer (nominal 1 mV/m/sec^2). By placing the response accelerometer on one end of the mass, and impacting on the other side of the mass, Since the mass is a solid piece of steel any structural resonances will be well above the frequency range of interest for this calibration consequently there will be no anti-nodes between the excitation and response points. This will essentially be a driving point measurement.

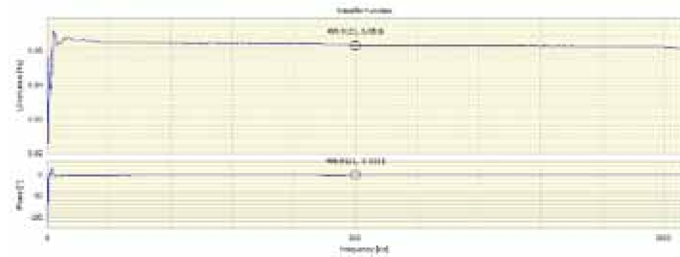


Figure 4: 1/Inertance (m/sec^2/N = kg) for known 0.057 kg mass

The FRF results in DATS software are saved in Real-Imaginary format, consequently I wrote a simple DATS worksheet to take the reciprocal of the saved FRF, change the units from N/m/sec^2 to kg (since 1 newton = 1 kg*m/sec^2), and convert the data format to Magnitude-Phase. This worksheet is shown in Figure 3.

The results I obtained confirmed both the hammer force transducer and the accelerometer were in good working order and the nominal sensitivities of the transducers were quite close.

For more accurate results, the mass of the accelerometer and cable plus the mass of the hammer tip should be taken into consideration. Any changes to the measurement system such as changing the mass of the impact hammer would suggest this calibration test be re-confirmed.

So, in conclusion, for the purposes of my testing, I was pleased with the results and I was able to continue customer tests with confidence.

Accessories

Signal input adapters

02-33-1164	SMA-to-BNC adapter (for DATS-solo)
02-33-0710	6-pin Lemo® to bare-ended cable
02-33-0874	4-pin Lemo® to bare-ended cable
02-33-1014	7-pin Lemo® to bare-ended cable
02-33-0711	6-pin Lemo® to BNC cable
02-33-0954	4-pin Lemo® to BNC cable
02-33-1015	7-pin Lemo® to BNC cable
02-33-0955	Multi-pin to BNC adapter for use with 8-channel card *
02-33-0956	Multi-way to Lemo® socket for use with 8-channel card **
02-33-1039	Multi-way to BNC socket for use with 8-channel card **

* For use with cards 03-33-8412 & 03-33-8512, 4 units required per 8 channel card
** For use with cards 03-33-8414 & 03-33-8514

Carry cases

04-66-1143	Replacement DATS-solo carry case
04-34-1144	Replacement DATS-tetrad carry case
04-34-1145	DATS-tetrad ExoCase
04-34-1146	DATS-tetrad protection sleeve

Power supplies, batteries & cables

02-34-1137	Mains power supply for DATS-tetrad
02-33-0867	Mains power supply for DATS-hyper12
02-34-1138	In-vehicle power cable for DATS-tetrad
02-33-0866	In-vehicle power cable for DATS-hyper12
06-34-1139	Battery replacement service for DATS-tetrad

Communication/interconnection cables

02-66-1135	PC to DATS-solo USB comms cable
02-33-0852	PC to DATS-tetrad/hyper12 USB comms cable
02-34-1136	PC to DATS-tetrad Ethernet comms cable
02-33-0855	DATS-tetrad interconnection cable
02-33-0873	DATS-hyper12 interconnection cable
02-66-1141	DATS-solo/DATS-tetrad ground cable
02-33-1142	DATS-hyper12 ground cable
02-33-1165	GPS aerial

DATS-tetrad EXOcase (04-34-1145)

- **Custom storage for the DATS-tetrad**
- **Scratch & water resistant - IP56 rated**
- **MIL-SPEC 81-41 approved**
- **6x stronger than a traditional flight case**
- **Operate the system while it's in the case**

The DATS-tetrad ExoCase ensures the highest levels of protection for the system during both transportation and in-field use.

The bespoke design not only allows you to operate the system while in the case, but also provides ample storage for all of your accessories.

Having everything in a single, easily-accessible place can significantly reduce setup time and also prevents damage or loss to any components when they are not in use.

DATS-tetrad Softshell Sleeve (04-34-1146)

- **Protect from knocks, dents & scratches**
- **4mm die-cut precision Neoprene**
- **Chemical resistant ***
- **Water resistant ***

The DATS-tetrad softshell provides basic protection for the system while in operation.

The precision die-cut sleeve keeps your system safe from knocks, dents and scratches, while allowing access to all of the system's controls and connectors.

Custom fitted, double-lined Neoprene means there's no fuss with a zip or any complicated fastening, just pull the sleeve over the DATS-tetrad and it will securely hold itself in place.

** Applies to covered surfaces only*

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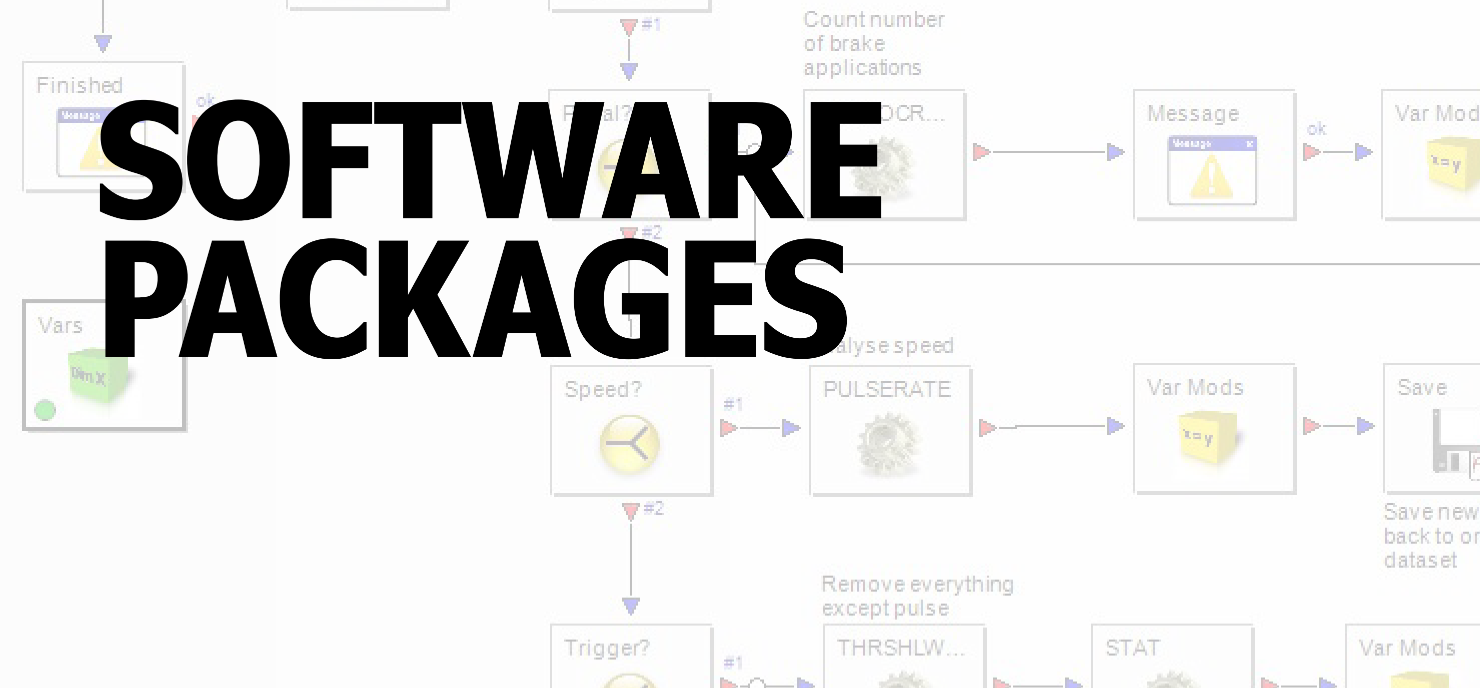
SOFTWARE PACKAGES



DATS-toolbox

- Data Capture / Acquisition ✓
- 2D & 3D Visualizations ✓
- Data Import & Export ✓
- Icon Programming ✓
- Frequency Spectra ✓
- Data Arithmetic ✓
- Time Domain Analysis ✓
- Built-in Report Generator ✓
- Digital Filtering ✓
- Curve Fitting ✓
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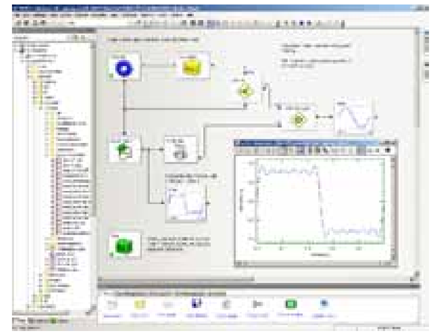
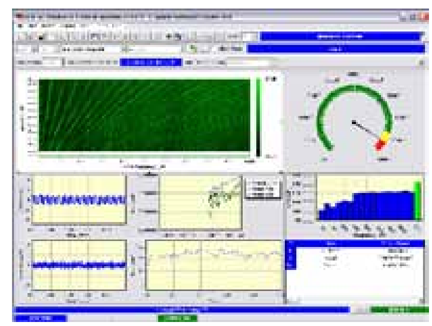
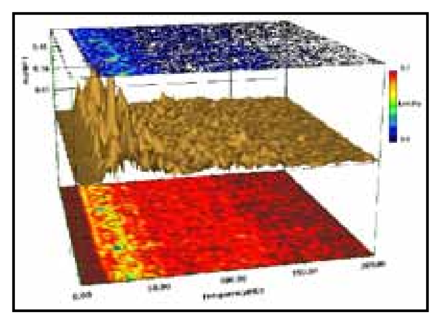
A comprehensive set of tools for the acoustics and vibration engineer



DATS is a comprehensive package of data capture and signal processing tools. DATS offers outstanding value both in cost and productivity gains. Many man-years of signal processing expertise have been spent on DATS during its 30 or more years of development. When you purchase DATS you are purchasing a share in our knowledge. We understand the requirements of our customers. DATS software has proved itself time and time again in diverse and demanding applications around the world.



Because Prosig manufactures both, the DATS software is fully integrated with the DATS-solo, DATS-tetrad and DATS-hyper12 hardware. The data acquisition software in DATS contains everything needed to view, calibrate, monitor, and store your data. As well as capturing data using the Solo, Tetrad or Hyper12, DATS can be used on data from a huge range of sources using its unique import and export filters.



DATS-toolbox software has an unparalleled depth of signal processing functions. A full list of analysis functions can be seen on the page opposite.

The DATS environment offers a rich selection of different graph styles to view and explore your data. 2D styles include lines, bars, symbols, X v Y graphs, bode plots, polar plots, modulus & phase and so on. For more complex applications there are 3D styles such as isometric, colored surface, waterfall, contour, color maps, intensity plots etc.

All of the capture, import/export, analysis and graph functions can be easily combined using the DATS Visual Scripting interface. Functions may be configured as necessary using a simple icon based interface. Visual Scripts can contain complex structures such as loops and if conditions as well as using forms and producing reports.

For programmers, DATS comes with a built in BASIC scripting language. This can be used to automate any part of the DATS processing mechanism and build complete applications. Scripts can contain data capture, data import, analyses, user input forms, graphical results, report generation and data export. DATS BASIC Scripts also support OLE automation for seamless integration with other products such as Microsoft Office.

The Intaglio Report Generator makes it easy to produce top quality reports time after time. It uses the powerful OLE technology built into Windows to add DATS graphs and other related information (numbers, labels and text) to standard Microsoft Word documents. Intaglio uses a system of templates that, once created, can be used over and over again.

DATS-toolbox comes with all of the functionality mentioned above. The following pages contain details of the add-on software options that are available.

Analysis functions included in the DATS-toolbox package

Arithmetic (Data & Data)
Data & Data Arithmetic (+ - * /)
Data + Ind. Var. Arithmetic (+ - * /)
Data and Reference

Arithmetic (Data & Constant)
Real Data & Constant (+ - * /)
Complex Data & Constant (+ - * /)
Mod/Phase Data & Constant (+ - * /)

Calculus
Differentiate
Integrate
Integrate X with Y
Omega Arithmetic
RMS over Frequency Band
Complex Functions
Complex to & from Mod/Phase
Complex to Real
Complex to Imaginary
Complex Output (Imag+iReal, Real+i0, Imag+i0, Real+iZ)
Data & Conjugate(Data) (+ - * /)

Curve Fitting
Alpha-Beta Smooth
Fit Stepped Data
Lagrange
Least Squares Polynomial
Mean Median Despike
Remove Spikes from Data
Smooth
Spline Fit

Data Acquisition
Spreadsheet Style Setup
Multi-channel realtime displays of numeric values, time histories, FFT, spectrum waterfalls, orders
Setup Information Stored with Data
Multi-channel range display
Dynamic/Static Signal Calibration Tools
Automatic Gain Ranging
Over-range indications
Automatic Increment of filenames

Event Processing
Extract Event
Mark Event
Remove Event

Export Data
ASCII
Comma Separated Variables (CSV)
SDF (HP/Agilent)
Matlab
MS Excel
RPC III
TecPlot
Universal File (UFF)
WAV

Filtering
Alpha Beta Filter
Bessel (Low, high & band pass & band stop)
Butterworth (Low, high & band pass & band stop)
Chebyshev (Low, high & band pass & band stop)
Equalisation Filter
Filter Octave (Band Pass)
Frequency Characteristics (Butterworth, Chebyshev & Bessel)
Impulse Response Filter
Notch
RC Filter
Shelving Filter
Smoothing

Frequency Analysis
Auto (Power) Spectrum
Auto (Power) Spectrum (Limit Hold)
Auto (Power) Spectrum (Hopping)
Cepstrum
Coherence Spectrum
Complex to Mod/Phase
Cross Spectrum
Cross Spectrum (Limit Hold)
dB Weighting
DFT
DFT (Goertzel)
Weighting (A,B,C,D)
FFT (Full Range)
FFT (Half Range)
Hopping FFT
Inverse FFT (Full / Half Range)
Inverse FFT (Long Complex Full Range)
Omega Arithmetic
Third Octave Bands
RMS Over Frequency Band
Autoregressive Filter Coefficients
Envelope (Complex Demodulation)
Envelope (Fourier)
Long FFT
Instantaneous Frequency
Interpolate Signal
Minimum Phase Spectrum
Maximum Entropy Autoregressive
Maximum Entropy Spectral Estimate
Short Time FFT
Spectrum Level
Spectrum Level (Limit Hold & Hopping)
Transfer Function
Winograd Transform
Zoom FFT
Zoom Auto Spectral Density
Zoom Cross Spectral Density

Generate Data
Sine (Sine, Damped, Linear & Log Sweep, Modulated & Pulsed)
Random (Autoregressive, Gaussian, Rectangular, Narrow Band, Pink & Red Noise, Rayleigh Random Numbers)
Impulse
Square (Pulse & Swept)
Step
Triangle
Saw Tooth
Exponential Decay
Straight Lines & Ramps

Import Data
Artemis
ASCII
Binary
B & K Pulse
Comma Separated Variables (CSV)
DASYLab
DIA / DIAdem
DX3
SDF (HP/Agilent)
LabVIEW
Matlab
MS Excel
nCode
PICOLog
Realwave Pocket Analyzer
RES Data
Rion WAV
RPC II / III
Store Plex (Racal)
TEAC
Universal File (UFF)
WAV
WaveView (Iotech)

Math Functions
Absolute
Arcsin, Arccos, Arctan, Arcosec, Arcsec, Arccotan
Arcsinh, Arccosh, Arctanh, Arccosech, Arcsech, Arccoth
Antilog
Backward & Forward Difference
Conjugate
Error Function (ERF)
Exponential
Gamma Function
Inverse ERF
Linear to dB, dB to Linear
Log e, Log 10
Negate
Nth Integer Root
Raise To Power
Reciprocal
Sin, Cos, Tan, Cosec, Sec, Cot
Sinh, Cosh, Tanh, Cosech, Arcsech, Arccoth
Square Root

Pulse Analysis
Angular Vibration of Shaft
Pulses to Rate
Pulses to Amount
Pulse Duration (All Crossings, Pos-Pos, Neg-Neg, Pos-Neg, Neg-Pos)
Create Speed Signal

Shock Spectral Analysis
Primary, Residual & Composite
Linear spacing, Logarithmic spacing & Octave spacing
Shock Time History (Lin, Log, Octave Spacing)

Signal Manipulation
Add Named Elements
Amend Control Record
Append Signal to Dataset
Apply Classic Window
Apply Exponential Decay
Apply Force Window
Apply Sine/Cosine/Ramp Taper
Copy Whole Signal
Copy Section of Signal
Extract Named Elements
Include Signals to Dataset
Join Signals
Mesh Two Signals
Modify Named Elements
Repair Signal
Replace Signal
Replace Single Named Element
Reverse Signal
Signal Quality Check
Sort Signal
View Signal History

Statistics
Signal Decimation
Statistical Counting
Level Count (Number of Intervals, Size of Duration Interval, Interval Size as %age, Output All Duration, Referenced to Signal Mean, Specify Reference Level)
Mean Crossing Peak Count
Net Peak Count
Peak and Trough Count
Rainflow Counting (Cycle Peak / Trough)
Rainflow Counting (Cycle Range / Mean)

Time Domain Analysis
ADC Simulation
Apply Threshold
Auto / Cross Correlation (Lagged Products or Fourier Transform)
Bias removal
Coherence Related Time History
Convolution in the Time Domain
Cosine Taper Function
Ensemble Statistics
Evaluate Trend (Mean, SD, RMS, skew, kurtosis, M5, M6, Min, Max)
Generate Actuator Stepped Sine Wave
Generate Actuator Swept Sine Wave
Generate Break Points
Generate Spectrum
Generate Gaussian Probability Density
Generate Log Probability Density
Generate Rayleigh Probability Density
Generate Sine Probability Density
Generate Data Window
Generate Cosine Taper Window
Generate Exponential Decay Window
Generate Force Window
Random Time History from spectrum
Joint Probability Density Function
Normalize

Probability Analysis
Percentile Calculations
Probability Density Function
Signal Decimation
Signal Generation
Statistics
Time Reverse
Trend Analysis
Trend Removal (Linear Averaging Points, Exponential Averaging & Linear Averaging Duration)

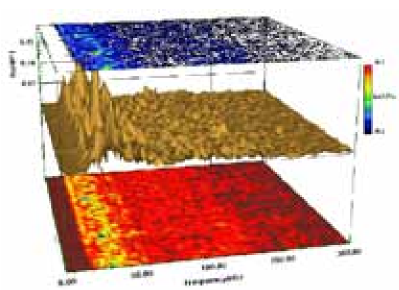
DATS-toolbox Software

01-55-622	DATS-toolbox software. Includes Intaglio reporting suite, DATS BASIC Script support and data acquisition software.

DATS NVH Analysis

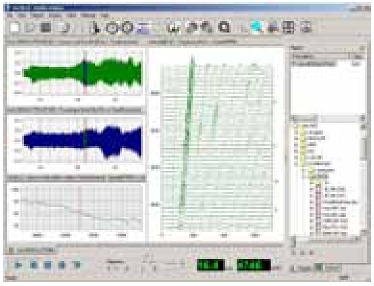
Waterfalls
Order Tracking
Sound Quality Metrics
Tacho Analysis
Time-Frequency Analysis
Sound Quality Audio Replay
Wavelet Analysis

The refinement of vehicles and rotating machines with respect to noise and vibration is central to creating a successful product. It's not just making less noise, but also making the "right" noise, that is important.



Using the depth and power of the DATS NVH software suite, it is possible to measure and refine the product.

Extensive tacho analysis is used to analyze angular speed. Waterfall and order analysis picks out those parts of the spectrum that are harmonically related. Waterfall averaging enables the engineer to get a more consistent view of the problem. 1/nth Octave Analysis is used extensively as a first level method of reducing the data into standardized bands, which reflect the human response to noise. There are also a large number of Sound Quality metrics, that can be used to quantify noises in ways that reflect more accurately the psychoacoustic response of the drivers and passengers.



The DATS Sound Quality Audio Replay (SQAR) package allows a user to listen to and analyze audio data. Multiple filtering with combinations of order filters and frequency filters allows detailed investigations and "what if?" analysis. The playlist may include both the original and the modified signals for direct comparison with each other.

In the Time-Frequency Analysis package, a number of different algorithms including Wigner Ville, Atlas Zhao Marks, and Born Jordan, all give slightly different emphasis to features of the signal.

Analysis functions included

Waterfall Analysis
Speed signal from tacho
Waterfall from tacho signal with phase
Waterfall from speed signal

Order Extraction
Frequency to order spectrum conversion
Order cuts from waterfall

Averaging, Weighting & Octaves
A, B, C, D spectral & time domain weighting
Spectrum averaging
Spectrum average & RMS in user-defined bandwidth
Waterfall averaging
1/nth octave band analysis

Sound Quality Metrics
AI Versus Time
Loudness
- Zwicker Diffuse (ISO532B)
- Zwicker Free (ISO532B)
- Zwicker Diffuse (Vehicle Biased)
- Zwicker Free (Vehicle Biased)
- Stevens (ISO532A)

Loudness Versus Time
Speech Articulation Index
- ANSI S3.5 1969
- Vehicle Biased

Composite Rating Performance Value
High Frequency Factor
Preferred Speech Interference Level
Spectral Balance

Misc
Nth Octave from Time
Difference dB Signals (in averaging weighting and octaves)
N10S10 calculation
Equalisation Order Filter

Time Frequency Analysis
Born-Jordan
Wigner Ville
Zhao Atlas Marks
Mother Wavelet Generation
Wavelet Transforms
Wavelet Reconstruction
Wavelet De-noising
Wavelet Filtering

SQAR Visualizations
Time Histories
Time-Speed Curve
Order Plots
Waterfall Plot
Waterfall Color Map
Sound Quality Metrics
Real-time Speed Readout

SQAR Filters
Order Pass
Order Reject
Butterworth Frequency (Band Pass)
Butterworth Frequency (Band Reject)
Filter Attenuation versus Speed

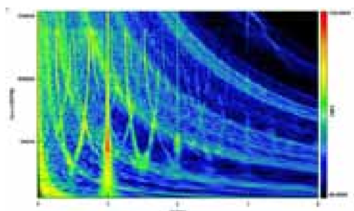
DATS NVH Analysis Suite

01-55-801	NVH analysis suite (Requires 01-55-622 DATS. toolbox)

DATS Rotating Machinery Analysis

Waterfalls & order tracking
Time sampled & angle sampled data
Special analysis for angle sampled data

The DATS Rotating Machinery option contains a complete set of tools for analyzing the sources of vibration and noise caused by cyclic forces such as those found in engines, gearboxes and wheel excitation.



Prosig acquisition software has additional real-time displays for use with Rotating Machinery Analysis.

The Time Sampled analysis enables a user to carry out classical Waterfall analysis, producing frequency spectra related to the speed of rotation. It includes comprehensive tacho conditioning software. The software allows waterfalls and orders to be visualized in many ways. Band-pass filtering and envelope analysis can be carried out for bearing analysis.

Various synchronous analyses can be used to view the data in the order domain. In particular a discrete Fourier transform (DFT) can be used to extract orders directly. Data which has been sampled using a fixed time sample rate can be resampled using the tacho as the synchronous marker, so that the same number of samples are generated for each cycle.

Analysis functions included

Time Sampled Data
Average Waterfalls
Speed Signal from Tacho
Extract Orders and Overall Level
Generate Waterfall
Generate Waterfall with phase
Equalisation Order Filter

Advanced Tacho Analysis
Angular Vibration from Tacho
Tacho Crossing times
Tacho Ideal Equivalent
Tacho to time periods
Raw Speeds
Average period Speeds
Smooth Curve Fitted Speeds
Interpolated Speeds
Tacho Crossing Checks

Synchronously Sampled Data
Angular Vibration of Shaft
Asynchronous to Synchronous Order Waterfall
Order Waterfall with Phase
Synchronous Orders
Calculate Average Cycle
Calculate Cycle Statistics
Tacho Synthesis

Order Domain Data Analysis
Auto Spectral Density
Cross Spectral Density
DFT
FFT
Multiple Spectrum RMS Level
Spectrum Level
Spectrum RMS Over Order Range
Transfer Function
Zoom Transfer
Zoom Auto Spectral Density
Zoom Cross Spectral Density

Time Frequency Analysis
Born-Jordan
Wigner Ville
Zhao Atlas Marks
Mother Wavelet Generation
Wavelet Transforms

DATS Rotating Machinery Analysis Suite	
01-55-802	Rotating Machinery analysis suite (Requires 01-55-622 DATS-toolbox)

DATS Hammer Impact Analysis

Frequency Response Functions (FRF)
Structural Response Measurements
Double Impact Detection
Accept/Reject by User
Automatic Averaging
Integrated with DATS Structural Animation

Structural Response Measurements are an essential requirement for engineers working on Noise and Vibration problems. The Hammer Impact Analysis guides the user through the process of making the measurements. Single-input, multi-output measurements are supported. Transducer gain, sensitivity and triggering setup are provided by the familiar DATS signal setup.

The software gives the user full control over all aspects of the test including...

- the frequency range and resolution
- force & exponential window settings
- number of averages per measurement point
- auto rejection of overloads
- use of Prosig remote keypad

A pre-test Wizard assists the user in setting up the trigger level, response window weighting factor, and the force window profile. The pre-test displays of time histories and FFT spectra enable the input signals to be checked before the impact tests are started.

Immediately after each test impact the latest Frequency Response Function (FRF) is displayed together with the accumulated average. A reference FRF can be superimposed if required. After each measurement the user has the option to discard the latest result and remove it from the accumulated average.

The measured Frequency Response Spectra are stored in numerically sequenced datasets. The user has control over whether the datasets to be saved are just the measured FRF's or if the coherence spectra and time histories are also required to be kept.

This package can also be used in other applications where a triggered acquisition, immediate data inspection and frequency response measurements are required.

DATS Hammer Impact Analysis Software	
01-55-627	Hammer Impact analysis software for DATS-solo/tetrad/hyper12 (Requires 01-55-622 DATS-toolbox)

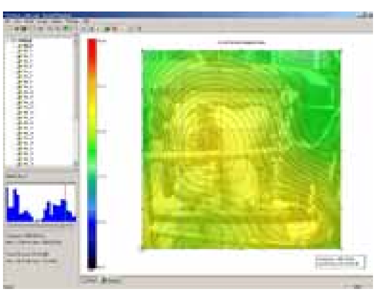
DATS Acoustic Analysis

1/N Octave Filters
Sound Power & Intensity
Sound Level Meter
Transmission Loss
Room Acoustics
Loudness, Sharpness, Roughness
Sound & Intensity Mapping

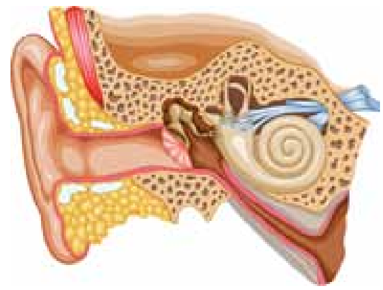
The 1/N Octave filters can be used with either time signals or narrow band spectral data.

The Sound Power Analysis scripted measurement procedure allows the measurement of sound power using either sound intensity probes or microphones according to the international standards ISO3744, ISO3745 and ISO9614-1.

The Sound Level Meter module provides a number of analyses that mimic the operation of a simple sound level meter.



The Transmission Loss modules are automated measurement and analysis procedures for determining the effectiveness of either panels for room acoustics or pipes for exhaust mufflers.



The Room Acoustics Reverberation Time T60 and Total Absorption modules use the noise source switch-off method. The T60 determination is based on a practical measurement with a decay in the room of less than 60dB.

The Two-Microphone Impedance Measurement Tube, (B&K Type 4206), is a completely scripted measurement procedure for guiding the user in making an accurate measurement of the acoustic properties of small material samples, it complies with ISO10534 and ANSI E1050.

Psychoacoustics metrics such as sharpness and roughness can be used to analyze data in way that accounts for physical and psychological effects of the ear and brain.

The DATS Sound Mapping software consists of two main facilities:

The first creates a 3-D sound map from sound pressure measurements, while the second does the same thing for sound intensity measurements. Both options display results as color or contour maps. In both cases the color or contour map can be overlaid on a picture of the test item to allow better visualization of the data.

Analysis functions included

Sound Quality Metrics
AI Versus Time
Loudness (Zwicker) (ISO532B)
Loudness by Stevens (ISO532A)
Loudness Versus Time
Nth Octave RMS Versus Time
Speech Articulation Index
- ANSI S3.5 1969
- Vehicle Biased

Composite Rating Performance Value
High Frequency Factor
Preferred Speech Interference Level
Spectral Balance

Sound Power
Sound Power from Pressure (ISO3744, ISO3745)
Sound Power from Intensity (ISO9614-1)

Sound Level Meter
rms S or F time constant
rms I time constant (peak)
rms selectable time constant
Leq with S orF time constant
Leq with I time constant (peak)
Leq with selectable time constant
LN Measure with S or F time constant
LN Measure with selectable time constant
SEL with S or F time constant
SEL with selectable time constant

Misc
Nth Octave
Nth Octave RMS Output
Nth Octave Filter
Microphone Calibration
Spectrum Weighting (A,B,C,D)
Sound Intensity & Intensity Density
Sound Power
Difference dB Signals
A, B or C Weight Time Signal
Impedance Tube Reflection
Impedance Tube Absorption
Reverberation Times
Transmission Loss through panels
Transmission Loss through pipes

Psychoacoustics
Loudness
Sharpness
Roughness
Fluctuation Strength
Tonality
Prominence Ratio

Prominence Standards
ANSI S1.13-2005
ECMA74
DIN 45681:2005

DATS Acoustic Analysis Suite

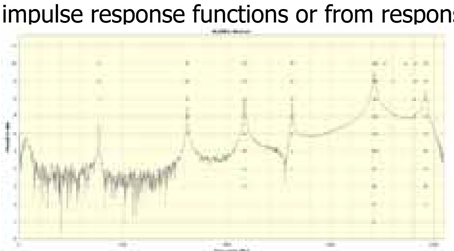
01-55-791	Acoustic analysis suite (Requires 01-55-622 DATS-toolbox)

DATS Modal Analysis

Experimental Modal Analysis (EMA)
Operational Modal Analysis (OMA)
Modal Parameter Identification
Alternative Curve Fitting Algorithms:
- SDOF
- MSDOF & MDOF (Freq. Domain)
- ERA-DC (Time Domain)

Stability Diagram
FRF Synthesis from Modal Parameters
Forced Response Prediction

The DATS Modal Analysis suite is provided for analysts who want to determine Modal Frequencies, Damping Factors & Modal Amplitudes from either measured frequency response functions, impulse response functions or from response-only data. A variety of frequency and time domain identification methods are provided for the extraction of these parameters. These include Half-Power methods, SDOF, MSDOF, MDOF and ERA-DC. The identified mode shapes can be displayed and animated using the Prosig Structural Animation package. A synthesis module is provided to enable Frequency Response Functions (FRF) to be regenerated from the identified parameters thereby revealing the accuracy of the modal model fitting. Forced responses can also be predicted by convolution of the regenerated FRFs with either simulated or known force inputs.



Features included

Parametric identification of
- Modal Frequencies
- Damping Factors
- Modal Amplitudes

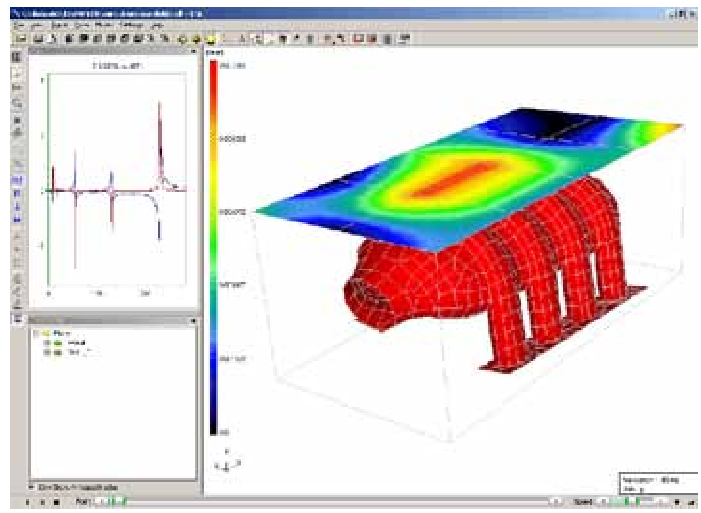
Frequency and Time Domain Methods
- Half-power estimates
- SDOF, MSDOF, MDOF
- ERA-DC

OMA Methods
- FSDD (Frequency)
- ERA-DC (Time)

FRF synthesis from Modal Parameters

DATS Structural Animation

Frequency & Time Based Animation
Operating Deflection Shapes
Full 3D Views
Sophisticated Model Editor
Built-in Hammer Acquisition Support



Frequency Animation uses the magnitude and phase of Frequency Response Functions (FRFs), FFTs, or cross spectra at each measurement position on the structure to reveal the motion at different frequencies.

Time Animation takes time based data and uses it to directly show the true position at each measurement point at each time step.

The graphical representation of the structure is achieved by setting up two- or three-dimensional space-frame models. The model may be created using a fully featured 3D graphical editor. Models can also be imported from NASTRAN, CSV and Universal files.

Features included

- Animates data in time or frequency domains
- Comes with fully featured, easy to use model editor
- Split view display with VCR/PVR style playback and navigation
- Built in interactive hammer acquisition for immediate results
- Interactive band reject and band pass filtering in the time domain
- Real time animation overlays for instant comparisons
- A wide range of displays including:
  - Stress/Vibration/Intensity color map
  - Magnitude and Divergence
  - Nodal Persistence
- Acceleration, Displacement and Velocity readouts
- Support for CAE and SAE coordinate systems
- Import geometry data from NASTRAN and generic CSV files
- Export animations to video for presentations and results sharing
- Attach all your model data with a single click
- Many examples and templates to get you going

DATS Modal Analysis Suite

01-55-848	Modal analysis suite (Requires 01-55-622 DATS-toolbox)

DATS Structural Animation Software

01-55-629	Structural Animation with Frequency Domain Animation and Time Domain Animation (Requires 01-55-622 DATS-toolbox)

DATS Human Response Biodynamics Suite

ISO2631 Weighting
ISO8041 Weighting
Vibration Dose Values
SEAT Testing
Vehicle Crash Biomechanics

The way in which we respond to vibration from tools, vehicles and machines affects the quality of our lives, and ultimately our health. The detrimental effect of vibration of the human body has been the subject of considerable research. The understanding of this subject has now advanced the knowledge of acceptable frequency limits for vibration exposure. The weighting filters for whole-body vibration, affecting vibrational and ride comfort, and those for the exposure to hand-arm vibration are included in the DATS Human Biodynamics package. Dose values can be calculated to ensure the acceptability of the product in an environment where the customer is ever more aware of comfort requirements, and the dangers of exposure to environmental effects.

Also included are functions to analyze vehicle crash data with special emphasis on data from dummies. Analyses include the Head Injury Criteria, FIR100 filtering and CFC filters. All the modules comply with the relevant SAEJ211 and NHTSA requirements.

The suite also includes analyses necessary for S.E.A.T. compliance testing.

Analysis functions included

Human Vibration
Building Vibration Assessment Weighting
DIN45669 Building Vibration Exposure
ISO2631 Vibration Effects (pts 1,4 & 5)
ISO5349 Hand Arm
ISO8041 Hand Arm Weighting
ISO8041 Body-X,Y,Z & Combined Weighting
ISO8041 Motion Sickness Weighting
ISO8041 No Weighting
Max Transient Vibration Value (MTVV)
Motion Sickness Dose Value (MSDV)
Root Mean Quad (RMQ) Measure
Vibration Dose Value (VDV & eVDV)
Vibration Quality Measures
Vibration (BS6841) Weighting
Vibration (ISO2631) Weighting
Vibration (ISO8041) Weighting
Vibration Quality Measures with Time
ISO6954 Ship Vibration (Habitability)
Short term Vibration Quality Measures

SEAT Testing
ISO Time History from EM Spectrum
ISO 10326 Excitation Limits
Generate EM Spectrum
Class ‘n’ (ISO 10326)
New Class
Gaussian Probability Density
Auto (Power) Spectrum
Probability Density Distribution
S.E.A.T Factor (ISO)
Corrected S.E.A.T. RMS (EEC)
Average S.E.A.T Factor (ISO)
Average Corrected S.E.A.T. RMS (EEC)
Compression in Time Domain

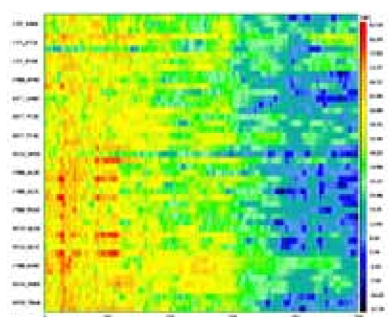
Crash Biomechanics
SAEJ211 Filter (CFC60, 180, 600 & 1000)
FIR100 Filter
Check Maximum Value
Chest Severity Index
Deflection of Dummy Ribs
Exceedance Duration
Head Injury Criterion (HIC)
Thoracic Trauma Index (TTI)
Viscus Criterion (VC)
Calculate x,y,z Resultant
Remove Signal Bias

DATS Human Response Biodynamics Suite	
01-55-803	Human Response Biodynamics analysis suite (Requires 01-55-622 DATS-toolbox)

DATS Source Path Analysis Suite

Transfer Path Analysis
Source Contribution Analysis
Path Contribution Ranking

The interior noise & vibration in a vehicle compartment is caused by various contributing exterior sources - primarily suspension and engine vibration. This raises two fundamental questions: “Which sources cause the most audible or tactile interior response?” and “Which paths are the most critical in transferring energy from the sources to the vehicle interior?” Transfer Path Analysis (also known as Noise Path Analysis or Source-Receiver Path Analysis) attempts to answer these questions by relating the vibrations measured at different locations around the vehicle to the sounds and vibrations measured inside the vehicle.



The first stage of experimental Transfer Path Analysis is the computation of the Principal Components of the system using Singular Value Decomposition (SVD). The SVD computation produces a transformation (eigenvector) matrix that is used to derive virtual cross spectra between the virtual (vibration) references and the measured (sound/vibration) responses. These virtual cross spectra are then used to calculate Reference Related Auto (RRA) spectra at every response position. Each RRA spectrum is related to just the coherent contributions from a particular reference source input.

Full Transfer Path Analysis requires not only data at the Source and Response locations, but also frequency response (FRF) functions referenced to the attachment points of the vibration isolators (anti-vibration mountings). The software estimates the dynamic forces present at the isolators and determines the contribution from each location as perceived at the (driver) response positions. The various contributions from the paths are ranked according to their severity at different frequencies or speeds.

DATS Source Path Analysis Suite	
01-55-799	Source Contribution Analysis software (Requires 01-55-622 DATS-toolbox)
01-55-872	Transfer Path Analysis software (includes Source Contribution Analysis) (Requires 01-55-622 DATS-toolbox)

Why is the microphone pressure reference $2*10^{-5}$ Pascals?

Dr Colin Mercer

This seemingly simple question is actually quite fundamental

This seemingly simple question is actually quite fundamental. To answer the question we need to consider sound intensity. Now sound intensity is defined as "the average rate of flow of energy through a unit area normal to the direction of wave propagation". The average rate of flow of energy is energy per second which we recognise as power in Watts. Intensity then has units of Watts per square metre, W/m^2, as it has dimensions of power transmitted per unit area. We could also define in old fashioned units like $erg/cm^2 sec$ and so on, but it is better to stay with Watts. Just in case you want a modern reference note that 1 Joule $= 10^7$ erg

Now the reference for Watts in air was defined as 10^{-12}W, which at the time was considered as the intensity of a just barely audible 1kHz tone to a normal human ear. This seems a reasonable choice as we are usually dealing with sound which is definitely audible, and often annoyingly so.

Moving on if we consider either plane or spherical acoustic waves in perfect free field conditions, that is a situation where no reflections occur, then it is straight forward to show that the intensity is given by

$$I = \frac{p^2}{2\rho c}$$

where p is the peak pressure of the sound wave, ρ is the air density and c is the speed of sound in air. Now for a sine wave

$$\frac{p^2}{2} = p_{rms}^2$$

so we have

$$I = \frac{p_{rms}^2}{\rho c}$$

If we take the dB level we have

$$dB_I = 10log\left(\frac{p_{rms}^2/\rho c}{10^{-12}}\right)$$

Now in air ρc is approximately 400 and we have $10^{-12} = (2*10^{-5})^2/400$

So we can write

$$dB_I = 10log\left(\frac{p_{rms}^2/\rho c}{10^{-12}}\right) = 10log\left(\frac{p_{rms}}{2*10^{-5}}\right)^2 = 20log\left(\frac{p_{rms}}{2*10^{-5}}\right)$$

That is by using 2×10^{-5} as the reference we are relating the rms pressure to the barely audible intensity in a free field, or conversely a pressure of 2 x 10-5 is the sound level we can just hear at 1kHz

In normal measurements we are not in perfect free field conditions, so the dB level of the pressure is referred to as the Sound Pressure Level, namely

$$dB_{SPL} = 20log\left(\frac{p_{rms}}{2*10^{-5}}\right)$$

If the rms pressure used is over the entire frequency range then we have the **Overall Sound Pressure Level**, often just called the **Overall Level**. If the rms pressure is the output of say a third octave filter then we call it the **dB Band Level** at that third octave centre frequency.

Standard Octave Bands

Dr Colin Mercer

What are the "standard" octave bands and where did they come from?

The "standard" centre frequencies for 1/3 octave bands are based upon the Preferred Numbers. These date from the 19th century when Col. Charles Renard (1849–1905) was given the job of improving captive balloons used by the military to observe enemy positions. This work resulted in what are now known as Renard numbers. Preferred Numbers were standardised in 1965 in British Standard BS2045:1965 Preferred Numbers and in ISO and ANSI versions in 1973. Preferred numbers are not specific to third octave bands. They have been used in wide range of applications including capacitors & resistors, construction industry and retail packaging.

In BS2045 these preferred numbers are called the R5, R10, R20, R40 and R80 series. The relationship is

Preferred Series No	R10	R20	R40	R80
1/N Octave	1/3	1/6	1/12	1/24
Steps/decade	10	20	40	80

The basis of audio fractional octave bands is a frequency of 1000Hz. There are two ISO and ANSI approved ways in which the exact centre frequencies may be defined. One scheme is the base 2 method where the ratio between 2 exact centre frequencies is given by 2^(1/N) with N as 3 for 1/3 octaves and so on. The other method is the base 10 method where the ratio is given by 10^(3/[10N]). This ratio may also be written as 2^(3/[10Nlog2]). For nearly all practical purposes both ratios are the same but tones at band edges can be interesting and may appear to be in different bands. The base 2 one is simpler to use (and is often favoured by non-engineering programmers!), but the base 10 one is actually numerically sounder.

One very good reason for using base 10 is that all the exact centre frequencies are the same for each decade. This is not the case for the base 2 frequencies.

As an example (using base 2) the theoretical centre frequency of the 1/3 octave below 1000 is found by dividing by 2^(1/3). This is 793.7005... . Using base 10 the corresponding centre frequency is 794.3282... . In both cases the nearest preferred frequency is 800Hz so that is what the band is called. When working out the edge band frequencies for a 1/3 octave then these are respectively

upper = centre * $2^{1/6}$

lower = centre / $2^{1/6}$

where the centre frequency is the exact one not the preferred one. For (1/N)th octave the relationship is simply

upper = centre * $2^{1/2N}$

lower = centre / $2^{1/2N}$

If we use the base 2 method and find the centre frequency of the third octave 10 steps below 1000Hz we get 99.21257... Hz, but with base 10 we get exactly 100.0Hz. If we continue further down to 10Hz and 1Hz then the base 2 centre frequencies are 9.84313...Hz and 0.97656...Hz respectively. The base 10 values are at 10Hz and 1Hz of course. The point to notice is that these low centre frequencies now differ by approximately (1/24)th of an octave between the two methods.

1.00	**1.60**	**2.50**	**4.00**	**6.30**
1.03	1.65	2.58	4.12	6.50
1.06	1.70	2.65	4.25	6.70
1.09	1.75	2.72	4.37	6.90
1.12	1.80	2.80	4.50	7.10
1.15	1.85	2.90	4.62	7.30
1.18	1.90	3.00	4.75	7.50
1.22	1.95	3.07	4.87	7.75
1.25	**2.00**	**3.15**	**5.00**	**8.00**
1.28	2.06	3.25	5.15	8.25
1.32	2.12	3.35	5.30	8.50
1.36	2.18	3.45	5.45	8.75
1.40	2.24	3.55	5.60	9.00
1.45	2.30	3.65	5.80	9.25
1.50	2.36	3.75	6.00	9.50
1.55	2.43	3.87	6.15	9.75

R80 Table - Preferred Values 1Hz to 10Hz, 1/24th Octave

Generally in audio work we are not too concerned about the very low frequencies. It does explain, however, why the standards use the 1kHz rather than the logical 1Hz as the reference centre frequency. If the 1Hz was used as the reference centre frequency then there would be serious discrepancies between the two schemes at 1kHz, which is very important acoustically. It is also interesting to note that third octave band numbering does use 1Hz as the reference point. We have 1Hz = 10^0 is third octave band 0, 10Hz = 10^1 is band 10, 100Hz = 10^2 is band 20, 1000Hz = 10^3 is band 30 and so on.

The R80 table above gives the 1/24th octave preferred frequencies. For 1/12th skip one to get 1.0, 1.06, 1.12 etc. For 1/6 skip three to give 1.0, 1.12, etc. For 1/3 then skip seven to get 1.0, 1.25 and so on.

Interpretation of the Articulation Index

Dr Colin Mercer

Find out about the different flavors of AI and how they differ

The Articulation Index (or AI) gives a measure of the intelligibility of hearing speech in a given noise environment. The metric was originally developed in 1949 in order to give a single value that categorized the speech intelligibility of a communication system. The basic interpretation of the AI value is the higher the value then the easier it is to hear the spoken word. The AI value is expressed either as a factor in the range zero to unity or as a percentage.

The basic method of evaluating AI uses the concept of an 'idealized speech spectrum' and the third octave spectrum levels of the background noise. Essentially if a particular background noise third octave spectrum level is above the corresponding idealized spectrum level then the contribution to AI is zero. If however the difference is positive then it will make a contribution. However if the difference is greater than 30dB then the contribution is 30dB. Each contribution is multiplied by a weighting factor specific to the particular third octave band. The sum of all the contributions is the AI value. This may be expressed as shown below.

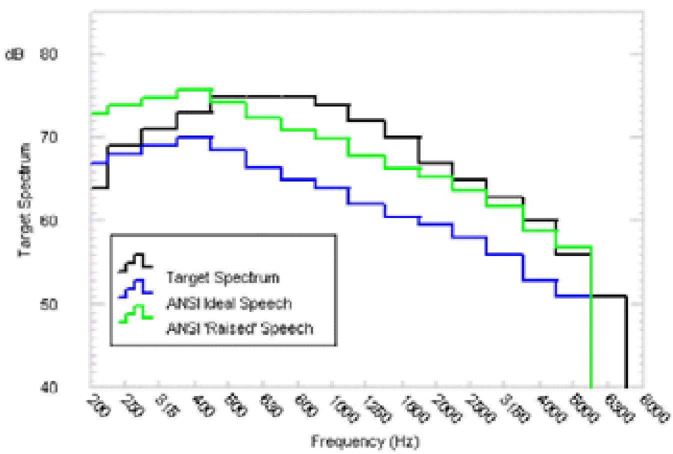


Figure 1: Standard example AI noise spectrum

Contribution = IdealisedSpectrumdB[k] – NoiseLeveldB[k]

If (Contribution < 0.0) Contribution = 0.0

If (Contribution > 30.0) Contribution = 30.0

Contribution = Contribution * WeightingFactor[k]

The contribution is found for each third octave band in the specified frequency range and summed to give the AI value.

There is however some confusion as there are three separate approaches for calculating the AI value. One method is the strict ANSI S3.5-1969 scheme, another one is generally known as the vehicle AI value and the third one as the Room AI value. We distinguish between these as $AI_ANSI, $AI_Veh and $AI_Room. The ANSI method uses third octaves in the bands 200Hz to 5kHz whilst the Vehicle and Room versions add the 6.3kHz band as well. The fundamental difference in the calculations is that the ANSI scheme attempts to take account of the existing overall noise level to adjust the levels of the Idealized Spectrum. The idea here is that if the background noise level changes then we speak either louder or softer as appropriate. That is it is strictly concerned with speech intelligibility and is not as concerned with the volume or loudness required. The vehicle and room versions of the AI are concerned with assessing sound quality in the interior environment of the vehicle or room. Thus they use what may be described as a fixed target speech spectrum. In consequence the overall level as well as the spectrum shape affect the metric. By convention the $AI_ANSI and $AI_Room values are usually given as an index from zero to unity but the $AI_Veh is usually given as a percentage. The $AI_Veh and $AI_Room give quite similar values. Figure 1 above shows the ANSI Ideal Speech spectrum, the fixed 'target' spectrum for $AI_Veh and a raised version of the ANSI spectrum whose overall matches that of the vehicle target spectrum.

The differences in the two principal spectra are obvious. However by comparing the ANSI 'raised' spectrum to the vehicle 'target' spectrum, it is clear that the vehicle target spectrum is more accommodating at the higher frequencies but less tolerant at the lower frequencies.

The ANSI method uses 65dB as the reference level to adjust for the overall level of the background noise level. If the background noise has an overall level of P dB, then (P –65) dB is added to each idealized spectrum third octave level. That is to a large extent $AI_ANSI is independent of the overall level. This is not the case for $AI_Veh which uses a fixed idealized speech spectrum level.

The ANSI scheme also has an absolute 'maximum tolerable level' and a 'threshold level' for each third octave band. Thus if any adjusted level is above or below these, then the corresponding limit value is used in the adjusted spectrum. There is also another aspect in the $AI_ANSI calculation for high overall level signals. This is an anechoic correction which basically reduces the idealized speech spectrum so that the $AI_ANSI value falls with very loud background noise levels. The $AI_Veh and $AI_Room calculations do not have these factors.

The final difference between the three approaches is that each has different weighting values. All the sets of weighting values are biased towards the 1.6 and 2kHz bands with the $AI_ANSI being slightly flatter. Actually the $AI_Room calculation method is slightly different as it uses a comparison vector for each third octave band. If a measured third octave noise level is less than j comparison levels in its vector then the added contribution is (j

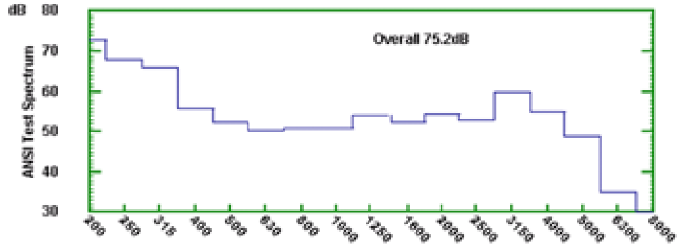


Figure 2: Third octave background noise level

* 0.01). There are of course 100 comparison levels.

Figure 2 shows the example third octave background noise level given in the ANSI specification. This has an overall level of 75.2dB and an ANSI Articulation Index of 0.547

The $AI_ANSI and $AI_Veh values were calculated for this spectrum and several identically shaped spectra adjusted to different overall levels. The loudness in Sones was also computed. Results are shown in the table below.

Overall dB	$AI_ANSI	$AI_Veh %	Loudness Sones
45	0.547	99.70	2.93
55	0.547	94.21	6.24
65	0.547	76.89	12.31
75	0.547	46.61	23.35
85	0.544	18.77	43.27
95	0.410	2.75	79.37
105	0.204	0	149.65

Note The $AI_ANSI value is shown as an index from zero to unity but that $AI_Veh is shown as a percentage.

From the table it is clear that the ANSI AI is sensibly independent of the overall level until the anechoic factors take effect at high overall levels. The Vehicle AI however with its fixed target does vary with overall level. It has essentially an inverse relationship of some form to loudness.

Both AI calculation methods are valid for the purposes for which they were designed. The ANSI version tests speech intelligibility, the vehicle and room versions test what may be called normal level speech quality.

A, B & C Weighting

A little history and insight into acoustic weighting

Dr Colin Mercer

The DATS analysis function WEIGHT provides the ability to apply A, B, C or D weighting to any frequency spectra. The input may be an FFT, an auto-spectrum or a cross spectrum and may be in real, complex or modulus & phase form.

Some devices, particularly digital tape recorders, apply A-weighting to all their data in order to achieve acceptable data compression. This is fine unless you want to analyze the unweighted data or apply a different weighting factor. Using DATS it is a simple task to instruct the WEIGHT module to either simply unweight the data or remove one weighting factor and apply another.

The presence of the Named Element $WEIGHT in a signal is used to tell DATS whether any weighting has been applied to a signal. Correctly setting this for data gathered with A-weighting will inform the WEIGHT module to treat it accordingly.

The screenshot above shows four DATS signals. Each one is the frequency spectrum of a broad band random input. The first ,dark blue is unweighted and the red trace shows the same data A-weighted. It can be easily seen how A weighting depresses frequencies below 500Hz whilst increasing slightly those above 1250Hz. For completeness the B-weighted signal is shown along with the C-weighted one. These weightings suppress frequencies below about 250Hz and 20Hz respectively.

D weighting, which for clarity is not shown, is similar to B weighting except that it significantly boosts frequencies in the 1250Hz to 10kHz region. It was designed specifically for assessment of aircraft noise.

Generally speaking the overall level found from A weighted spectra correlates well with subjective assessment of loudness. The C weighting curve gives equal emphasis over the normal hearing range from 31.5Hz to 8kHz.

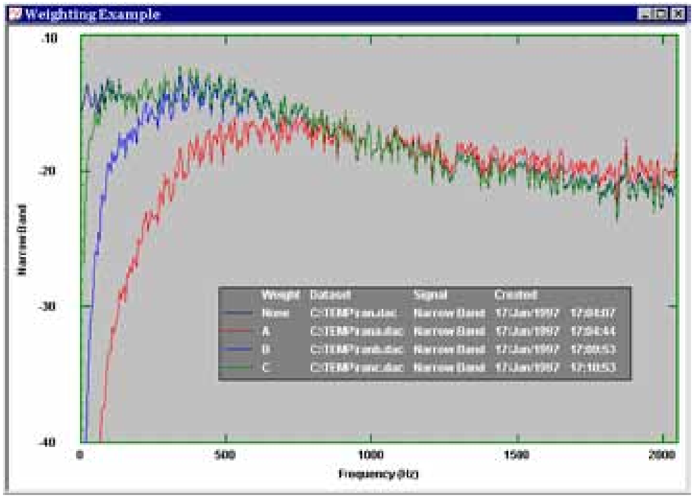


Fig. 1 : Example of A, B & C weighting

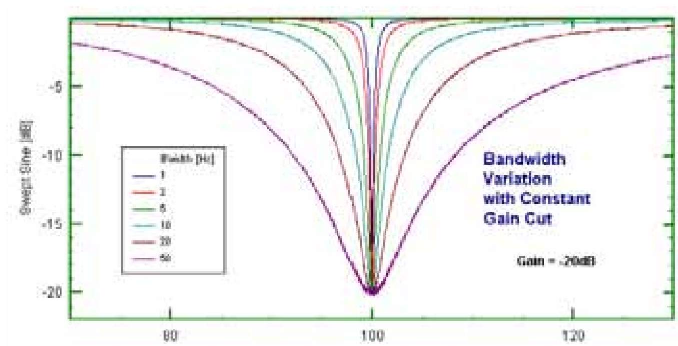
Audio Equalization Filter & Parametric Filtering

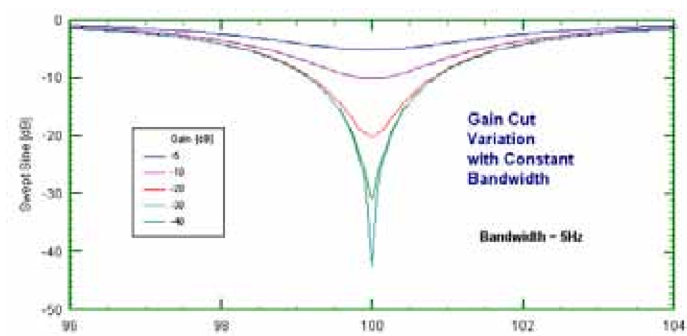
Dr Colin Mercer

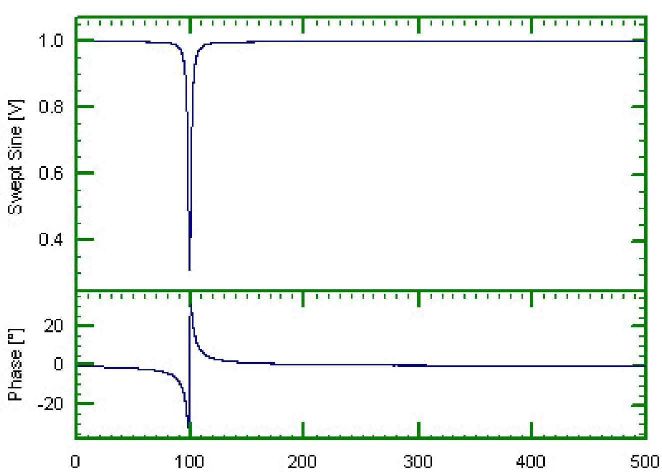
An interesting audio filtering technique

When working with audio signals a common requirement is to be able to equalise, cut or boost various frequency bands. A large number of hardware devices on the market provide this capability. The key aspect is that such filters are able to control bandwidth, centre frequency and gain separately. There are broadly two classes of filter used, a "shelving" filter and an "equalizing "filter (also known as a "peak" filter). A shelving filter is akin to low pass and high pass filters. An equalizing filter is like a bandpass or band reject filter.

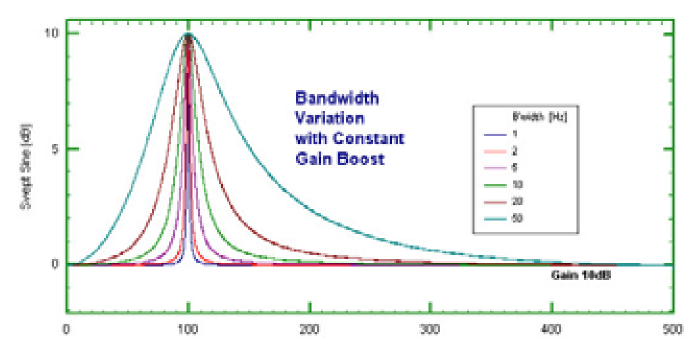
For sound quality replay and similar the interest is in equalizing filters, specifically in conjunction with removing narrow band resonances or, when dealing with rotating machines, with removal of orders. Actually it is not necessarily the removal of an order but its reduction, or increase, by a specific amount (gain).

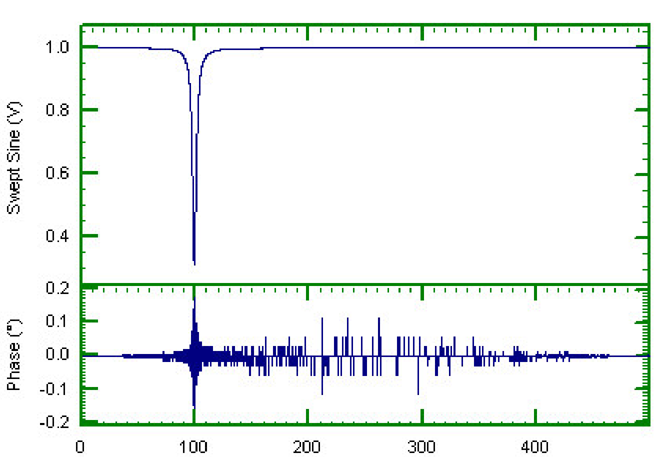




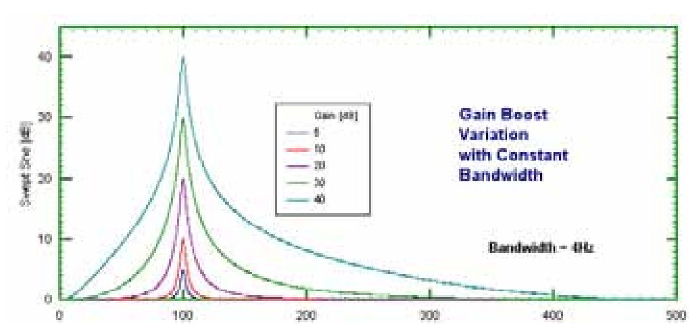


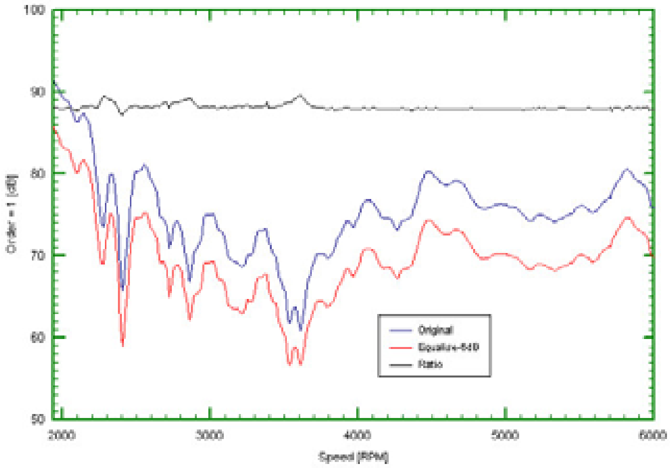
Standard phase





"Phaseless"





The basic component of an equalizing filter is an All Pass filter used in a feedback loop. Equalizing filters could be based upon standard filters such as Butterworth, Chebyshev and similar. However, experience in audio reconstruction suggests that a more "rounded" filter characteristic is better for audio replay. Accordingly a software version of a standard audio equalizing filter has been implemented in DATS. The initial version allows simple band reject (cut) and band increase (boost) by setting the dB gain as negative for cut and positive for boost. The bandwidth and centre frequency are specified independently.

Another aspect caused by filtering is the phase of the output signal. The software allows a choice of "phaseless" or standard filtering. The results for a typical set of values (gain = -10 dB, bandwidth = 4Hz) are shown above. In this example the cut filter gives a +/- 30 degrees of phase change. The phaseless implementation reduces this to less than +/- 0.2 degrees.

A major use of the equalizing filter is in enhancing or cutting orders by a specific amount. For example it then becomes possible to consider what a signal sounds like if a particular order was reduced by N dB.

In the example below we have used the standard run down dataset and reduced first order by 6dB. We used one tacho pulse per rev.

After equalizing the original time history both signals were waterfall analyzed and first order extracted. The results are shown below.

The 6dB reduction is clearly seen. The ratio (black line) between the two order cuts should be a constant, which it is except at rapid rates of change of the order. Even there the variations are generally within +/- 1dB.

Fourier Analysis - The Basics & Beyond

Dr Colin Mercer

A close look at the Fourier Analysis that we know and love

Fourier analysis takes a signal and represents it either as a series of cosines (real part) and sines (imaginary part) or as a cosine with phase (modulus and phase form). As an illustration we will look at Fourier analyzing the sum of the two sine waves shown below. The resultant summed signal is shown in the third graph.

If we now carry out a Fourier Analysis, in this case with an FFT, of the combined signal then we obtain the following result.

We see immediately that there are two distinct peaks in the modulus curve and two distinct changes in the phase curve at 64Hz and at 192Hz as expected.

The amplitude shown is exactly half of the original constituent sine waves. That is, the sine wave of unity amplitude at 64Hz is shown as 0.5 and the sine wave of amplitude 0.25 is shown as 0.125. Why is this? The reason is that when we do a frequency analysis of a signal some of the 'energy' is represented for positive frequencies and half for the negative frequencies. For a real time signal, as opposed to a complex time signal, then this energy is split equally and we get exactly half. Some software packages do a doubling to overcome this but this is not done in DATS. This is to make so called half range analysis compatible with full range analyses.

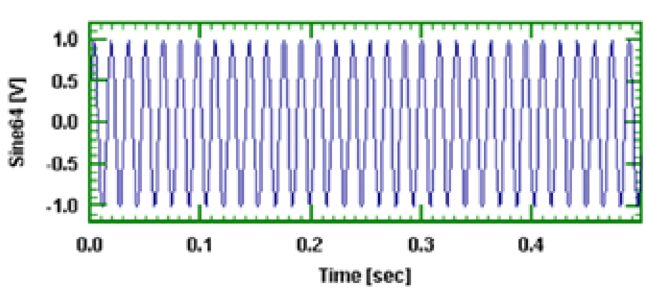


Figure 1 Sine wave, 64Hz, unit amplitude, zero phase

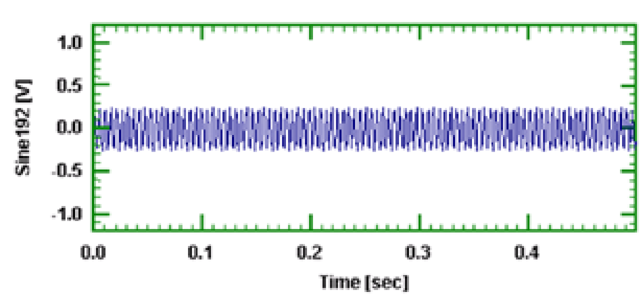


Figure 2 Sine wave, 192Hz, 0.25 amplitude, 30° phase

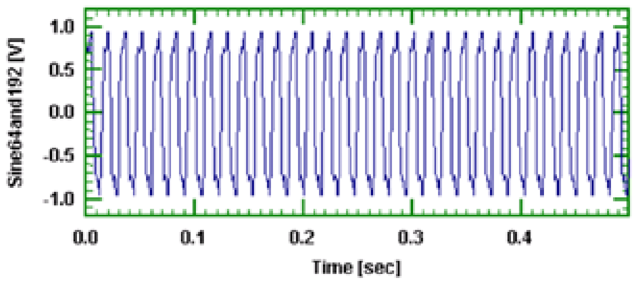


Figure 3 Combined Sine waves

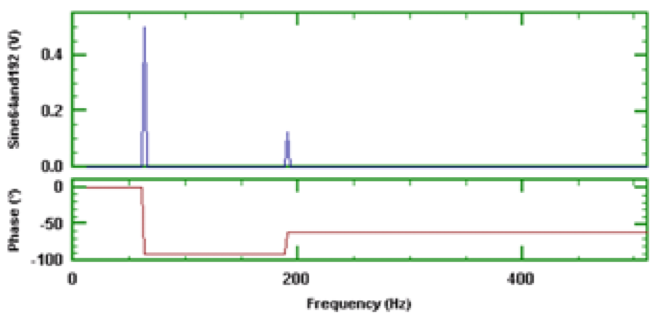


Figure 4. FFT of 64Hz & 192Hz signals

Sine Wave Amplitude	Peak to Peak Value	FFT or DFT Value
A	2A	A/2

Table 1. Amplitude Relationship

Now consider the phase part. The original 64Hz sine had a zero degree phase and the 192Hz had a 30° phase. From the phase plot at 64Hz the phase jumps from 0° to -90°. Why? This is because Fourier analysis uses cosines and sines. It is cosines, not the sines, which are the basic reference. Because a sine wave is a -90° phase shifted cosine then that is what we get. The phase shift at 192Hz was not 30° but -60°. This is totally correct as we have (-90+30) = -60°. Further explanation of this is given in the slightly more mathematical part at the end of these notes.

In the above examples the signals were represented by 512 points at 1024 samples/second. That is we had 0.5 seconds of data. Hence, when using an FFT to carry out the Fourier analysis, then the separation between frequency points is 2Hz. This is a fundamental relationship. If the length of the data to be frequency analyzed is T seconds then the frequency spacing given by an FFT is (1/T)Hz.

Selecting the FFT size, N, will dictate the effective duration of the signal being analyzed. If we were to choose an FFT size of say 256 points with a 1024 points/second sample rate then we would use 1/4 seconds of data and the frequency spacing would be 4Hz.

As we are dealing with the engineering analysis of signals measuring physical events it is clearly more sensible to ensure we can set our frequency spacing rather than the arbitrary choice of some FFT size which is not physically related to the problem in hand. That is DATS uses the natural default of physically meaningful quantities. However it is necessary to note that some people have become accustomed to specifying "block size". To accommodate this DATS includes an FFT module shown as FFT (Select) on the frequency analysis pull down menu. This module does allow a choice of block size.

"Non Exact" Frequencies

In the above examples the frequency of the sine waves were exact multiples of the frequency spacing. They were specifically chosen that way. As noted earlier 0.5 seconds of data gives a frequency spacing of exactly 2Hz. Now, suppose we have a sine wave like the original 64Hz sine wave but at a frequency of 63Hz. This frequency is not an exact multiple of the frequency spacing. What happens? Visually it is very difficult to see any difference in the time domain but there is a distinct difference in the Fourier results. The graph below shows an expanded version of the result of an FFT of unit amplitude, zero phase, 63Hz sine wave.

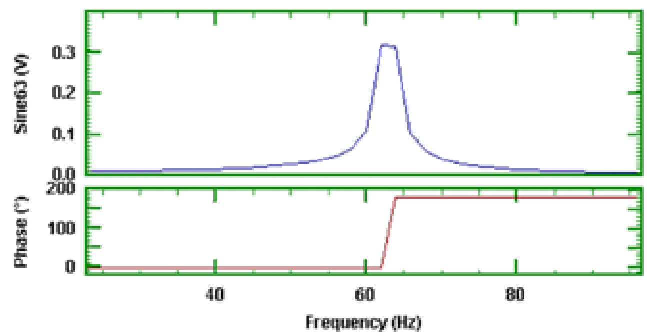


Figure 5 FFT of 63Hz, a "non-exact" frequency

Note that there is not a single spike but rather a 'spike' with the top cut off. The values at 62Hz and 64Hz are almost identical, but they are not 0.5, rather they are approximately 0.32. Furthermore the phase at 62Hz is 0° and at 64Hz it is 180°. That is the Fourier analysis is telling us we have a signal composed of multiple sine waves, the two principle ones being at 62 and 64Hz with half amplitudes of 0.32 and a phase of 0° and 180° respectively. In reality we know we had a sine wave at 63Hz.

If we overlay the modulus results at 63Hz and 64Hz then we note that the 63Hz curve is quite different in characteristic to the 64Hz curve.

This shows that care needs to be taken when interpreting FFT results of analyzing sine waves as the value shown will depend upon the relationship between the actual frequency of the signal and the "measurement" frequencies. Although the amplitudes vary significantly between these two cases if one compares the RMS value by using Spectrum RMS over Frequency Range then the 64Hz signal gives 0.707107 and the 63Hz signal gives 0.704936.

The above results were obtained using an FFT algorithm. With the FFT the frequency spacing is a function of the signal length. Now given the speed of the modern PC then we may also use an

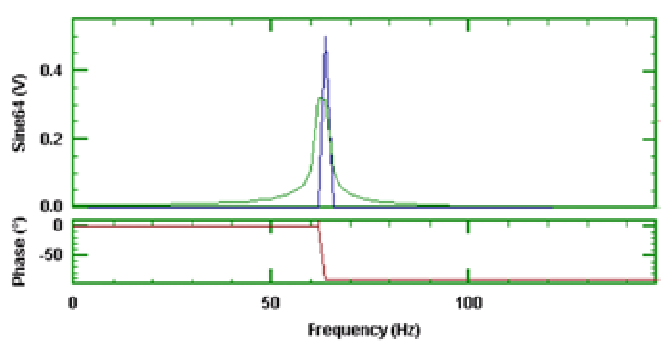


Figure 6. Overlay FFT of 63Hz and 64Hz signals

original Direct Fourier Transform method. In particular the DFT (Basic Mod Phase) version in Frequency Analysis (Advanced) allows a choice of start frequency, end frequency and frequency spacing. The DFT is much slower than the FFT. Choosing to analyze from 40Hz to 80Hz in 0.1Hz steps gives the results shown below with the continuous curve. The * marks are those points from the corresponding FFT analysis.

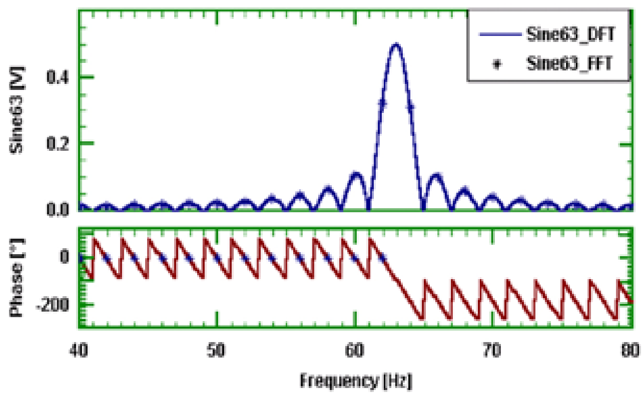


Figure 7. DFT analysis of 63Hz Sine wave

This now shows the main lobe of the response. The peak value is 0.5 at 63Hz and the phase is -90°. Also from 62Hz to 64Hz the phase goes from 0° to -180°. Note that this amount of phase change from one "Exact" frequency to the adjacent one is typical.

The above plot shows all the "side lobes" and illustrates another aspect of digital signal processing, namely the phenomenon known as spectral leakage. That is in principle the energy at one frequency "leaks" to every other frequency. This leakage may be reduced by a suitable choice of data window. The shape of the curve in Figure 7 is actually that of the so-called "spectral window" through which we are looking at the data. It is often better to think of this as the shape of the effective analysis filter. In this example the data window used is a Bartlet (rectangular) type. Details of different data windows and their corresponding spectral window are discussed in a separate article.

In this note we have been careful to use "frequency spacing" rather than "frequency resolution". It is clear that with DFT and other techniques we can change the frequency spacing. For an FFT method the spacing is related to the "block size". But what is the frequency resolution? This is a large subject but we will give the essence. The clue is the shape of the spectral window as illustrated in Figure 7. A working definition of frequency resolution is the ability to separate two close frequency responses. Another common definition is the half power (-3dB) points of the spectral window. In practice the most useful definition is a frequency bandwidth known as the Equivalent Noise Band Width (ENBW). This is very similar to the half power points definition. ENBW is determined entirely by the shape of the data window used and the duration of the data used in the FFT processing.

Signal Duration Effects

If we have data taken over a longer period then the frequency spacing will be narrower. In many cases this will minimize the problem, but if there is no exact match the same phenomenon will arise.

Fourier analysis tells us the amplitude and phase of that set of cosines which have the same duration as the original signal. Suppose now we take a signal which again is composed of unit amplitude 64Hz sine wave and a 0.25 amplitude 192Hz sine wave signals but this time the 64Hz signal occupies the first half and the 192Hz signal occupies the second half. That is we now have a one second signal as shown below.

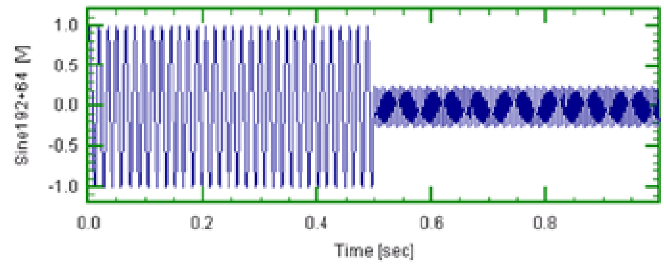


Figure 8. Two sines joined

The result of an FFT of these two joined signals is shown below.

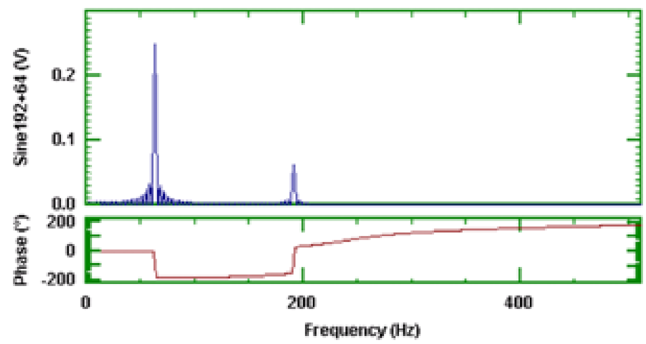


Figure 9 FFT of two joined sinewaves

There are, as expected, significant frequencies at 64Hz and 192Hz. However the half amplitudes are now 0.25 (instead of 0.5) and 0.0625 (instead of 0.125). One interpretation of what the FFT is telling us is that there is a cosine wave at 64Hz of half amplitude 0.25 for the whole one second duration and another one of half amplitude 0.0625 for the whole duration. But we know that we had a 64Hz signal with a half amplitude of 0.5 for the first part of the time and a 192Hz signal with a half amplitude of 0.125 for the second part. What is happening?

A closer look at the spectrum around 64Hz as shown below reveals that we have a large number of frequencies around 64Hz. This time they are 1Hz apart as we had one second of data. Their relative amplitudes and phases combine to double the amplitude at 64Hz over the first part and to cancel during the second part. The same of course happens in reverse around those frequencies close to 192Hz.

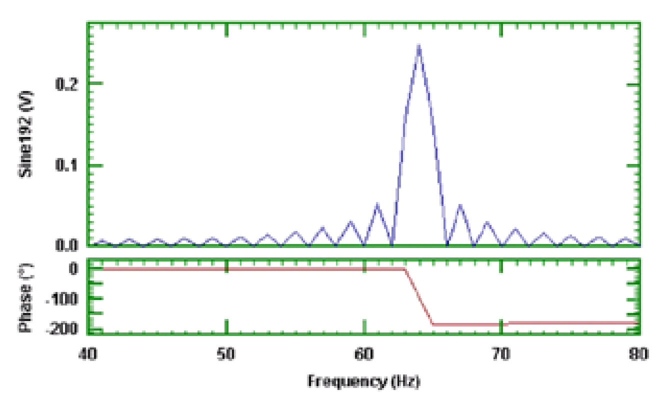


Figure 10. FFT (part) of joined signals

Another example is where a signal is extended by zeroes. Again the amplitude is reduced. In this case the reduction is proportional to the percentage extension by zeroes.

The important point to note is that the Fourier analysis assumes that the sines and cosines last for the entire duration.

Swept Sine Signal

With a swept sine signal theoretically each frequency only lasts for an instant in time. A swept sine signal sweeping from 10Hz to 100Hz is shown below.

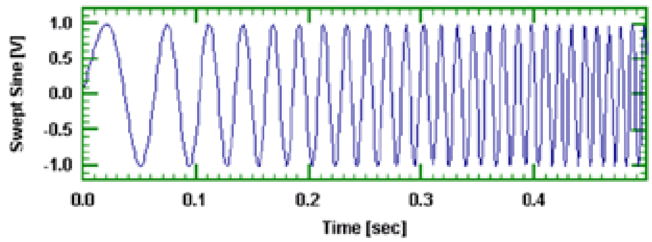


Figure 11 Swept sine, unit amplitude, !0 to 100Hz.

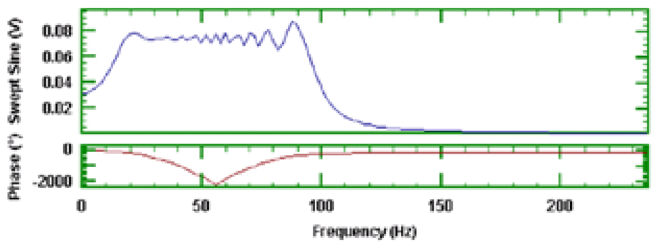


Figure 12 FFT of Swept Sine

This has 512 points at 1024 samples/second. Thus the sweep rate was 180Hz/second. The FFT of that signal shows an amplitude of about 0.075. Over the duration of the sweep the phase goes from around zero to -2000° and then settles to -180° above 100Hz. If the sweep rate is lowered to around 10Hz/second then the amplitude becomes about 0.019. The relationship between the spectrum level the amplitude and sweep rate of the original swept sine is not straight forward.

It is clear that one has to interpret a simple Fourier analysis, whether it is done by an FFT or by a DFT, with some care. A Fourier analysis shows the (half) amplitudes and phases of the constituent cosine waves that exist for the whole duration of that part of the signal that has been analyzed. Although we have not discussed it, a Fourier analyzed signal is invertible. That is if we have the Fourier analysis over the entire frequency range from zero to half sample rate then we may do an inverse Fourier transform to get back to the time signal. One point that arises from this is that if the signal being analyzed has some random noise in it, then so does the Fourier transformed signal. Fourier analysis by itself does nothing to remove or minimise the effects of noise. Thus simple Fourier analysis is not suitable for random data, but it is for signals such as transients and complicated or simple periodic signals such as those generated by an engine running at a constant speed.

We have not considered Auto Spectral Density (also sometimes called Power Spectral Density) or RMS Spectrum Level Analyses here. They are discussed in another article. However for completeness it is worth noting that the essential difference between ASD analysis and FFT analysis is that ASDs are describing the distribution in frequency of the 'power' in the signal whilst Fourier analysis is determining (half) amplitudes and phases. While ASDs and RMS Spectrum Level analyses do reduce the effects of any randomness, Fourier analysis does not. Where confusion occurs is that both analysis methods may use FFT algorithms. This is not to do with the objective of the analysis or its properties but rather with efficiency of implementation. After all every analysis will use addition. That is just a mathematical operation and so, in that sense, is the use of an FFT.

A Little Mathematics

We will not go into all the mathematical niceties except to see that a Fourier series could be written in the forms below. In real and imaginary terms we have

$$x(t_k) = \frac{a_o}{2} + \frac{1}{N}\sum_{n=1}^{N-1}(a_n \cos 2\pi f_n t_k + b_n \sin 2\pi f_n t_k)$$

and in modulus and phase form as

$$x(t_k) = \frac{1}{N}\sum_{n=0}^{N-1}\frac{C_n}{2}\cos 2\pi(f_n t_k + \theta_n/360)$$

The above forms are a slightly unusual way of expressing the Fourier expansion. For instance θ is in degrees. More significantly the product $f_n t_k$ is shown explicitly. Usually in an FFT then f_n is expressed as n/NΔt and t_k as kΔt where Δt is the time between samples. This gives the relationship of the form

$$x_k = \frac{1}{N}\sum_{n=0}^{N-1}(a_n \cos 2\pi nk/N + b_n \sin 2\pi nk/N)$$

However, the point of using $f_n t_k$ explicitly above is to indicate that nothing in the Fourier expansion inhibits the choice of actual frequency at which we evaluate the Fourier coefficients. The FFT gains speed by being selective about where it evaluates the coefficients and also restrictive in the values of N that are permitted. There are ways around these but in most implementations, for practical purposes N is restricted to being a power of 2.

This means that with a DFT we can actually evaluate the Fourier coefficients at any frequency provided we obey the anti aliasing (Nyquist) criterion. The DFT is slower than an FFT. Another way of getting at the finer detail and still getting some speed advantage is to use the so-called Zoom FFT based on the Chirp-z transform. Again the relative advantages are discussed elsewhere.

As a historical note it is perhaps interesting to recall that Fourier did not generate his series in order to carry out frequency analysis but rather to determine a least squares error approximation to a function.

What is Resonance?

James Wren

What is it and why is it important?

First, in order to explain resonance we have to explain the terms we will use.

- A period is the amount of time it takes to complete one cycle
- The number of cycles in one second is the frequency of an oscillation.
- Frequency is measured in Hertz, named after the 19th century German physicist Heinrich Rudolf Hertz
- One Hertz is equal to one cycle per second.

A resonance occurs when a structure or material naturally oscillates at a high amplitude at a specific frequency. This frequency is known as a structural resonant frequency. Typically a structure will have many resonant frequencies.

A dictionary definition of resonance gives us -

> *"the state of a system in which an abnormally large vibration is produced in response to an external stimulus, occurring when the frequency of the stimulus is the same, or nearly the same, as the natural vibration frequency of the system."*

When the damping in a structure is small, the resonant frequencies are approximately equal to the natural frequencies of the structure, which are the frequencies of free vibrations of the molecules of the material itself.

Furthermore, an individual resonance is the condition when a natural frequency of a structure or material and the frequency at which it is being excited are equal or very nearly equal. This results in the structure or material vibrating strongly and is the classical resonance state. This resonance state can often lead to unexpected behavior of the structure or material.

The lowest natural frequency, often called the fundamental frequency, is related to the material of which the structure is made. The greater the mass or density of the material the lower the fundamental frequency of vibration. The natural frequency is also related to the speed that a waveform can propagate through the structure. This is determined largely by the molecular make up of the material. Gas, for example, has many free molecules with high kinetic energy, so the waveform can move quickly through the material. A solid has far fewer free molecules and is much denser, therefore the waveform moves more slowly.

In order to measure a resonance of a structure or material with a system such as DATS-tetrad data acquisition hardware and DATS signal processing software it is necessary to attach an accelerometer to the structure. It is then required to excite or stimulate the structure with the frequencies that it is normally exposed to in its working life. For example, an automotive car tire would need to be subject to the frequencies it would encounter whilst in use. This would normally be accomplished by use of a shaker or a large heavy hammer. The tire for example would need to be tested in isolation, and not connected to anything else like the vehicle suspension or wheel rim as these other parts have their own resonant frequencies and would make the capture and analysis of the tire resonant frequency difficult.

The measured response from the accelerometer will be relative to the excitation and will only exhibit frequencies that are present in the excitation. The excitation must be an acceptable representation of the normal working frequencies applied to the structure or material. If the structure has a resonance in this frequency range there will be a large peak in the response spectrum. The frequency of this peak will correspond to one of the resonant frequencies of the structure or material. If no peak is detected then the resonant frequencies lie outside the operating range of the structure or material. In order to find the resonant frequencies of a structure or material it may be necessary to apply a wider range of frequency excitation.

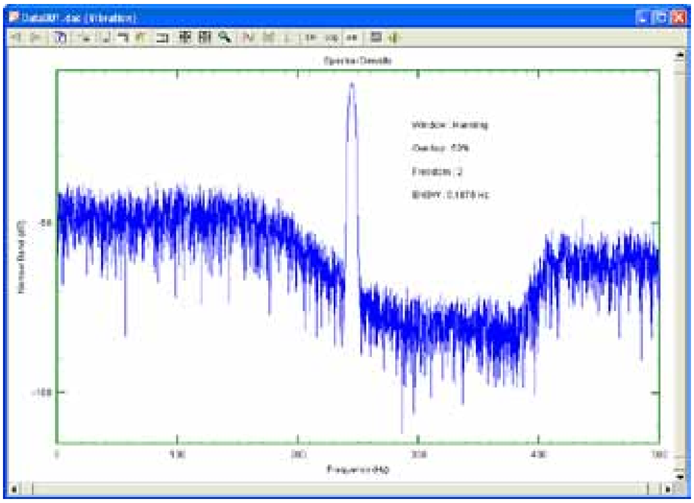


Figure 1

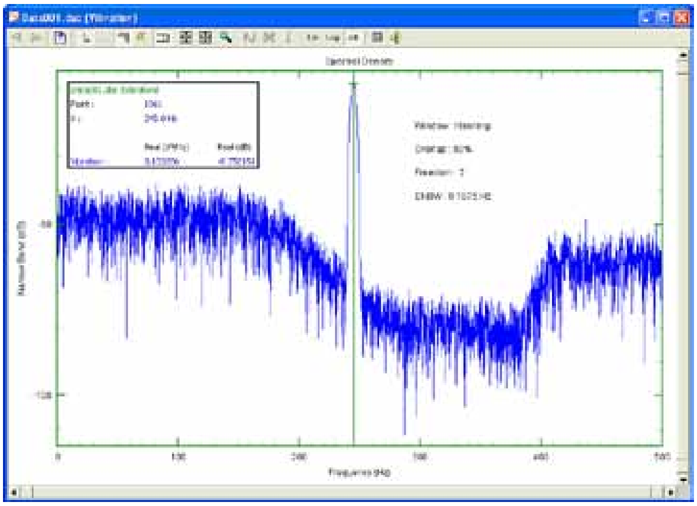


Figure 2

Figure 1 shows a frequency spectrum, this spectrum is a response of a structure to its excitation. A large spike can clearly be seen at approximately 250 Hz.

Figure 2 shows a frequency spectrum, this spectrum as in Figure 1 shows a frequency response. However, Figure 2 shows, using cursors, the exact frequency of the resonance. In this case the resonant frequency is 245 Hz.

This means that this structure should probably not be used if in its working life it will be exposed to this frequency. Figure 2 also shows that if this structure was to be used, and only exposed to 300Hz to 400 Hz or perhaps 0Hz to 200Hz , this particular resonant frequency would not be excited, and therefore the structure would not vibrate abnormally.

Vibration, Torsional Vibration & Shaft Twist

James Wren

These terms are often confused. So what do they mean?

When analyzing rotating shafts some terms are often confused. This post will attempt to explain the differences. So, what are vibration, torsional vibration and twist?

What is Vibration?

We measure at least one sensor on one channel of our measurement device to get a measure of vibration in one direction.

This one channel will measure the movement of a mechanical structure and provide the data as either displacement, velocity or acceleration. The type of sensor will dictate which of these three characteristics is measured. The simplest and most convenient sensor is an accelerometer. As the name suggests this measures acceleration.

Vibration may or may not be caused by the rotating of mechanical parts.

Vibration is usually studied in the time or frequency domain.

What is Torsional Vibration?

Torsional vibration will also usually require one measurement channel. Generally this would be a measurement from a shaft encoder or a toothed wheel with a high number of teeth. This will produce a high number of pulses for each revolution.

Torsional vibration (also known as angular vibration, transmission error (TE) or jitter) is the analysis of the torsional dynamic behavior of a rotating shaft.

Torsional vibration is different from the 'normal' vibration discussed above. It is the change in rotational velocity through a revolution. However, it can be expressed in displacement, velocity or acceleration.

As an example, consider a torsional system composed of a compressor, driver and coupling. This system can be modelled as a mass-elastic system (inertia and stiffness) to predict stresses in each component. The mass-elastic properties of the system can be changed by

- adding a flywheel (additional inertia)
- using a soft coupling (change in stiffness)
- viscous damping (absorb natural frequency stimulation)

The torsional vibration can be studied against time, in the angle (synchronous) domain and as orders.

What is Shaft Twist?

Twist requires the measurement of at least two positions. For example, either end of a crank shaft on an automotive engine. Usually, but not always, the measurement is from a shaft encoder or a toothed wheel with a high number of teeth.

The analysis is effectively the difference between the two measurement positions. So we are looking to see if one end leads or lags the other. These are usually analyzed in the time domain, often over one or two revolutions. As in the above examples this phenomenon is often analyzed as orders. However, this analysis is a magnitude and not a displacement, velocity or acceleration.

Amplitude Quantization Error

John Mathey

What is Amplitude Quantization Error and how can it be avoided?

Amplitude Quantization Error – Summary

This article explains what amplitude quantization error is and what can be done to avoid it. Typically quantization error is caused during acquisition of data. It is synonymous to rounding error. In order to explain this, a real time signal has been artificially modified to an extreme to show the difference between a signal sampled with adequate amplitude resolution and the same signal sampled with insufficient amplitude resolution. Also, this example was taken into the frequency domain to demonstrate how quantization error can affect the spectrum of the signal.

The Problem

Amplitude Quantization Error is typically caused by not having sufficient amplitude resolution to accurately capture the exact amplitude of the signal. With the modern ADC's using 24 or even 16 bit resolution this is not typically a problem, however measurement systems with fewer bits of amplitude resolution can compromise the data.

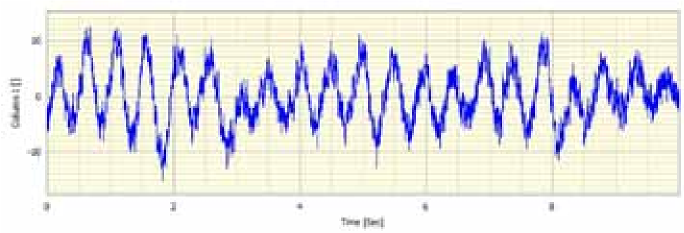


Figure 1: Data sampled with adequate amplitude resolution

The cause of this insufficient resolution is based on the sampling of the level of a signal, or in other words, having too few bits to accurately define the signal level during the sampling process. Typically analog to digital converters have a specified voltage range which they can accept. The Prosig hardware ADC range is ±10 Volts. The ADC's in these systems are 24-bit which means the total range of 20 Volts (±10 Volts) are 16777216 (224) different levels. This should be more than sufficient number of levels to adequately define the signal level from most transducers. To demonstrate this, I have sampled a signal with adequate amplitude resolution (Figure 1).

However, under rare circumstances if the transducer used has a very low sensitivity (e.g. 1μV/EU) and the system gain is not adjusted to use the full ADC range, this can lead to the amplitude not having adequate resolution to accurately described the signal. If only ±1 V or less of the ±10 V available of the ADC range is used there are far fewer levels which can be used to define the signal level.

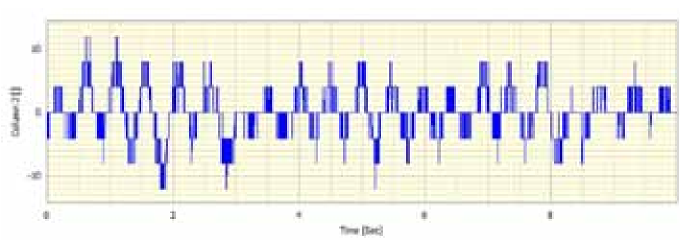


Figure 2: Extreme quantization (rounding error)

Now the same data artificially modified introducing an extreme amount of rounding error is shown in Figure 2. The levels of approximately -15 EU to +12.5 EU as displayed in Figure 1 has been rounded to quantized levels of -12, -8, -4, 0, 4, 8, and 12. This, of course, is an exaggeration, but it becomes visually apparent that there is a problem with this data.

What does this mean about the spectrum of a measurement taken with inadequate amplitude resolution? I have processed both these signals using the RMS Auto spectrum analysis tool and overlaid the results (See Figure 3). At first thought one might

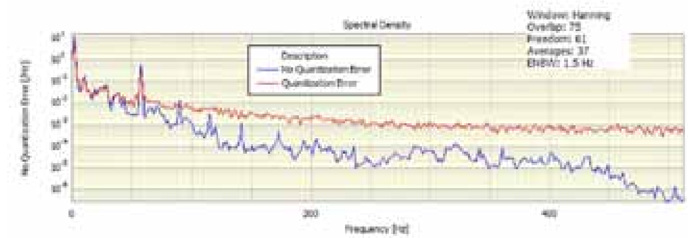


Figure 3: RMS Auto Spectrum

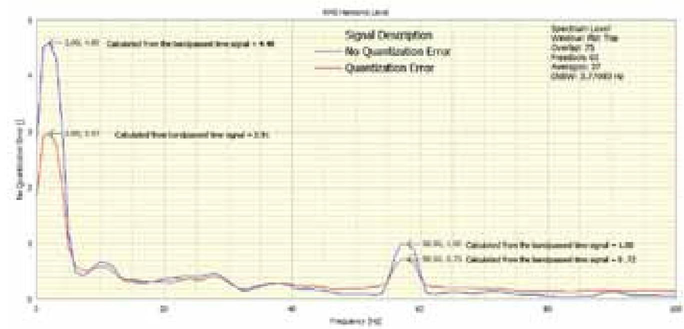


Figure 4: Spectral Amplitude Comparison

expect to see this quantization carried over into the spectrum. This is not the case due to how the amplitude of each of the frequency bins are calculated in the FFT calculations.

A significantly higher noise floor is apparent for the signal with the quantization error. This can potentially mask peaks in the spectrum at or below this noise floor (Figure 3).

Taking a closer look, it appears there is a discrepancy in the amplitudes of the two signals (Figure 4). Further investigation into this was to bandpass the time signals (w/o quantization error and w/ quantization error) around the frequencies of the 2 dominant peaks in the spectral data (2 Hz and 58 Hz). The RMS values of each of these 2 bandpass time signals were calculated and there is very close agreement to those values in the spectral calculations (See these values displayed in Figure 4).

Summary and Solutions

There are two quick solutions to the problem of amplitude quantization error. First is to use the appropriate transducer for the range of levels measured with the transducer and the second is to apply gain to the signal to use at least 50% of the available ADC range of the measurement system. In most cases one or both suggestions used together will resolve the situation of inadequate amplitude resolution.

Aliasing, Orders & Wagon Wheels

Dr Colin Mercer

Who remembers the old B&W Westerns and wagon wheels spinning backwards?

These days most people collecting engineering and scientific data digitally have heard of and know of the implications of the sample rate and the highest observable frequency in order to avoid aliasing. For those people who are perhaps unfamiliar with the phenomenon of aliasing then an Appendix is included below which illustrates the phenomenon.

In saying that most people are aware of the relationship concerning sample rate and aliasing this generally means they are aware of it when dealing with constant time step sampling where digital values are measured at equal increments of time. There is far less familiarity with the relevant relationship when dealing with orders, where an order is a multiple of the rotational rate of the shaft. For example second order is a rate that is exactly twice the current rotational speed of the shaft. What we are considering here then is the relationship between the rate at which we collect data from a rotating shaft and the highest order to avoid aliasing.

The relationship depends on how we do our sampling as we could sample at constant time steps (equi-time step sampling), or at equal angles spaced around the shaft (equiangular or synchronous sampling). We will consider both of these but first let us recall the relationship for regular equi-time step sampling and the highest frequency permissible to avoid aliasing. This is often known as Shannon's Theorem.

Standard Aliasing

With regular time based sampling using uniform time steps we have a sample rate of say S samples/second. That is digital values are taken 1/S seconds apart. For convenience let δt be the time increment in seconds so that δt = 1/S seconds.

With regular time domain processing we have a time and frequency relationship. That is if we carry out a Fourier analysis of a regularly spaced time history then we get a frequency spectrum.

Shannon's aliasing theorem states that if we have a sample rate S then the highest frequency we can observe without aliasing is (S/2) Hz. (S /2) is known as the Nyquist frequency. As previously mentioned the implication and results of aliasing are illustrated in an appendix below.

So if we have a time step δt then the highest frequency, f_{max}, is given by

$$f_{max} = \frac{1}{2\delta T} = \frac{sample\ rate}{2} = \frac{S}{2} Hz$$

This is a relationship between time steps in seconds and the highest frequency in Hz. It is worth noting that originally frequencies were specified in units of "cycles per second", and that the fundamental units of Hz are 1/second.

Highest order with time based sampling

First recall that orders are multiples of the rotation speed of the shaft. So if a shaft is rotating at R rpm (revs/minute), then the Nth order corresponds to a rotational rate of (N*R) rpm. So if a shaft is notating at 1000 rpm then second order is 2000 rpm but if the shaft rotation speed was 1500 rpm then second order corresponds to a speed of 3000 rpm. Orders are independent of the actual shaft speed, they are some multiple or fraction of the current rotational speed.

The relationship between order and frequency for a given basic rotation speed of R rpm is simply:

$$1\,order = Rrpm = \frac{R}{60} revs/sec = \frac{R}{60} cycles/sec = \frac{R}{60} Hz$$

Putting this relationship into the regular time based relationship to find the highest order to avoid aliasing gives:

$$Kth\ order = KRrpm = \frac{KR}{60} revs/sec = \frac{KR}{60} cycles/sec = \frac{KR}{60} Hz$$

That is the highest order, K_{max}, when using time based sampling at S samples/second is given by:

$$K * (R/60) = S/2$$

Synchronous or Equiangular Sampling

With equiangular sampling we take N points per revolution, typically by using a toothed wheel or similar to give exactly N points per revolution. This is again independent of the actual shaft speed. So our sample rate is N points/revolution.

$$K_{max} = \frac{S/2}{R/60} = \frac{Sample\ Rate}{2 * rev\ per\ sec}$$

With equiangular sampling we are in the "revolution" domain and the corresponding domain is the order domain. That is if we carry out a Fourier analysis of an equiangular spaced signal then we get an order spectrum.

Applying the Shannon Theorem directly we have quite simply the result that with synchronous sampling where we have a sample rate of N points per revolution then our highest order to avoid aliasing, O_{max}, is given by

$$O_{max} = N/2$$

Incidentally, if we Fourier analyze over an exact number of revolutions, say P revs, then our order spacing is 1/P orders.

Spatial Sampling

Just for completeness, the same applies if we do spatial sampling by measuring at equally spaced distance increments. The corresponding domain is wave number. Thus if we sample a road surface at L points per metre then our highest wave number to avoid aliasing, w_{max}, is given by

$$w_{max} = L/2$$

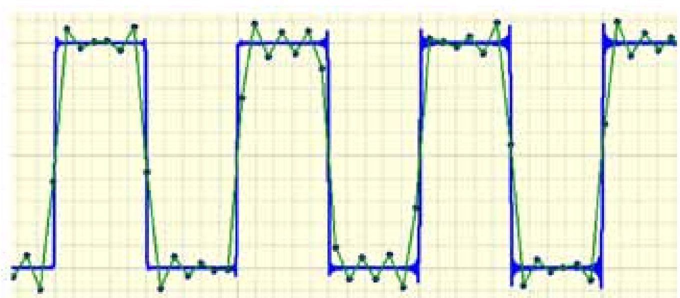
Why use synchronous sampling?

With high speed data acquisition systems it is quite usual to be able to sample at 100K samples/second. So if we are dealing with a shaft speed of say 6000 rpm, which is 100 revs/sec, then the highest order is [100000/ (2x100)] = 500. With synchronous sampling we would need an encoder with 1000 points/rev to achieve the same level. Often we are not interested in such high orders and there is rarely any high order content. That means we can use lower time based sample rates or lower points per revolution. So what is the advantage?

The essential point is that when we transform a synchronously sampled signal we are taken directly into the order domain. If we have sampled data covering exactly B revolutions then our order spacing will be (1/B) orders. Further the measurements at each order will be exact as they are precisely centred at each order.

If we have a time based signal there are two approaches. One way is to use waterfall analysis and the other is to convert the data to synchronously sampled data by software. Both of these methods require the tachometer signal to be captured at the same time. The accuracy depends upon precisely locating the tacho edges. The problem is illustrated below. Suppose the blue colored signal is the actual tacho signal and the green one is the digitally sampled tacho. Now the * on the measured signal (green) are the actual data points. The actual tacho goes between a low of zero and a high of unity. So suppose we set the tacho crossing level at 0.6 and use the first data point that occurs at or above the threshold level on a rising edge as determining the actual tacho crossing point. This will directly lead to time jitter, adding and subtracting from each period. This will appear as noise and possibly as false frequencies.

Clearly the software could improve the situation, for example in



DATS the software uses an interpolation process to determine a better estimate of the time location of the crossing point.

With waterfalls the next step is to determine a speed curve and then to carry out Fourier analysis at appropriate speed steps. The orders are then extracted from those frequency points which are the closest to the actual order being extracted. This is yet another approximation. An improved amplitude estimate is to determine the rms value over a short interval in the frequency domain. As one can see from the very description of the stages there exists considerable room for errors in the final order estimate.

When converting to synchronous sampling again it is essential to determine the tacho edge accurately. The next step is to interpolate the amplitude. This in theory can be done precisely by using a {(sin x) /x} basis but it requires signals and integrations of infinite length. In practice a reasonable estimate can be achieved with a relatively small number of points by using the most appropriate interpolation technique, such as a Lagrange based method.

Synchronous sampling avoids these complications.

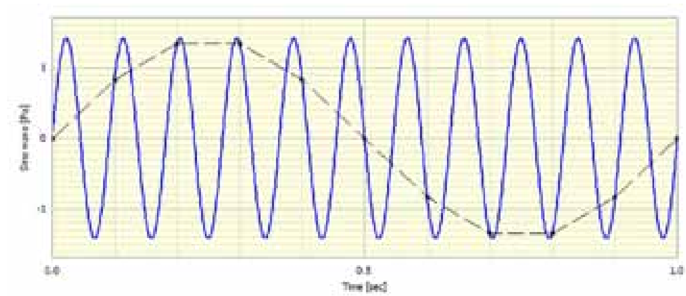
Appendix - Aliasing Demonstrations

One of the classical demonstrations of aliasing is the so called "wagon wheel effect" where in films as the stage coach goes faster the wheels appear to go slower ad then to go backwards. If you would like to see a visual demonstration of the wagon wheel effect then the link below is excellent.

Initially the spokes rotate anticlockwise then slow and begin to rotate clockwise. What is particularly nice in this presentation is that there are other 'markers' at a smaller angular spacing that remain in the "non-aliased" region. Thus one can see parts of the wheel rotating and other parts stationary, most impressive and convincing. There are sliders that allow manual control of the wheel speed.

See "Wagon-wheel effect" from **Michael's "Optical Illusions & Visual Phenomena"** (https://michaelbach.de/ot/mot-wagonWheel/index.html)

Another common form of illustrating aliasing is to show a high frequency sine wave sampled at too low a rate. If we sample too slowly we do not see the blue curve but rather we only see the data at the * points, which have been aliased to a much lower frequency.



For those of a mathematical leaning then we may find the apparent digital frequency from the following formula:

$$f_d = |f_a - K f_s|$$

where f_s is the sample rate, f_a is the actual frequency, f_d is the 'digital' frequency, and K is the integer, starting from zero, such that $|f_a - K f_s|$ is at its smallest.

We may write this as

$$f_d = min_{k=0}^{\infty} |f_a - K f_s|$$

For example consider a sample rate of 500 samples per second and what happens to various frequencies.

Actual Frequency f_a Hz	Minimizing value K	Alias frequency f_d Hz
180	0	180
280	1	220
380	1	120
480	1	20
580	1	80
680	1	180
780	2	220
880	2	120
980	2	20

Note how the pattern repeats.

Amplitude Modulation

James Wren

A strange phenomenon that appeared during some data analysis

When recently dealing with some vibration data from a runup of a pump, we observed some strange phenomena in the results. It turned out to be a classic case of amplitude modulation.

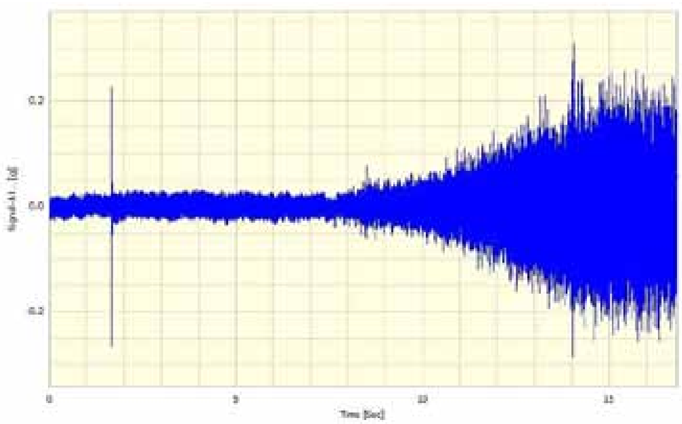


Figure 1: Original time history

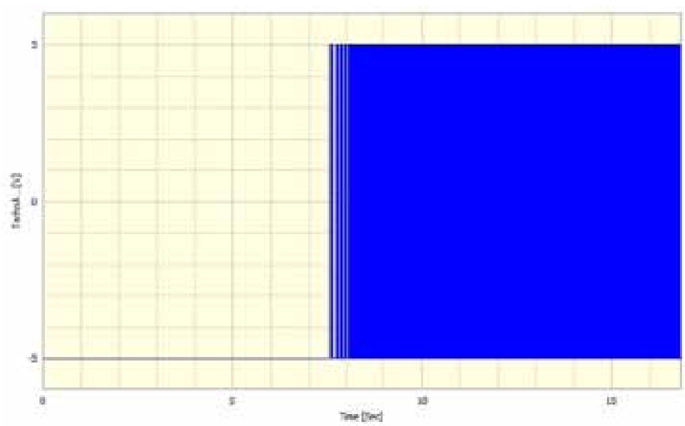


Figure 2: Tachometer signal

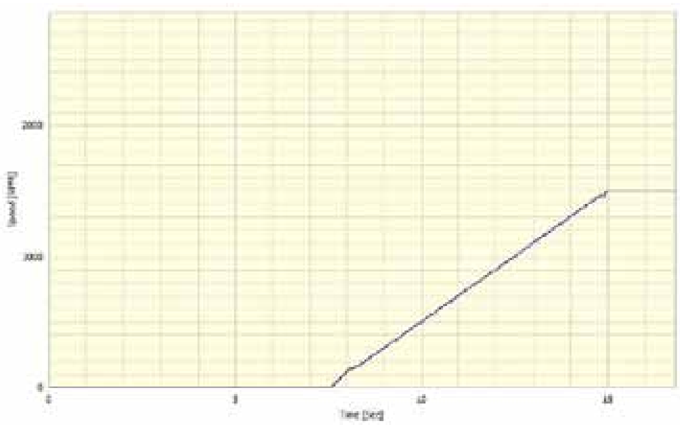


Figure 3: Speed v time curve

As shown in Figure 1 the original data looks a fairly standard run up time series. Figure 2 shows an associated tachometer signal and Figure 3 shows the speed curve calculated from the tachometer signal.

Waterfall analysis of the data, as shown in Figure 4, shows a number of interesting frequency components.

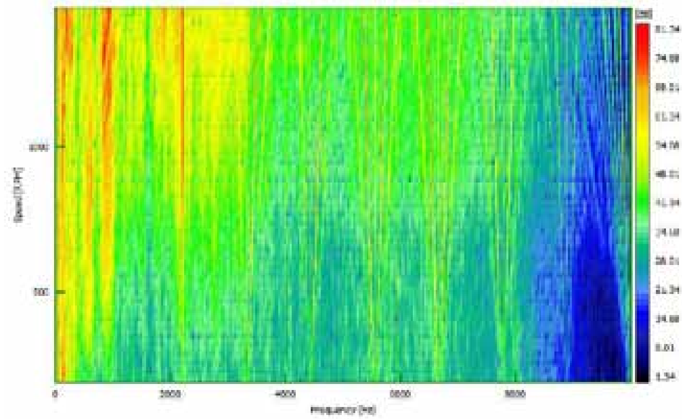


Figure 4: Speed v frequency waterfall

This shows a sharp central frequency with wider outer peaks. The presence of the outer peaks both above and below the central frequency suggests 'side-bands'. The presence of side bands around a central frequency is a well-known phenomenon which we discussed further in – Bearing & Gearbox Vibration Analysis using Demodulation Techniques? (Part 1)

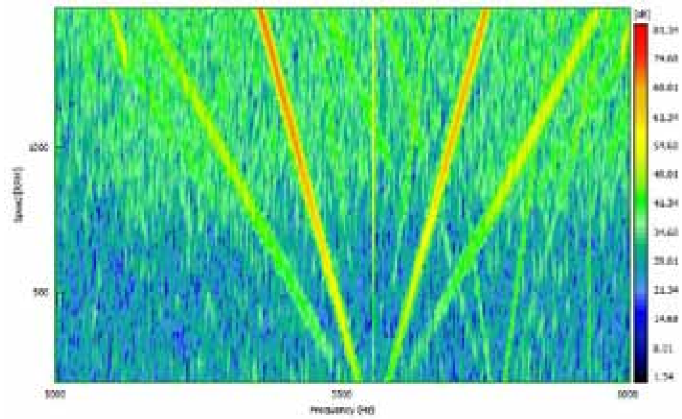


Figure 5: Zoomed section of speed v frequency waterfall

Figure 5 focuses on one particular frequency component in detail. The central, resonant frequency can be seen, with the side bands, and their harmonics, moving away from the central frequency on both the left and right-hand sides of the spectrum.

Sidebands are usually caused by amplitude modulation of a central carrier frequency. If the carrier frequency is A Hz and the frequency of the modulation is B Hz then the side bands will appear at frequencies (A+B) and (A-B) Hz.

In this case it is important to note that the carrier frequency is fixed whilst the sideband locations are changing with speed.

If we examine the frequency difference between the fixed frequency and the sidebands then we find that the difference is always a factor of eight of the rotational speed. For example, at 1000RPM (16.6Hz) the difference between the carrier and the side bands is around 132Hz (Figure 6) and at 1200RPM (20Hz) the difference between the carrier and the side bands is approximately 160Hz (Figure 7). Secondary sidebands are also visible at 16x rotational speed.

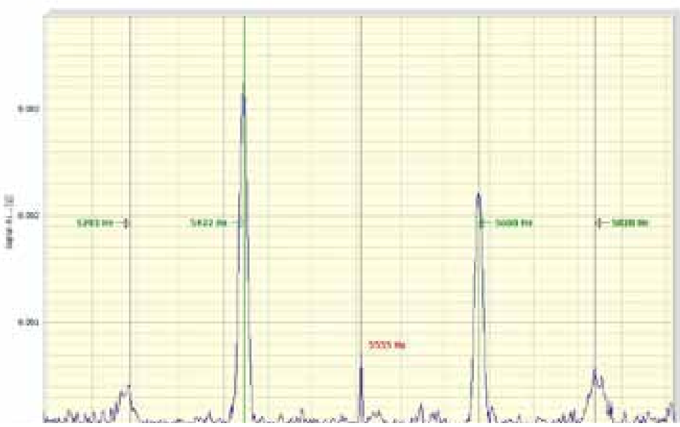


Figure 6: At 1000rpm

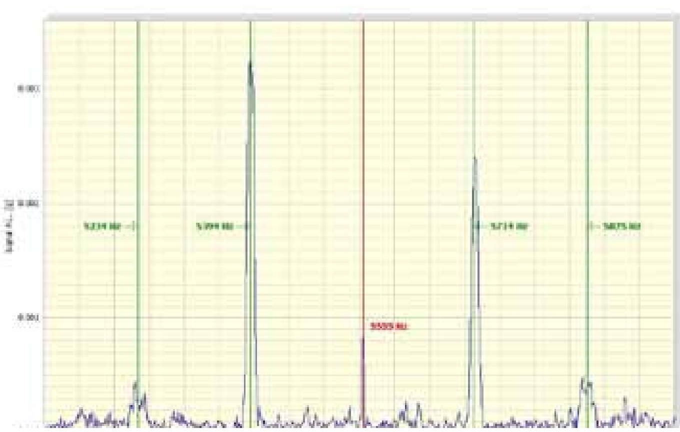


Figure 7: At 1200rpm

The observations show that a number of fixed frequency components are present in the vibration. These could well be due to resonances within the pump body or housing. The variable frequency components are no doubt caused by modulation which affects the signal on a cyclic basis. We know that the particular pump under test has 8 impeller blades, which matches the modulation frequency of 8 times the rotational speed.

As the impeller rotates around at speed, it can decelerate and then accelerates slightly around the revolution causing the modulation. The root cause may well be a blade fault on the impeller.

The modulation can be shown more clearly by band pass filtering the original pump vibration data. The result is a modulated sine wave.

Figure 8 shows a section of the original time signal. Figure 8 shows the same section of the original time signal, but band

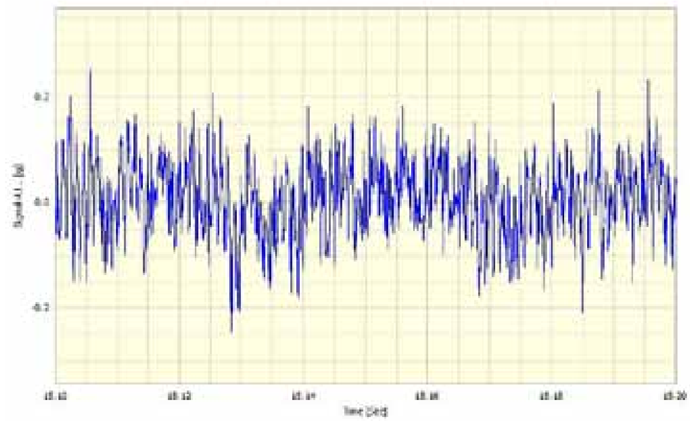


Figure 8: Section of original time history

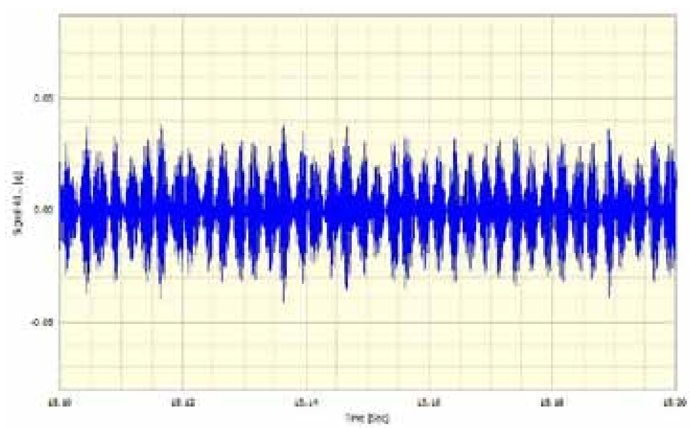


Figure 9: Section of original time history after filtering

pass filtered around one of the phenomena, in this case 5000Hz to 6200Hz.

In Figure 9 it is hard to make any clear deductions, indeed there appears to be general a noise issue, however once band pass filtered, as shown in Figure 8, the amplitude modulation effect can clearly be seen.

This wave shape shows that the initial assumptions were indeed proven correct. A classical case of amplitude modulation causing sidebands in the frequency domain.

The presence of side bands around a central frequency is a well known phenomena and is discussed further in the following articles, written by Don Davies, that can be found later in this handbook...

Bearing & Gearbox Vibration Analysis using Demodulation Techniques? (Part 1) (pp.92-94)

Bearing & Gearbox Vibration Analysis using Demodulation Techniques? (Part 2) (pp.96-97)

How to Analyze Noise & Vibration in Rotating Machines

Chris Mason

What are the basic steps when analyzing rotating machinery?

In this article we will look at the basic steps behind a simple rotating machinery study. We won't look in great detail at some of the techniques involved – we deal with these elsewhere in this handbook and on the **Noise & Vibration Measurement Blog** at http://blog.prosig.com. This material is suitable for a newcomer to the field who understands the basic concepts of noise & vibration analysis, but has not dealt with rotating machinery before.

Why do we need to measure noise & vibration in rotating machines?

The analysis of rotating machinery is central to refinement activities in automotive and general industry. It also allows engineers to trace faults in gearboxes, transmission systems and bearings.

Every rotating part in a machine generates vibration, and hence noise, as a result of small imperfections in the balance or smoothness of the components of the machine. In addition, there are so-called "blade-passing" phenomena associated with blades of fans and pumps. In every case, we can relate the frequency of the vibration to the speed of the rotating machine. For example, a fan with five equally spaced blades will generate noise at five times the speed of rotation and sometimes at higher multiples still depending on the number of supports used to hold the fan in place. If these are close to the blades, then the frequency becomes the product of the number of blades and the number of supports.

These vibrations act as a forcing function on the structure of the machine or vehicle, where they are mounted. The most severe effects occur when the frequency of excitation generated by the rotating part matches one of the natural frequencies of the structure. These "coincidence" frequencies are often the target of much design effort to limit the effects, whether they be fatigue, vibration or resulting noise.

With variable speed machines, it is a considerable challenge to reduce noise and vibration to acceptable levels. The rotating components are often transmitting very large amounts of power and, unfortunately, even very small amounts of power, converted to vibration or noise, can produce undesirable effects.

Analyzing Data From Rotating Machinery

Let's assume we have captured a noise or vibration signal from some sort of rotating machine while we accelerate it through it's entire speed range. We will use a short noise signal recorded from a 4-cylinder race car engine. In figure 1 we can see the time history of the signal.

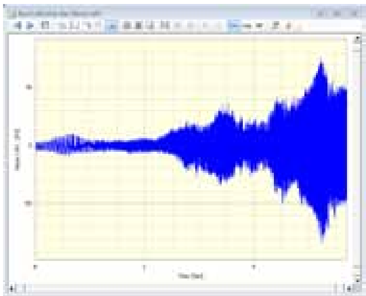


Figure 1: Time history of noise signal

The simplest way to analyze the frequency content of this data would be to calculate the auto-spectrum. We see the result in Figure 2.

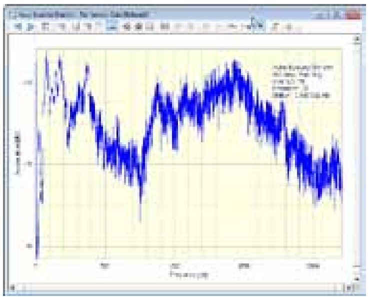


Figure 2: Frequency spectrum of the entire noise signal

Because the engine we were testing was changing speed it is almost impossible to draw any meaningful results from this spectrum. It is clear we need a different approach. Our first approach might be to segment the data into sections and look at a series of spectra generated from those segments. This is done with a "Hopping FFT". This takes a fixed length section of the time history, performs an FFT and then moves along a small increment and repeats the process. This produces a series of FFT spectra spread across the whole time period of the test as seen in Figure 3.

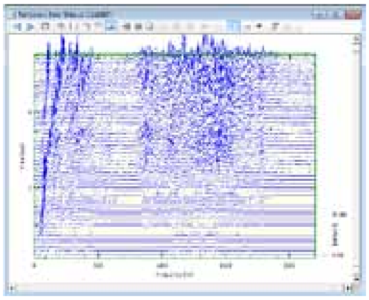


Figure 3: A classic waterfall plot (frequency v time)

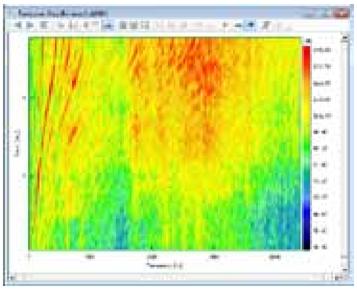


Figure 4: An intensity or color plot

Figure 3 is what most people visualize when they think of a waterfall plot. This data can be represented in many different ways. One of the most popular and useful is an intensity or color plot as seen in Figure 4.

The plot shows frequency along the bottom axis and time runs from bottom to top. It is clear that the red lines seen on the left part of the map represent the rising frequencies as the engine accelerates over time. So these are the effects due to rotation that we are interested in. What we are still missing, though, is any information about the speed of the engine.

Analyzing Speed Of Rotation

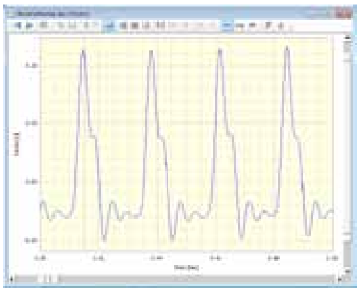


Figure 5: Tacho signal (slow speed)

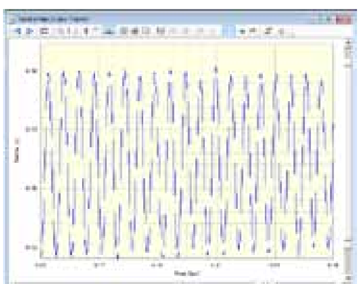


Figure 6: Tacho signal (high speed)

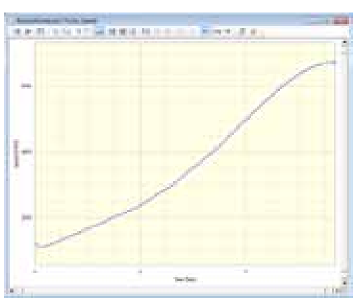


Figure 7: Speed v Time

We need to know the speed of the engine across the time period of our test. There are several ways to obtain this, but the most accurate is to capture some sort of tacho signal. Ideally, a tacho signal should be a square wave or pulse. It can be anything from a once per revolution pulse measured from some moving part of the engine to several thousand pulses per revolution generated by an encoder. Figures 5 and 6 show our tacho signal at two different times in our test.

Our signal was generating two pulses per revolution. Both figures show 0.1s of data so it is clear that in Figure 6 the engine was rotating significantly quicker than in Figure 5. It is plain that our signal is not a clean square wave, but so long as the tacho processing algorithm is sophisticated and robust enough this is not important. The tacho processing software analyses the entire tacho signal and produces another signal with represents speed over the time of the test. This can be seen in Figure 7.

Here we can see that, during our test the engine accelerated from just over 1000rpm to over 6000rpm in around 5.5 seconds.

Waterfall Plots

The next step in our processing is to perform our "Hopping FFT" processing again, but this time instead of calculating the FFTs at equally spaced time steps we calculate them at equally spaced speed steps using the information from our speed v time data. This is what is known as "Waterfall" processing. The waterfall plot of our noise signal can be seen in Figure 8.

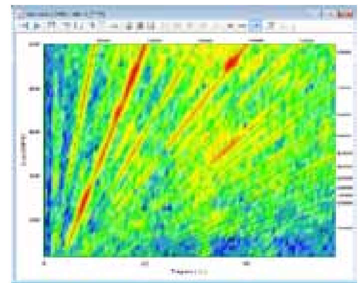


Figure 8: A Waterfall Plot (Frequency v Speed)

Now we can clearly see the linear nature of the speed related noise. This shows up as sloped lines in the waterfall plot. Because this is a four cylinder, four stroke engine, the dominant frequency is twice the speed of rotation or the second harmonic. In the case of rotating machinery we call this second order.

Orders

Details of the orders can be extracted from the waterfall plot and an example from our data can be seen in Figure 9.

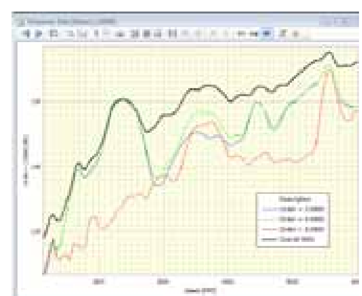


Figure 9: Orders and overall level

Figure 9 shows the 2nd, 4th and 6th orders overlaid with the overall level. The overall level represents the total energy at each speed.

Synchronously Sampled Data (The Order or Angle Domain)

Once data has been captured along with a tacho signal, software such as Prosig's DATS package, can resample the data and convert it to the angle domain. This means that each data point represents equally spaced positions around a rotation cycle rather than equally spaced points in time. This enables easy analysis with a DFT to extract the orders directly, even if the speed varies dramatically during a cycle. Data which is synchronously sampled in this way can be averaged across cycles in the angle domain, thus eliminating the noise of signals from other sources which are not related to the rotation. Further details of these techniques can be found elsewhere in this handbook and on the Noise & Vibration Measurement Blog - http://blog.prosig.com.

A Simple Frequency Response Function

James Wren

A look at the basic concepts and theory

The following article will attempt to explain the basic theory of the frequency response function. This basic theory will then be used to calculate the frequency response function between two points on a structure using an accelerometer to measure the response and a force gauge hammer to measure the excitation.

Fundamentally, a frequency response function is a mathematical representation of the relationship between the input and the output of a system.

So, for example, we can measure the frequency response function between two points on a structure. It would be possible to attach an accelerometer at a particular point and excite the structure at another point with a force gauge instrumented hammer. Then by measuring the excitation force and the response acceleration the resulting frequency response function would describe as a function of frequency the relationship between those two points on the structure.

The basic formula for a frequency response function is

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

Where H(f) is the frequency response function,

Y(f) is the output of the system in the frequency domain

and X(f) is the input to the system in the frequency domain.

Frequency response functions are most commonly used for single input and single output analysis, normally for the calculation of the H1(f) or H2(f) frequency response functions. These are used extensively for hammer impact analysis or resonance analysis.

The H1(f) frequency response function is used in situations where the output to the system is expected to be noisy when compared to the input.

The H2(f) frequency response function is used in situations where the input to the system is expected to be noisy when compared to the output.

Additionally there are other possibilities, but they are outside of the scope of this article.

H1(f) or H2(f) can be used for resonance analysis or hammer impact analysis. H2(f) is most commonly used with random excitation.

The breakdown of H1(f) is as follows,

$$H1(f) = \frac{S_{xy}(f)}{S_{xx}(f)}$$

Where H1(f) is the frequency response function,

$S_{xy}(f)$ is the Cross Spectral Density in the frequency domain of X(t) and Y(t)

and $S_{xx}(f)$ is the Auto Spectral Density in the frequency domain of X(t).

In very basic terms the frequency response function can be described as

$$H1(f) = \frac{Cross\ Spectral\ Density\ of\ the\ Input\ and\ Output}{Auto\ Spectral\ Density\ of\ the\ Input}$$

The breakdown of H2(f) therefore is as follows,

$$H2(f) = \frac{S_{yy}(f)}{S_{yx}(f)}$$

Where H2(f) is the frequency response function,

$S_{yx}(f)$ is the Cross Spectral Density in the frequency domain of Y(t) and X(t)

and $S_{yy}(f)$ is the Auto Spectral Density in the frequency domain of Y(t)

In very basic terms the frequency response function can be described as

$$H2(f) = \frac{Auto\ Spectral\ Density\ of\ the\ Output}{Cross\ Spectral\ Density\ of\ the\ Input\ and\ Output}$$

In the following example we will discuss and show the calculation of the H1(f) frequency response function.

The excitation or input would be the force gauge instrumented hammer, as shown in Figure 1 as a time history.

In this case the response or output would be the accelerometer, as shown in Figure 2.

Figure 1: X(t)

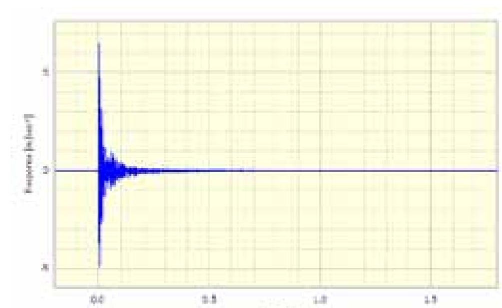


Figure 2: Y(t)

However as discussed earlier the frequency response function is a frequency domain analysis, therefore the input and the output to the system must also be frequency spectra. So the force and acceleration must be first converted into spectra.

The first part of the analysis requires the Cross Spectral Density of the input and output, this is $S_{xy}(f)$. This is calculated using the response as the first input and the excitation as the second input to the Cross Spectral Density Analysis in DATS. The result is shown in Figure 3. Were $S_{xy}(f)$ being calculated for use with H2(f), for example, then the excitation would be the first input and the response the second input to the Cross Spectral Density Analysis in DATS.

Figure 3: Sxy(f)

Next the Auto Spectral Density of the input, or excitation signal is required. This is calculated using the Auto Spectral Density Analysis in DATS, this analysis is sometimes known as Auto Power, the result of which is shown in Figure 4, this is $S_{xx}(f)$.

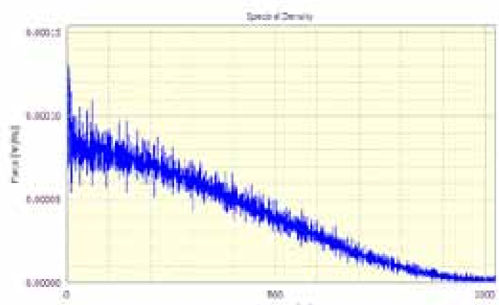


Figure 4: Sxx(f)

The Cross Spectrum is then divided by the Auto Spectrum and the resulting frequency response function is shown in Figure 5.

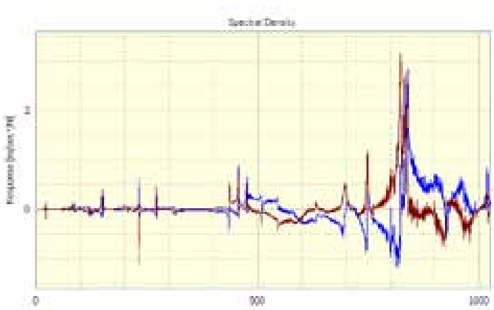


Figure 5: H1(f)

The response function would normally be shown in modulus & phase form as shown in Figure 6.

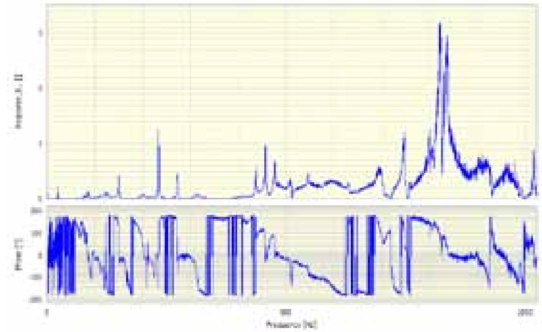


Figure 6: H1(f) shown as modulus & phase

The entire analysis as used in DATS-toolbox is shown in Figure 7, the data flow from the original input and output, force and response, can be seen through to the frequency response function. The DATS software does, of course, provide a single step transfer function analysis. We have deliberately used the long-hand form below to illustrate the steps in this article.

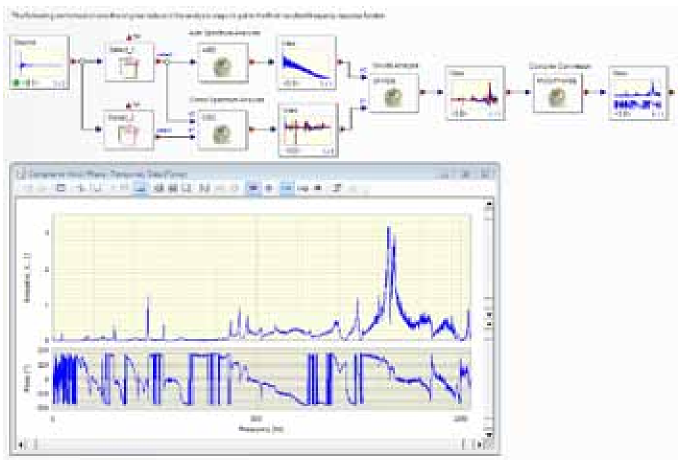


Figure 7: Complete DATS worksheet

It is necessary to understand that for the purposes of clarity in this article some important steps have been glossed over; windowing of the input for example. This is to assist in the basic understanding of the frequency response function.

Frequency, Hertz & Orders

Dr Colin Mercer

Some of the basics relating to the frequency domain

The most common form of digitizing data is to use a regular time based method. That is data is sampled at a constant rate specified as a number of samples/second. The Nyquist frequency, f_N, is defined such that f_N = SampleRate/2. As discussed elsewhere Shannon's Sampling Theorem tells us that if the signal we are sampling is band limited so that all the information is at frequencies less than f_N then we are alias free and have a valid digitised signal. Furthermore the theorem assures us that we have all the available information on the signal.

If we Fourier Analyze a signal, x(t), then we get its components expressed at frequencies measured in Hz. This is completely reversible. That is if we have x(t) we can get X(f). Similarly if we have X(f) we can get back to x(t). This is sometimes written in the form

$$x(t) \leftrightarrow X(f)$$

As a simple example consider a test signal composed of two sinewaves. The first one had an amplitude of 1.0, a frequency of 60Hz and a 0° phase. The second one had an amplitude of 0.5, a frequency of 180Hz and a 45° phase. In this example we will sample at 2048 samples/second and acquire 2 seconds of data. The choice of 2048 samples/second is to ensure it exactly matches an FFT size. This is not necessary but it avoids any discussion on end effects. A section of the signal is shown in Figure 1 below.

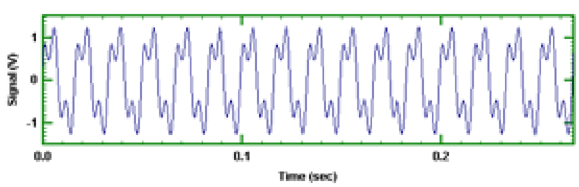


Figure 1 Time History of Two Sinewaves

If we FFT this composite signal then we get the modulus and phase plot as shown below.

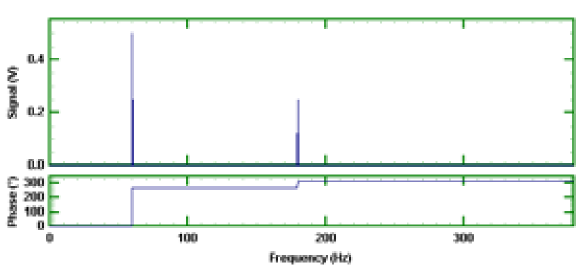


Figure 2 Standard FFT of Two Sinewaves

As expected the amplitudes are 0.5 and 0.25 respectively – recall that DATS for Windows gives half amplitudes. In the phase plot there is a 270° phase change at 60 Hz and a 45° phase change (270° to 315°) at 180 Hz. Clearly the 45° phase change is as expected but why the 270° change? Surely it should be 0° and 45° not 270° and 315° . The reason is because Fourier analysis uses cosines and sines with the cosine, not the sine, for the real part. A sine has a -90° or +270° shift relative to a cosine. In other words the basis is a cosine wave not a sine!

All of the above is fairly basic signal analysis. Now suppose we have a rotating shaft and we are measuring the vibrations of the shaft. The nature of the rotating items is that the vibrations occur at multiples and submultiples of the rotational speed. For example if the shaft is rotating at 3600rpm, which is 60 Hz, then we would expect to see responses at multiples of this frequency. These multiples are the orders (or harmonics in musical terms). First order is a frequency which is the same as the shaft rotational speed. In our example this is 60 Hz. The third order would be 3 * 60 = 180 Hz. The general relationship between the order, OR, the shaft speed, R, in rpm, and the frequency, f, in Hz is

$$f = OR*(R/60)$$

Why use orders? The reason is of course that the order remains constant with shaft speed; first order is always at the shaft speed; second order is always twice shaft speed and so on. This means we can step into the rotation and it is as if we were moving with the shaft. Instead of sampling at equal increments of time we sample at equal increments of rotation. This is called synchronous sampling; we have synchronised our sampling with the shaft rotational speed. Suppose we had a toothed wheel fixed to the shaft. Instead of a clock providing the command pulses to drive the analogue to digital converter then pulses from each gear tooth will give us equi angular or synchronous sampling at P samples/rev.

We now have data which is sampled in units of a fraction of a rev rather than as a fraction of a second. If we Fourier transform this data we again get a measurement as a function of a frequency type scale but now it is in increments of Orders not Hz. The result is that we now have a signal which gives the modulus and phase as a function of Orders rather than as a function of Hz.

We do not have to sample synchronously to get orders because we can use the relationship between frequency in Hz, f, order number, OR, and rotational speed, R. The procedure is then to FFT the time history, and by using the rotational speed to convert the frequency in Hz to a 'frequency' in Orders. This is in principle fine for a constant speed but if the speed is changing over the length of the FFT we have an incorrect result. Also it is unlikely that the frequencies in Hz will map exactly onto integer values in Orders. This means then grouping several order lines to form an rms value.

Thus in dealing with signals from rotating machinery synchronous sampling is preferable but regrettably converting to synchronous sampling is difficult in practice. It is impossible to sample synchronously with some data acquisition equipment, in particular those with sigma-delta type ADCs must sample at regular time steps. Successive approximation ADCs as used in the DATS-tetrad & DATS-hyper12 do not have this restriction. This however is not always of practical significance as usually it is difficult enough to get a reliable once/rev tacho pulse let alone N pulses per rev.

The solution is to use signal processing to digitally resample the data. Again we note the implication of Shannon's Sampling Theorem that if we sample at least twice as fast as the highest frequency present then we have all the information about the signal. With the correct signal processing algorithms we can then resample the initial equi speed time increment data into equi spaced angle increment data. We will not go into this theory and the relevant equations here except to note that the resampling is based on the (sin x)/x function, which is called appropriately a sinc function. This resampling algorithm can be used just to change the sampling rate from say 20000 samples/second to 44100 samples/second for sound replay. When being used for resampling to achieve equal angle it is clear that a once per rev tacho signal is also required. This provides the relationship between time and the total 'angle' travelled.

The DATS module ATOSYNC uses a once per rev tacho to convert a regular time series to a synchronous time series. To illustrate its use we will resample the mixed sinewave signal used earlier. A tacho signal which matched the 60 Hz component was used, that is first order will correspond to the 60 Hz signal. Thirty two points/rev were used. A section of the new synchronous signal is shown in Figure 3 below. It looks identical to the original regular time sampled data except that the x axis is now in terms of the total angular distance travelled in Revs.

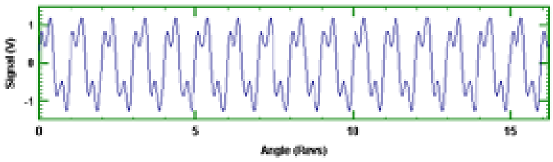


Figure 3 Two Sinewaves Synchronously Sampled

The FFT of the synchronous signal is shown below

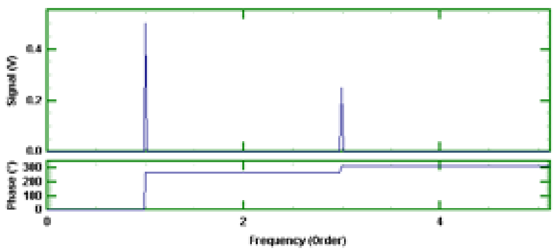


Figure 4 FFT of Synchronous Signal

This is identical, as expected, to the FFT of the original time history except that the x axis is now marked with units of Orders, not Hz. The two responses are at exactly one and three orders as expected. The mathematics has not changed, just the interpretation and our frame of reference.

As a further observation sampling is sometimes carried out as a function of distance. For example a vibrating beam could be measured at equi spaced points along the beam at one instant in time. The increment is then in steps of metres. If we Fourier transformed that signal then we would have a frequency axis in units of wavelength. This option however is not available in DATS.

The more complex signal shown below comes from a steady state run on a vehicle. There is clearly some beating going on. The vehicle speed was not constant so if one analyzed in the time domain the amplitudes would be 'smeared' over several frequencies.

Figure 6 below shows the Order analysis where it is quite clear that the energy is mostly at first order with some side bands and

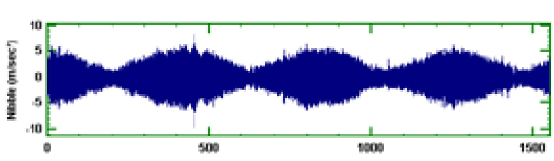


Figure 5 Synchronously Sampled Car Vibration

also some small contributions at the second and third orders.

An even more revealing analysis is to analyze a simple swept sinewave such as shown below (Figure 7).

If we analyze this as a standard time history we get the expected spectrum from 30 Hz to 100 Hz as shown in Figure 8.

Now if we synchronously sample the swept sine using itself as its own once/rev signal and frequency analyze then we get the synchronous signal as shown in Figure 9.

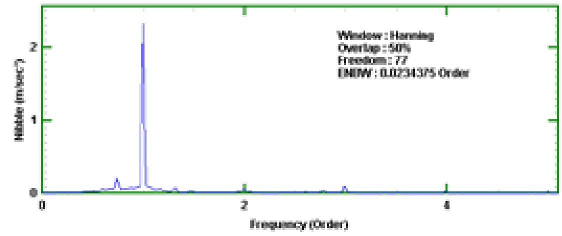


Figure 6 Order Analysis of Car Data

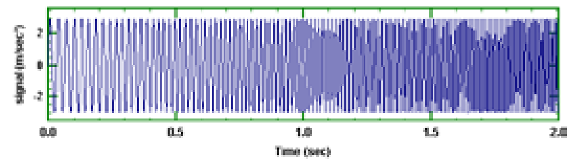


Figure 7 Swept Sine from 30 Hz to 100 Hz

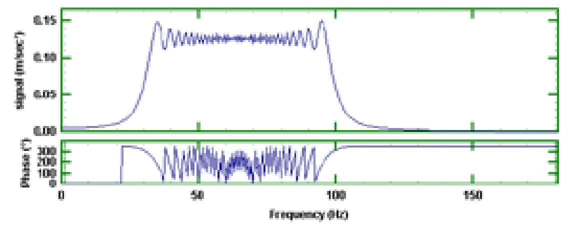


Figure 8 FFT of Swept Sine

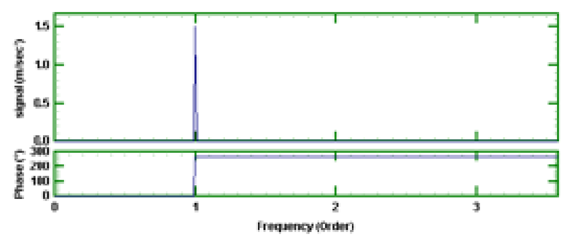


Figure 9 Synchronously Sampled Swept Sine

The entire signal is now concentrated entirely at the first order. The amplitude is the half amplitude and the phase jump is 270° . If we look at the synchronously sampled signal, which is shown in Figure 10, then the 'swept sinewave' is now just a 'constant sinewave'. We have a constant number of points per revolution and all evidence of the change in speed has gone

The software does however retain the time versus angle travelled data in another signal so that in practice no information is lost.

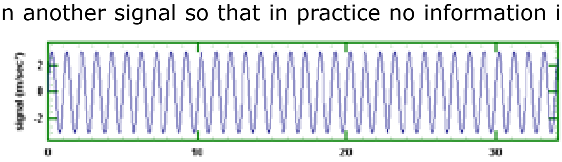
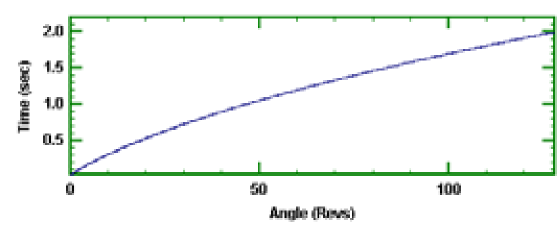


Figure 10 Order Analysis of Synchronous Swept Sine

This is shown below in Figure 11.



Because the rate of change of speed was a constant, then the time vs angle travelled curve is of the form time = k*sqrt(angle).

Is That Tone Significant?

Dr Colin Mercer

How the Prominence Ration can help us see aurally prominent tones in a signal

The Prominence Ratio is a technique designed to see if there are any aurally prominent tones in a signal. Primarily, the prominence ratio is applicable where we have a noise source with a few tones and we need an objective measure to assess if the tones are "prominent". That is, to assess whether the tones are likely to be heard.

It is formally defined in ANSI S1.13-2005 "Measurement of Sound Pressure Levels in Air" for general use and in ECMA 74 9th Edition December 2005, "Measurement of Airborne Noise Emitted by Information Technology and Telecommunications Equipment" specifically for noise from computers and associated peripherals.

The considered view is that our hearing appears to assess sounds in frequency bandwidths. That is, the ear behaves like a bank of filters where, in general, the bandwidth of each filter increases with frequency. What this means is that two or more tones, spaced such that they are all in the same band, sound as if they are just one tone within that band; conversely two tones separated by more than a critical bandwidth are perceived as being separate. In a simplified form then the ear appears to "analyse" the sound into the "energy" in each band. This then forms the basis of the prominence ratio. We use the term "energy" as the measure is based on the square of the sound pressures, X_p, in three adjacent bands. If we denote these by X_L, X_M and X_U for the Lower, Middle and Upper bands respectively then the Prominence Ratio for the middle band, PR_M, is given by

$$PR_M = \frac{X_M}{(X_L+X_U)/2} dB$$

In the world of psychoacoustics there is debate over the width of each critical band. There is one measure of critical bandwidth in ERBs, associated with Moore, and another in Bark, associated with Zwicker. Figure 1 below shows these bandwidths plotted versus frequency together with some 1/Nth octave bandwidths.

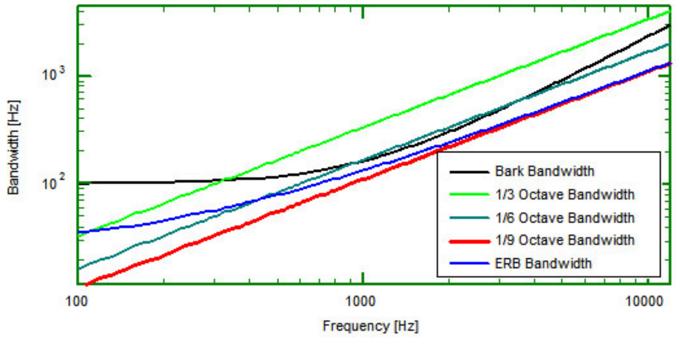


Figure 1: Bandwidths v frequencies

Above 1kHz the ERB bandwidth is very similar to 1/9th octave and the Bark bandwidth is approximately like 1/6th octave.

ANSI S1.13 and ECMA 74 use the Bark based Critical Bandwidths. However, when investigating the possibility of several close narrow bands it is useful to be able to select one of the 1/Nth octave bandwidths.

A tone is classed as prominent if the Prominence Ratio exceeds the following limits

$$PR(f) \geq 9.0 + \log(1000/f) \quad \text{for } f < 1000\text{Hz}$$
$$PR(f) \geq 9.0 \quad \text{for } f \geq 1000\text{Hz}$$

The shape of the Prominence Ratio curve is not initially what one might expect. Consider first a standard white noise signal. In Figure 2 below, the first graph shows a section of the signal and the second graph is the corresponding Prominence Ratio.

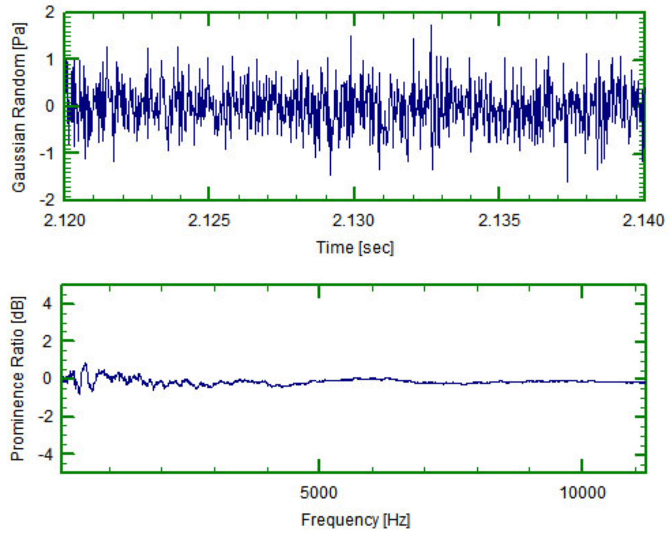


Figure 2: White noise and corresponding prominence ratio

As expected the prominence ratio is essentially zero. However, if a small sinewave at 2kHz is added to the time history then we see results like those in Figure 3 below.

There is no discernible difference in the time signals and as expected the prominence ratio shows a peak at 2kHz.

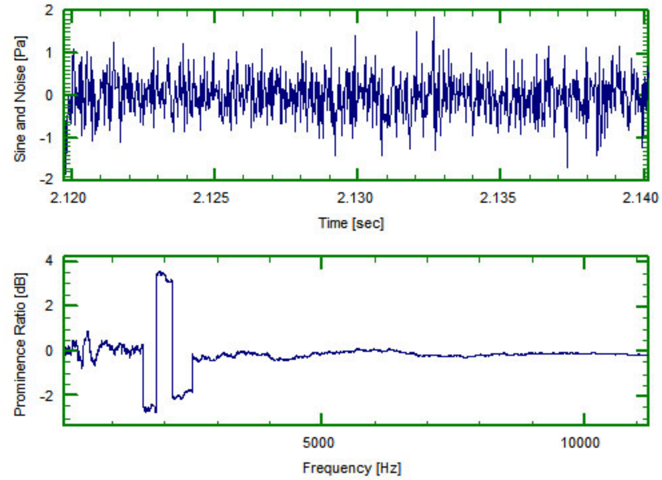


Figure 3: White noise plus 2kHz sinewave with corresponding prominence ratio

But it also shows significant dips before and after the peak. These 'dips' are a direct result of the way the Prominence Ratio is defined. Recall our equation for PR_M above. When the Prominence Ratio is being computed over the region from about 1600 to 1850Hz then the peak at 2kHz will be in the upper band, thus causing the Prominence Ratio to be lowered. A similar situation occurs over the 2150 to 2550Hz region except that now the 2KHz tone is in the lower band.

The width of the region of high prominence ratio is approximately 300Hz as expected as that is the width of a Bark based critical band at that frequency.

A more realistic signal is shown in Figure 4 below.

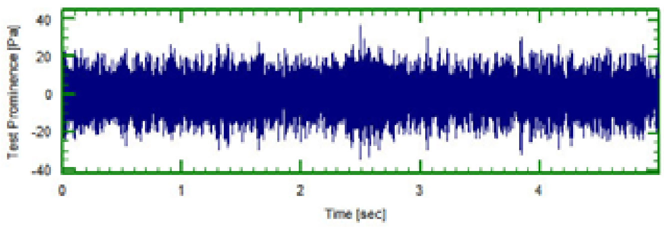


Figure 4: A "real" signal

This looks like yet another random signal. This time, however, we have what appears to be three regions of prominence as you can see in Figure 5.

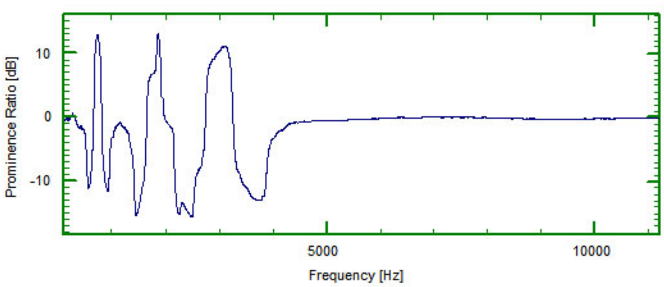


Figure 5: Prominence ratio of signal from Figure 4

The multiplicity of the negative dips can be annoying! As an option a low limit threshold is useful as illustrated below. It is also clearly useful if the software determines the number of tones that exceed the ANSI/ ECMA criteria and, of course, makes available these values together with an estimate of the frequency at which they occur. The DATS analysis software stores these with the resulting signals. They can be seen as annotations on the graph in Figure 6.

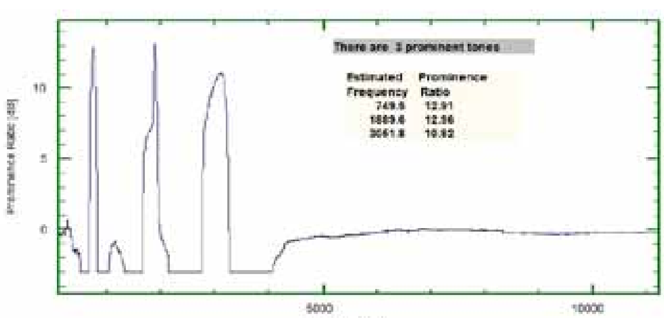


Figure 6: Prominence ratio with dips removed and prominent tones annotated

The prominence ratio of the middle "peak" has a distinct "lump" on the leading edge at around the 7 dB level. If we analyze using a narrower bandwidth, in this case 1/9th octave Prominence Ratio, then the figure below actually shows four peaks. Because we are using a narrower bandwidth then the levels are a little lower. Also shown below is the frequency spectrum of the signal, which again shows four distinct spikes.

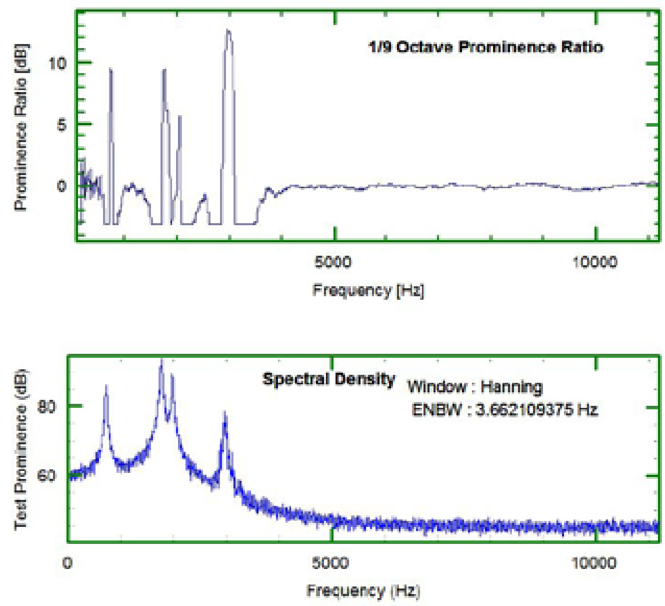


Figure 7: Prominence ration using 1/9th octave & corresponding frequency spectrum

Although the Prominence Ratio was defined for steady state signals it is often useful to see it as a frequency time analysis.

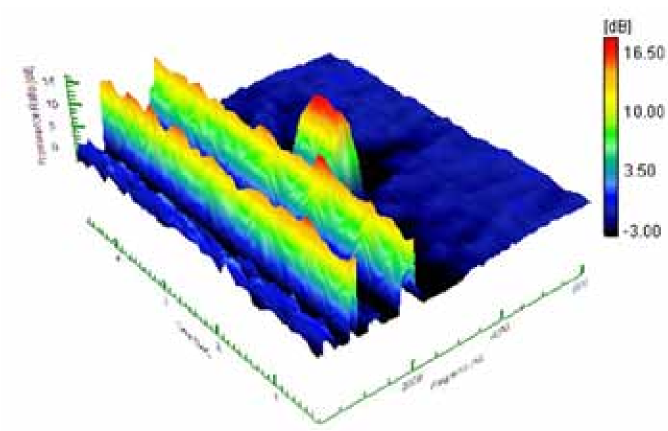


Figure 8: Prominence ratio plotted against time

This shows that the tone at around 3kHz is actually a tone burst occurring approximately in the centre of the signal.

Fatigue & Durability Testing - How Do I Do It?

James Wren

A look at the test & measurement process for fatigue & durability testing

The following application note describes the test and measurement process for the fatigue testing and development cycle of an automotive suspension component, specifically a tie rod. The component had been known to fail at various intervals. An estimate of the predicted fatigue life of the component was required in order to assess the feasibility of its continued use and to see if a design change was required. The component under test is shown in Figure 1. The testing was carried out by a major automotive manufacturer. Strain gauges were used to monitor the strain levels.

Figure 1: Vehicle suspension under test

First, the vehicle was instrumented and data acquired whilst the vehicle was on a test track. The vehicle under test was then mounted on a four-post shaker rig that simulates actual road conditions. Particular sections of the actual test run from the test track could be replicated for much longer periods than otherwise would be possible, additionally this could all be done in the controlled conditions of the test laboratory.

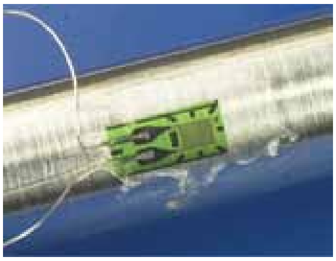


Figure 2: Strain gauge mounted on component under test

This article will focus on the capture of the strain data and the processing of such data using 'Stress Life' and 'Fatigue Life Prediction' methods to predict the expected life of the component.

Initially, the vehicle was instrumented with strain gauges using a Prosig P8000 portable data acquisition system. The vehicle was driven around a test track and data from the strain gauges acquired with the P8000 data acquisition unit. The data capture and signal processing procedure can be seen and followed from start to finish.

Figure 2 shows an example of a strain gauge attached to the component under test by adhesive. In this case the gauge is a two wire device.

After launching the Prosig Data Acquisition software various setup fields must be filled in such as signal names, signal types and so on. For the purposes of clarity in this application note only one strain gauge is being used and it is operating in a quarter bridge completion configuration. It is a 120 Ohm gauge and an excitation of 10VDC has been selected to be applied as shown in Figure 3.

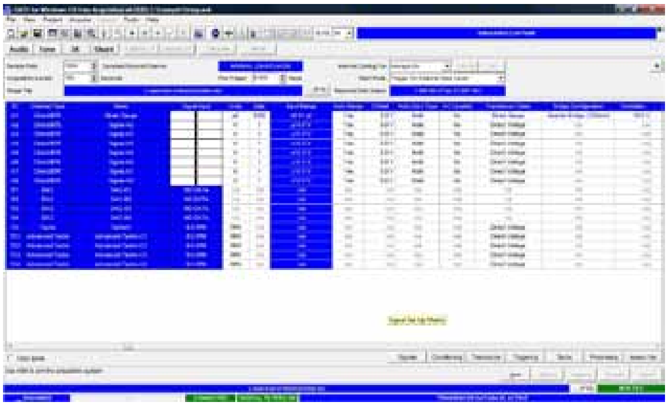


Figure 3: Acquisition setup screen

Next the Prosig P8000 and associated acquisition software are used to capture the strain data from the vehicle as it driven around the test track. The real time display, Figure 4, shows the micro strain and the oscillatory nature of the data as it is being captured.

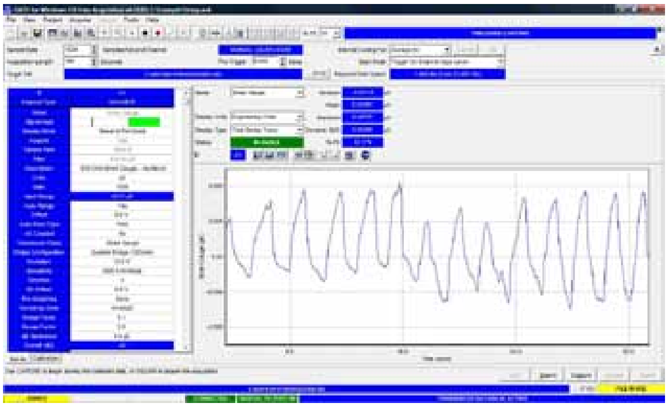


Figure 4: Real time display of data

On completion of the data capture stage, the next step was to return to the testing laboratory and use the vehicle shaker test rig. The DATS signal processing software was used to identify significant frequencies and amplitudes of strain data that the vehicle had been subject to on the test track. The analogue output facility of the P8000 was then used to 'replay' the signals into the shaker to 'mimic' particular sections of the track. This allowed the test engineers to control the amount of time the component under test was subjected to specific frequencies or amplitudes. During any one hour period on the test track the

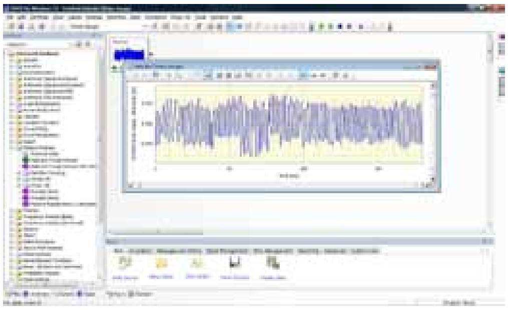


Figure 5: Strain v time

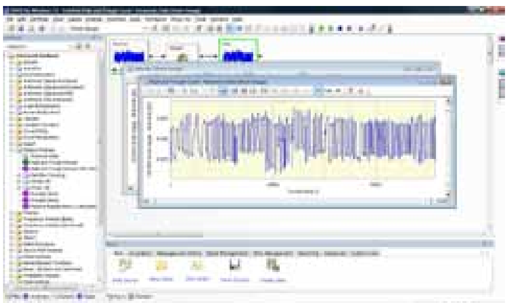


Figure 6: Peak & trough data

component might be subjected to a particular excitation pattern for only a few seconds. But, by capturing the data while the vehicle is moving on the track and then using the shaker rig, it is possible to subject the component to a period of accelerated saturation testing of particular frequencies and amplitude characteristics.

For this application note the suspension component was excited for 180 seconds. The excitation was captured with the P8000 and opened in the DATS signal processing software (Figure 5). At this stage the strain data is stored and displayed with respect to time. The degree of micro strain the component was subjected to can be seen over the entire capture period.

In order to begin the fatigue life prediction it is first necessary to analyze the peak and trough content of the captured data. This

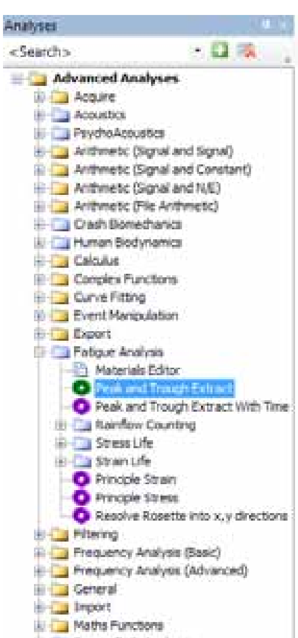


Figure 7: Peak & Trough analysis section

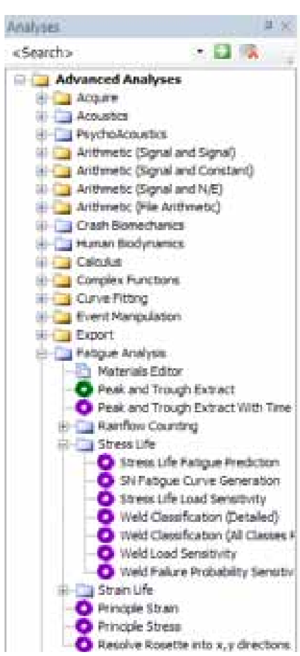


Figure 8: S-N curve analysis selection

is easily achieved using the DATS Fatigue Life Analysis software tool kit.

The peak and trough data shown in Figure 6 was produced by selecting the relevant analysis module as highlighted in Figure 7.

Next, it was required to generate an S-N curve for the component as depicted in Figures 7 and 8. It is important to note that this is not an S-N curve for the material used in fabricating the component, but the S-N curve for the component itself.

At this stage limited data is available for predicting failure rates and moreover this information is rough and sporadic. This, however, will not be used to produce the S-N curve at this point.

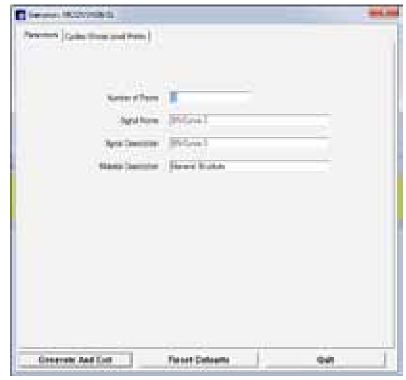


Figure 9: S-N curve generation parameters

Figure 10: S-N curve values

The failure data will be used later to refine the S-N curve.

Using the DATS Fatigue Life Analysis module for S-N curve generation (Figures 9 & 10) it is possible to produce an S-N curve.

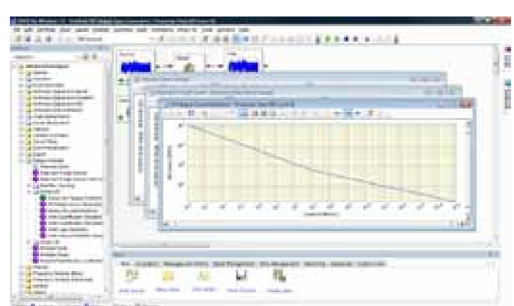


Figure 11: Generated S-N curve

S-N curves are by their nature very simple, they can usually be approximated by two intersecting straight lines on a graph of log stress verses log cycles. In this case three points are used to create the curve.

A set number of cycles to failure and stress levels are required. As mentioned previously the S-N curve will be refined later.

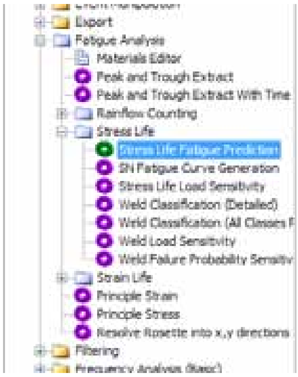


Figure 12: Stress Life Prediction analysis selection

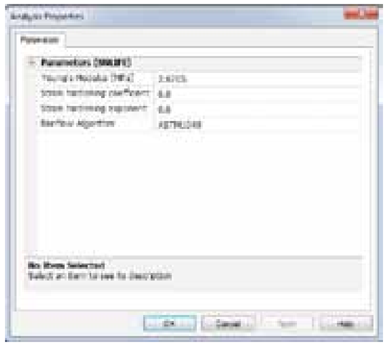


Figure 13: Stress Life Prediction parameters

At this stage, the values for the 'Weld Classification' are used. These are chosen arbitrarily as it is a known curve that closely follows that of the material under test. The generated S-N curve is then created as shown in Figure 11.

With the peak trough data and S-N curve it is possible to complete a fatigue life prediction, using the 'Stress Life Fatigue Prediction' analysis module (Figure 12). To complete this analysis both the S-N curve and the initial peak and trough data are required.

When the analysis module begins it prompts the user for certain values (Figure 13).

The fatigue life prediction analysis module requires a Young's modulus for the material, in this case 2.07×10^5MPa. A rain flow algorithm must also be selected, in this case the ASTM1094. (American Society for Testing and Materials, Revision 1985).

The conversion from Micro Strain to Stress uses the following formula. The micro strain values, ε, are translated into stress, S, by solving

$$\epsilon = \frac{S}{E} + \left(\frac{S}{K'}\right)^{1/n'}$$

Where

E is Young's Modulus
K' is Strain Hardening Coefficient
n' is Strain Hardening Exponent
If K' or n' or both are zero then the module uses S = εE

This analysis takes two input datasets: the peak and trough count and the S-N curve. The resultant 'Stress Life Fatigue Prediction' damage curve is shown in Figure 14, with a fatigue life prediction of 3.4×10^{20} seconds.

Figure 14: Stress Life Prediction data

The Fatigue Life Prediction analysis also calculates more detailed results stored as Named Elements. These are shown highlighted in blue in Figure 15. They include Damage, Duration of original time sample, Number of cycles and so on.

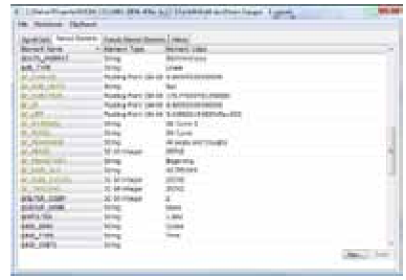


Figure 15: Stress Life Prediction data (Named Elements)

To summarize thus far, it has been possible to complete a fatigue life prediction from a sample of strain data taken over a specific time period.

This has given a predicted life of 3.4×10^{20} seconds.

As discussed earlier, the S-N curve was not a refined curve and was almost arbitrary in its construction. This could potentially lead to errors. Therefore at this stage the S-N curve must be refined to allow recalculation of more accurate results and thus remove any potential errors.

The component in question has been reported to fail in the field after various time periods, hence the reason for the trial. Although the stress and strain levels are not known for these failures the time to failure is important. Because it is possible to apply the expected strain level for general use to the component for the known period of time, it is, therefore, possible to extrapolate the stress levels. Note, the stress levels and cycles to failure are not known for these situations. Only the time to failure is known.

The automotive component was also tested to failure, with failures occurring at the following intervals. As these failures were under controlled test environments they can be considered to be more accurate than the prediction result discussed previously.

Time to failure 6.48×10^5 seconds with a stress of 0.003010 MPa

Time to failure 6.75×10^7 seconds with a stress of 0.000165 MPa

The following have known times to failure, but with unknown strain levels. For these cases the known failure stress levels can be used, in this case 0.000165 MPa is chosen.

1.52×10^7 seconds

7.78×10^7 seconds

2.64×10^6 seconds

The cycles of the vehicle suspension component, importantly not the cycles of material, were less that 2Hz. However, the material cycles the component was subjected to were 3253 in a 180 second snap shot.

Therefore, it is possible to calculate the number of material cycles for the known failure times and then to accurately adjust our initial S-N curve.

It is also possible to calculate cycles to failure for the situations where the known failure times do not have strain information. This can be achieved because it is possible from experimental testing to deduce what the expected or average use and therefore strains will be.

Known or unknown strain	Time to failure (seconds)	Cycles per 180 seconds	Cycles to failure
Known	6.48×10^5	3253	11710800
Known	6.75×10^7	3253	1219875000
Unknown	1.52×10^7	3253	274697777
Unknown	7.78×10^7	3253	1406018888
Unknown	2.64×10^6	3253	47710666

It is now possible to refine the original S-N curve (Figure 16)

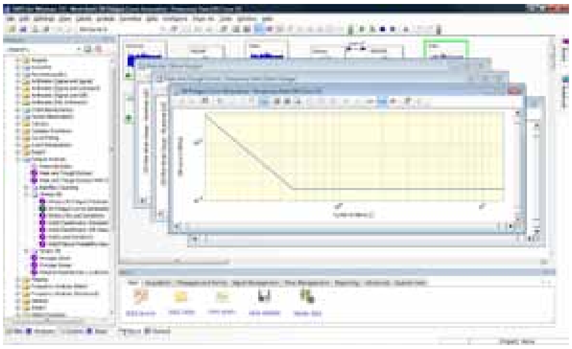


Figure 16: Recalculated S-N curve

Figure 17: Recalculated life prediction

with the 5 pairs of values calculated,

0.003010 MPa and 11710800 cycles to failure

0.000165 MPa and 1219875000 cycles to failure

0.000165 MPa and 274697777 cycles to failure

0.000165 MPa and 1406018888 cycles to failure

0.000165 MPa and 47710666 cycles to failure

Therefore it is possible to extrapolate what the S-N curve could have been and thus re-process the results using the automatic reprocessing features of DATS as shown in Figure 16.

The result of the re-processed fatigue life prediction is 4.40×10^{6} seconds.

The conclusion is that after approximately 51 days of use at the expected level of 10 hours use per day this component could be expected to fail. Clearly this is a fragile component that is likely to fail in an unacceptably short amount of time for an automotive application and both a technical design change and further testing are required.

It is evident that the more testing to failure that is carried out the more accurate the final life prediction will be.

[Note: This article has been reduced in complexity compared with the original test report and uses deliberately modified initial strain values]

Sampling & Filtering for ISO8041

Dr Mike Donegan

Find out about the sampling, filtering, weighting and analysis required for ISO8041 compliance

Various human body vibration measurements and assessments are defined in the ISO2631, ISO5349, ISO6954 standards and also in the EEC vibration directive 2002/44/EC. These assessments are based on the analysis of acceleration data that has been weighted (filtered) with appropriate vibration weighting values. The data must be in units of m/sec².

The vibration weighting values are defined in ISO8041:2005. Basically they are a set of filters that must be applied to the acceleration data in order to make evaluations and assessments on the effects of vibration on humans. They are defined as a set of weighting classes. Each class is associated with particular measurement directions, measurement positions and assessment type.

A selection of defined weighting classes are shown in Table 1. The directions are orientated such that the x direction is positive back to chest, the y direction is positive right side to left side and the z direction is positive from feet to head.

Weight Class	Description
W_b	Weighting for vertical whole body vibration (seat z direction): ISO2631-4
W_c	Weighting for vertical whole body vibration (seat back z direction): ISO2631-4
W_d	Weighting for horizontal whole body vibration (seat surface x and y directions): ISO2631-1
W_e	Weighting for rotational whole body vibration: ISO2631-1
W_f	Weighting for vertical whole body vibration (z direction motion sickness): ISO2631-1
W_h	Weighting for hand arm vibration (all directions): ISO5349-1

Weight Class	Description
W_k	Weighting for vertical whole body vibration (z direction): ISO2631-1
W_j	Weighting for vertical head vibration (x direction recumbent): ISO2631-1
W_m	Weighting for whole body vibration in buildings (all directions): ISO2631-2 and ship vibration measurements (all directions): ISO6954

Table 1

The vibration weighting is applied by initially pre-filtering the data with a band pass filter between a lower frequency, f_1 and an upper frequency, f_2 as defined in Table 2. The appropriate weighting functions are then applied to the resulting band filtered data.

Weight Class	Lower Frequency f_1 (Hz)	Upper Frequency f_2 (Hz)	ISO8041 Compliant Minimum Sample Rate (samples/sec)
W_b	0.4	100	900
W_c	0.4	100	900
W_d	0.4	100	900
W_e	0.4	100	900
W_h	6.3	1259	11331
W_f	0.08	0.63	6
W_i	0.4	100	900
W_k	0.4	100	900
W_j	0.79	100	900

Table 2

Strict ISO8041 Compliance

ISO8041:2005 specifies that the acceleration data must be sampled at an adequate rate for the appropriate vibration weight class filter to be applied. In order to achieve strict ISO8041:2005 compliance the minimum adequate sample rate must be at least nine times the relevant low pass pre-filter cut off frequency, f_2. As shown in *Table 2* the minimum ISO8041 compliant sample rate is 900 samples/second ($9*f_2 = 9*100$) for all vibration weight classes with the exception of class W_f where it is 6 samples/second (9*0.63) and class W_h where it is 11331 samples/second (9*1259).

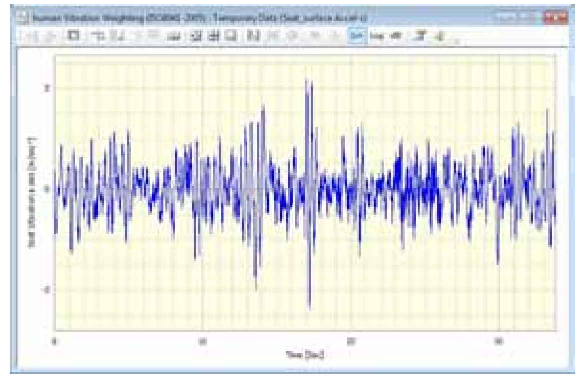


Figure 2: X direction seat surface acceleration after Wd filter

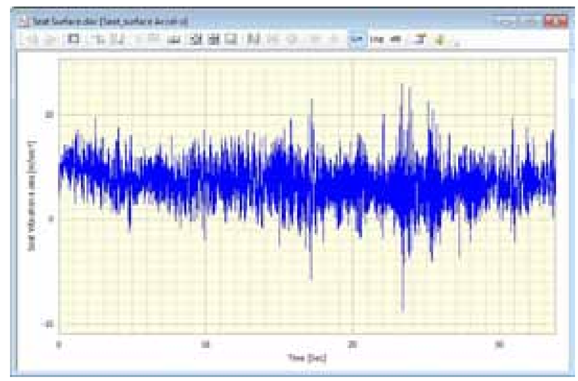


Figure 1: X direction seat surface acceleration

The DATS software provides a particular function to perform the vibration weighting for a specified vibration weight class. In addition, DATS also provides a package to generate vibration assessment reports compliant with ISO2631-1, ISO2631-2, ISO2631-4, ISO5349 and ISO6954.

An example seat surface acceleration is shown in Figures 1 and 2. Figure 1 shows the raw acceleration data and Figure 2 shows the acceleration after the appropriate ISO8041 compliant vibration weighting has been applied. In this example the data is seat surface data in the X direction so the appropriate vibration weight class is the Wd filter.

Measure Acceleration, Velocity or Displacement?

Dr Colin Mercer

We can convert between them, but what is the best quantity to measure?

When using vibration data, especially in conjunction with modelling systems, the measured data is often needed as an acceleration, as a velocity and as a displacement. Sometimes different analysis groups require the measured signals in a different form. Clearly, it is impractical to measure all three at once even if we could. Physically it is nigh on impossible to put three different types of transducer in the same place.

Accelerometers are available in all types and sizes and there is a very large choice. Some types will measure down to DC (0Hz), others handle high shock loading and so on.

True **velocimeters** are quite rare, but they do exist. One interesting class based on a coil and magnet scheme is self powered.

Direct **displacement measurement** is not uncommon. Some use strain gauges, but many others use a capacitive effect or induced radio frequency mechanism to measure displacement directly. The capacitive and inductive types have the advantage that they are non-contacting probes and do not affect the local mass.

But in any case it doesn't matter, because if we measure either acceleration, velocity or displacement then it is simple mathematics to convert between them by a judicious use of integration or differentiation as illustrated below.

Measured Signal Type	Operation	Result
Displacement	differentiate	Velocity
Displacement	double differentiate	Acceleration
Velocity	differentiate	Acceleration
Velocity	integrate	Displacement
Acceleration	integrate	Velocity
Acceleration	double integrate	Displacement

So now let us look at this with a classical sine wave signal and see the effects of either differentiating or integrating it. To avoid other side effects the example uses a 96Hz sinewave of unit amplitude with 32768 samples generated at 8192 samples/second. It is useful to look at these as time histories and as functions of frequency. That is, the original generated sinewave was processed using a DATS worksheet as illustrated in Figure 1.

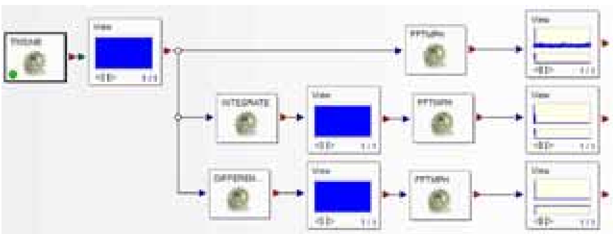


Figure 1: Worksheet calculating frequency spectra

Looking at a section of the wave forms, we have a classical result as shown in Figure 2.

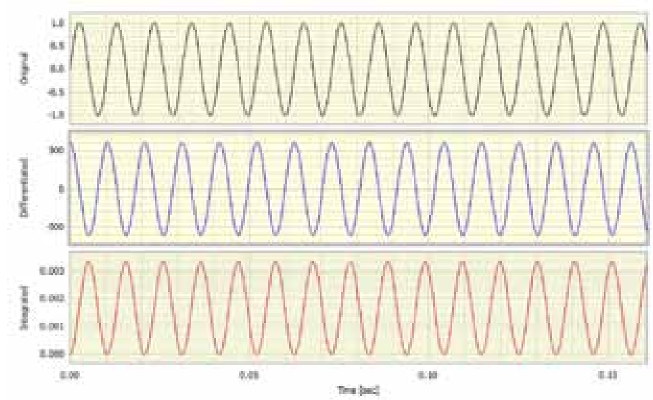


Figure 2: Original, differentiated and integrated time histories

In mathematical terms if $y(t) = A \sin(2\pi ft)$ then the differential is $\{2\pi Af\cos(2\pi ft)\}$ and the integral is $\{-(A/2\pi f)\cos(2\pi ft)\} + C$ where C is the so called 'constant of integration'. In both cases there is a phase shift of 90° which turns the sine into a cosine. The differential is multiplied by $2\pi f$. The integral is divided by $2\pi f$, is also negated and has had an offset added to it, which in this case is half the resultant amplitude, resulting in the integrated signal being entirely positive. If, for example, the original signal had represented an acceleration then the integrated signal is a velocity, and clearly we would not expect that to be entirely positive. This integration constant is an artefact of the standard integration methods.

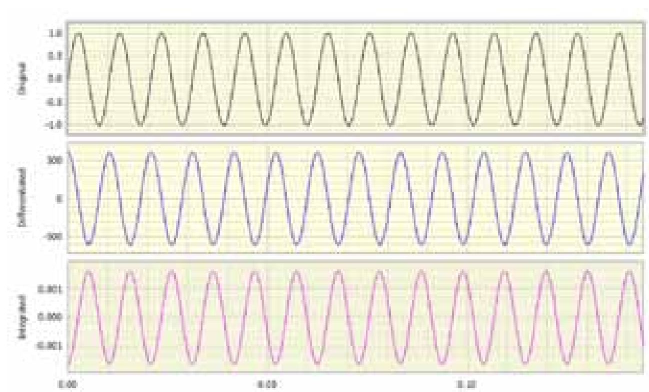


Figure 3: Adjusted to zero mean

For the mathematically inclined, it is the result of carrying out of what is usually referred to as an *indefinite integration*. The solution is quite simple. After doing a standard time based integration then we should automatically reduce the result to have a zero mean value. That is, we ensure there is no residual DC offset. The calculation process was modified to include that action and the result is shown in Figure 3. Note how the integrated signal is positive and negative as we would expect.

It is also interesting to look at the Fourier Transforms of the three signals. These are shown in Figure 4 in modulus

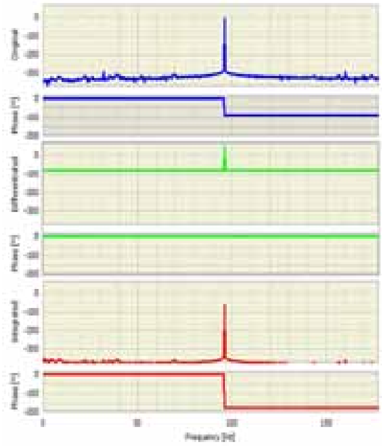


Figure 4: Fourier transforms

(amplitude) and phase form. The moduli are shown in dBs and the phase is in degrees.

Looking first at the phase, the original sinewave has a phase shift of -90°. This is entirely as expected because the basis of the FFT is actually a cosine. The differentiated signal has a zero phase change as it is now a pure cosine. The integrated signal has a 180 degree phase change, denoting it is a negative cosine.

The dynamic range of the original signal is well over 300 dB which is not surprising as it was generated in software in double precision. This is approximately equivalent to a 50 bit accuracy ADC! The integrated signal shows a similar dynamic range but, what may appear as surprising initially, the differentiated signal has lost half of the dynamic range. We will return to this point later.

Small DC offsets are not uncommon in many data acquisition systems. Some offer AC coupling (highpass filtering) to minimise any offset. How would this affect the resultant signals? To illustrate this point a small DC offset of 0.01 (1% of the amplitude) was added to the original sinewave signal and the results are shown below.

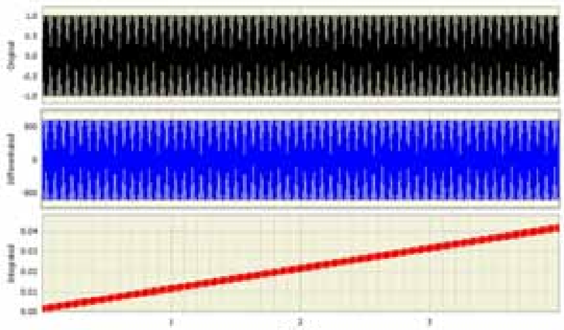


Figure 5: Result of integrating with small DC offset

The effect on the original is essentially not noticeable. Similarly the differentiated signal is unchanged as would be expected. But the effect on the integrated signal is quite dramatic. The small DC offset has produced a huge trend. We have integrated a 0.01 constant over 4 seconds, which gives an accumulated 'drift' of 0.04. The underlying integrated signal is still evident and is superimposed on this drift.

How do we avoid this? Simply reduce the input to have a zero mean, which is often called *normalizing*.

Note, that at this juncture, we have not had to do anything to the initial signal when we are differentiating, but we have had to remove any DC offset before integration to prevent the 'drift'

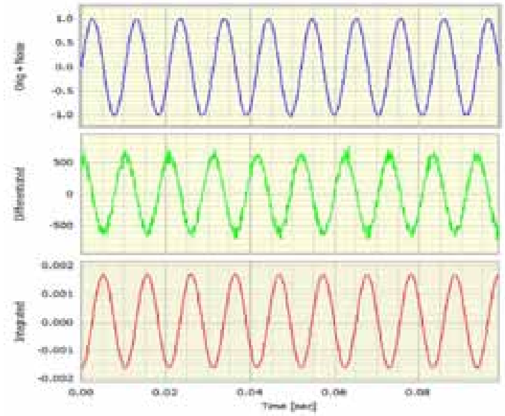


Figure 6: Addition of a small amount of noise

and also remove the DC offset from the integrated signal to eliminate the constant of integration. So at this stage one might be tempted to conclude that using a differentiating scheme might the best way forward. However, when we add noise the situation changes.

As a start, a small random noise signal was added to the original sinewave.

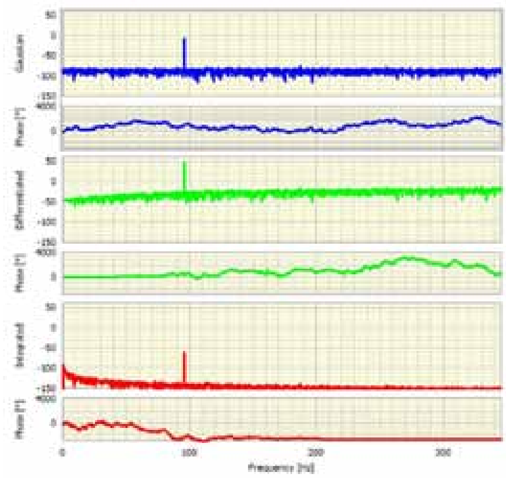


Figure 7: Spectra from noisy signals

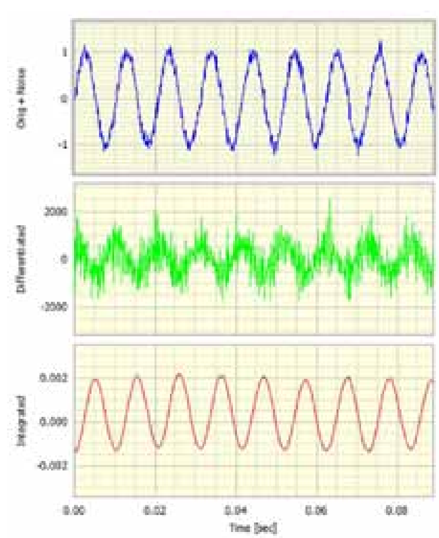


Figure 8: Time series with more noise added

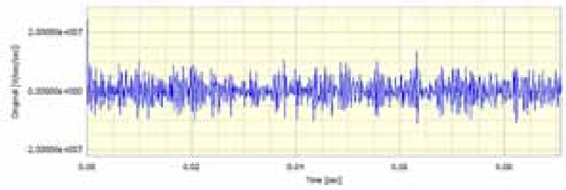


Figure 9: Noisy signal differentiated twice

The noise is not discernible to the eye on the original signal, but the differentiated signal has become very noisy. The integrated signal remains smooth. We can however identify the dominant frequency quite well.

If one examines the phase of the noisy signals, one can see it is now all over the place and essentially no longer any value. Automatic phase unwrap was used, if the phase had been displayed over a 360° range it would have totally filled the phase graph area.

The dynamic range of the original signal with added noise is around 90dB, with the differentiated and integrated signals having a similar range. That is, the added noise has dominated the range.

One other aspect to notice is that the background level of the noise on the integrated signal rises at the lower frequencies. This is known as 1/f noise (one over f noise). This sets an effective lower frequency limit below which integration is no longer viable.

To emphasise the challenge of noise the next example has a very much larger noise content.

Here the noise on the original signal is evident. The differentiated signal is effectively useless, but the integrated signal is relatively clean. To really illustrate the point, the noisy sinewave was differentiated twice. The result is shown below. All trace of the original sinewave seems to have gone or, rather, has been lost in the noise.

The conclusion is now clear. If there are no special circumstances, then experience suggests it is best to measure vibration with an accelerometer. However, care is required to remove the very low frequencies if any integration to velocity or displacement is needed.

As a final point, why should differentiation be much noisier than integration? The answer is that differentiation is a subtraction process and at its very basic level we take the difference between two successive values, and then divide by the time between samples. The two adjacent data points are often quite similar in size. Hence the difference is small and will be less accurate, then we divide by what often is a small time difference and this tends to amplify any errors. Integration on the other hand is addition. As any broadband noise tends to be successively, differently-signed then the noise cancels out.

This article, of course, does not tell the whole story, but it provides a very simple guide to good practice.

How to Calculate a Resultant Vector

Dr Mike Donegan

Learn about scalars, vectors and resultants

We can distinguish between quantities which have magnitude only and those which have magnitude and are also associated with a direction in space. The former are called scalars, for example, mass and temperature. The latter are called vectors, for example, acceleration, velocity and displacement.

In this article a vector is represented by **bold face type**. The magnitude of any vector, **r**, (also called the modulus of **r**) is denoted by *r*. The magnitude is a measurement of the size of the vector. The direction component indicates the vector is directed from one location to another. Scalars can be simply added together but vector addition must take into account the directions of the vectors.

Multiple vectors may be added together to produce a resultant vector. This resultant is a single vector whose effect is equivalent to the net combined effect of the set of vectors that were added together.

The use of a frame of reference allows us to describe the location of a point in space in relation to other points. The simplest frame of reference is the rectangular Cartesian coordinate system. It consists of three mutually perpendicular (triaxial x, y and z) straight lines intersecting at a point O that we call the origin. The lines Ox, Oy and Oz are called the x-axis, y-axis and z-axis respectively.

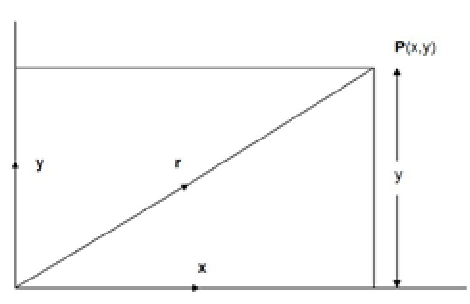


Figure 1

For convenience consider a two dimensional x,y coordinate system with an x-axis and a y-axis. In this configuration any point P with respect to the origin can be related to these axes by the numbers x and y as shown in Figure 1. These numbers are called the coordinates of point P and represent the perpendicular distances of point P from the axes.

If these two measurements represent vector quantities, for example displacement x and y, measured in the x and y directions respectively then we can use vector addition to combine them into a single resultant vector r as shown in Figure 1. In vector terms

$$\mathbf{r} = \mathbf{x} + \mathbf{y}$$

Any vector can be written as $\mathbf{r} = (\mathbf{r}/r)*r$ where $(\mathbf{r}/r)$ is a unit vector in the same direction as r. A unit vector is simply a vector with unit magnitude. By convention we assign three unit vectors **i**, **j** and **k** in the directions x, y and z respectively. So we can write

$$\mathbf{r} = \mathbf{x} + \mathbf{y} = x\mathbf{i} + y\mathbf{j}$$

where x is the magnitude of vector **x** and y is the magnitude of vector **y**.

Sometimes we are only interested in the magnitude or size of the resultant vector. Looking at Figure 1 we can use Pythagoras' Theorem to calculate the magnitude of vector **r** as

$$r^2 = x^2 + y^2$$

In vector terms, the scalar product **a**.**b** (also known as the dot product) of two vectors **a** and **b** is defined as the product of the magnitudes a and b and the cosine of the angle between vectors **a** and **b**. Therefore,

$$\mathbf{r}.\mathbf{r} = rr\cos(0) = r^2 = (x\mathbf{i} + y\mathbf{j}).(x\mathbf{i} + y\mathbf{j}) = x^2\mathbf{i}.\mathbf{i} + y^2\mathbf{j}.\mathbf{j} + 2xy\mathbf{i}.\mathbf{j}$$

By definition unit vectors have unity magnitude so $\mathbf{i}.\mathbf{i} = 1 * 1 * \cos(0) = 1$ and $\mathbf{i}.\mathbf{j} = 1 * 1 * \cos(90) = 0$. Substituting these values we come to the same formula

$$r^2 = x^2 + y^2$$

The modulus or magnitude, r, of the resultant vector **r** at point P with coordinates x and y is then given by

$$r = (x^2 + y^2)^{0.5}$$

This can be extended to a tri-axial (x,y,z) configuration. For example, if we have three measurements x, y and z representing accelerations measured by a tri-axial accelerometer in the x, y and z directions respectively then the resultant vector **r** is given by

$$\mathbf{r} = x\mathbf{i} + y\mathbf{j} + z\mathbf{k}$$

where **i**, **j** and **k** are unit vectors in the x, y and z directions.

The magnitude, r, of the resultant vector is then the net acceleration and is given by

$$r = (x^2 + y^2 + z^2)^{0.5}$$

There is a particular module in the DATS software that takes a tri-axial group of signals (three signals) and generates the resultant magnitude as shown below. In this example x, y and z accelerations were captured and analyzed to produce the magnitude of the resultant net acceleration.

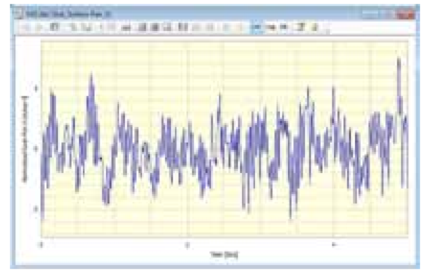


Figure 2: X direction acceleration

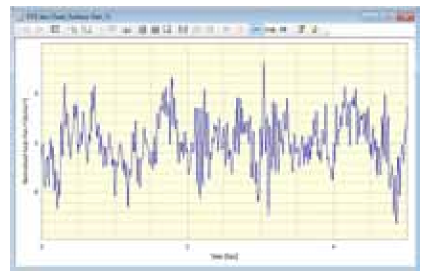


Figure 3: Y direction acceleration

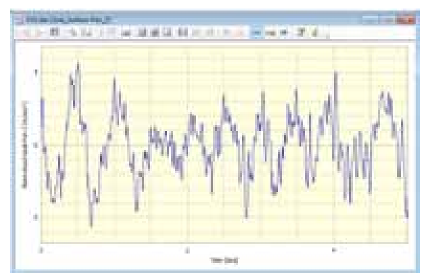


Figure 4: Z direction acceleration

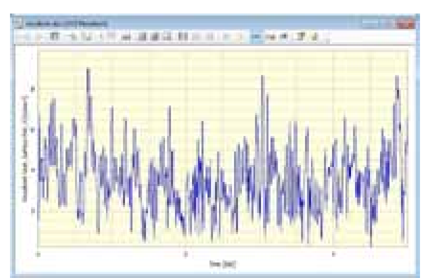


Figure 5: Magnitude of the X, Y, Z resultant acceleration

Cross Correlation Function

Dr Colin Mercer

An example of using the cross correlation function as a location algorithm

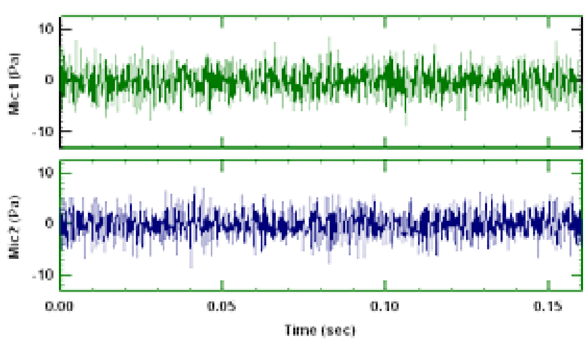
To illustrate the use of the cross correlation function, a source location example is shown below. For this it is assumed that there is a noise source at some unknown position between 2 microphones. A cross correlation technique and a transfer function like approach were used to determine the location.

To simulate the noise a broad band Gaussian signal was bandpass filtered from 500 to 1500Hz. This random signal, s(t), was generated at 10000 samples/second. Two delayed signals, p1(t) and p2(t), were then formed. Assuming the speed of sound in air is 1000ft/second then p1(t) was formed from s(t) with a 25 msec delay by ignoring the first 250 values. Similarly p2(t) was formed with 14msec delay by ignoring the first 140 values. In order to represent dispersion and other specific path effects, two unrelated Gaussian broad band signals, n1(t) and n2(t), were also generated, each with approximately 20% of the overall energy of the original signal. The microphone simulation signals, x1(t) and x2(t), were then formed from

$$x_1(t) = p_1(t) + n_1(t)$$
$$x_2(t) = p_2(t) + n_2(t)$$

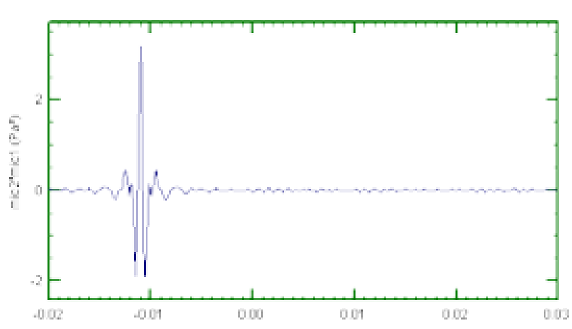
A section of $x_1(t)$ and $x_2(t)$ is shown below

Correlating x_2 with x_1 as reference gave a peak in the cross



correlation at -11 msecs as shown below

The distance between the microphones was 39 feet. A delay of



-11 msecs represents a distance of -11 feet.

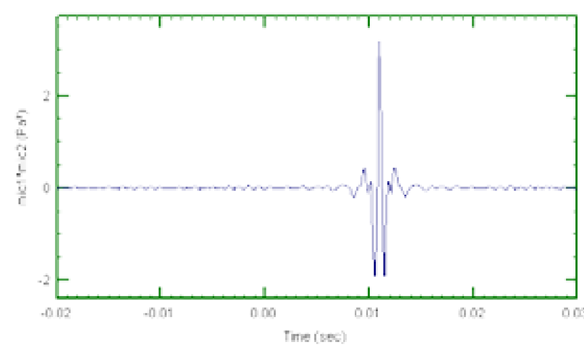
Thus if we let the distance of the source from microphone one be d_1 and from microphone two be d_2 then we have

$$d_1 + d_2 = 39$$
$$d_2 - d_1 = -11$$

Solving gives d_2 = 14 feet and d_1 = 25 feet which are the correct results.

Note that if we correlate x1 with x2 as the reference then the delay is 11 msecs as shown below.

In this case we have



$$d_1 + d_2 = 39$$
$$d_1 - d_2 = 11$$

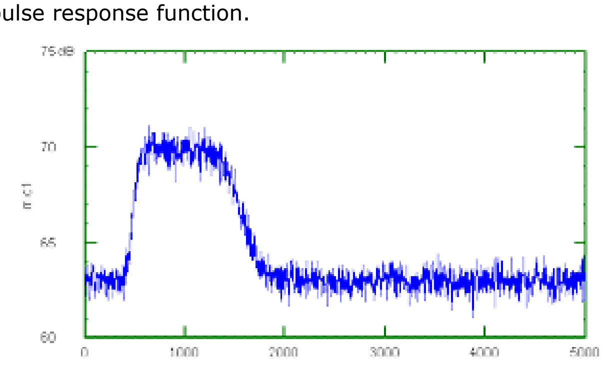
Solving again gives d_1 = 25 and d_2 = 14.

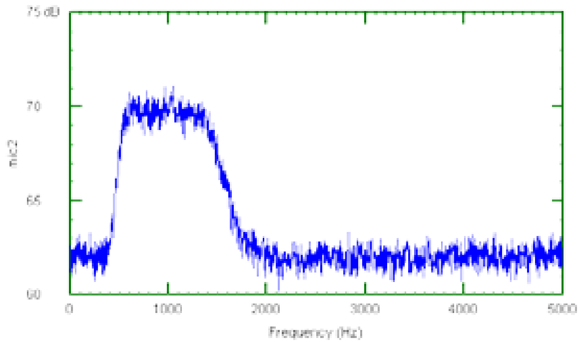
An alternative approach is to carry out a form of transfer function analysis between the 2 microphones responses. This is not a strict transfer function which is normally an excitation and response to that excitation. However the delay information is still contained in the signals. Now the transfer function H(f) is defined by

$$H(f) = \frac{G_{xy}(f)}{G_{xx}(f)}$$

where G_{xx} is the auto spectrum of the "excitation" and G_{xy} is the cross spectrum of the response with respect to the excitation. Once H(f) is known then the impulse response function h(t) may be found by inverse Fourier transforming H(f). The way in which h(t) is formed means that positive time delays are from 0 time to the mid time point and negative delays are from the last time point back to the midpoint. An option is available to carry out the time shift automatically.

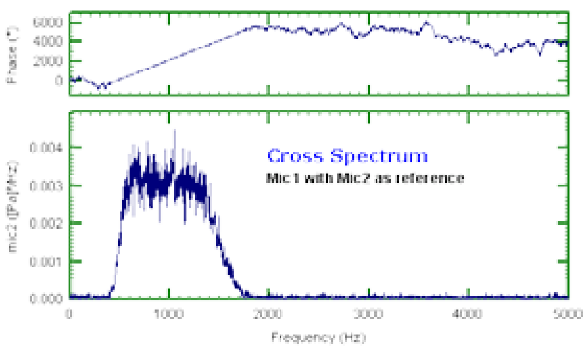
The graphs below show the two auto spectra, the cross spectrum, the pseudo transfer function, the coherence function and the impulse response function.

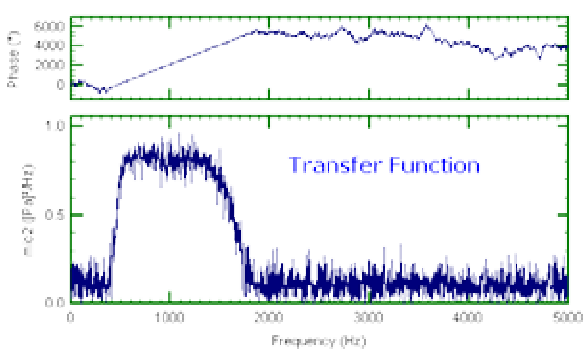




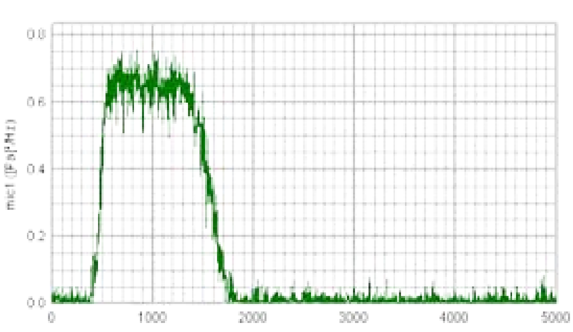
The two auto spectra above are very similar as would be expected but differences are discernible. Also the broadband nature of the uncorrelated noise is evident at around the 62dB level.

The cross spectrum and the transfer function are also very similar except for scale. In both cases the phase has been unwrapped which shows the effective linear delay.



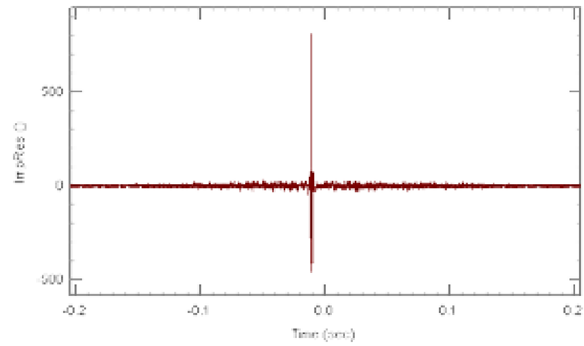


For reference purposes we have also shown the Coherence function.

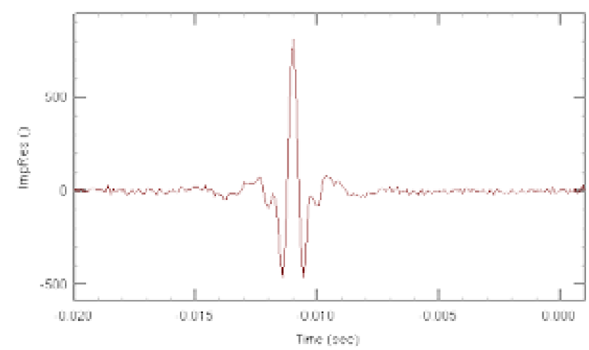


As would be expected in this case the coherence outside the effective bandwidth of the signal is essentially zero. The coherence indicates that everything between about 500 to 1500 Hz is safe to believe but outside that region there is little or no relationship.

Finally the "Impulse Response" function was formed by calculating an inverse FFT of the "transfer function" and selecting the timeshift option to get positive and negative time shown.



The peak is just before zero. Showing an expanded view reveals the time delay as -11 msec as expected.



If we had just inverse transformed the cross spectrum we would have just got the correlation function. Dividing by the input auto spectrum is useful however because it effectively normalizes the data. Another helpful scheme is to multiply the cross spectrum by the coherence as this effectively eliminates the unrelated parts. This is not necessary here as we have a sufficient signal to noise ratio.

Digital Filtering Basics

Dr Colin Mercer

A quick run through the fundamentals of filtering digital signals

Recently when discussing with an engineering student the characteristics of filters, it became clear that some confusion exists around this subject area. Here we will attempt to explain the differences between low pass, high pass, band pass & band stop filters.

To begin we will cover some basics of signal processing. This article uses swept sinewaves to explain filtering, so first we must understand what they are.

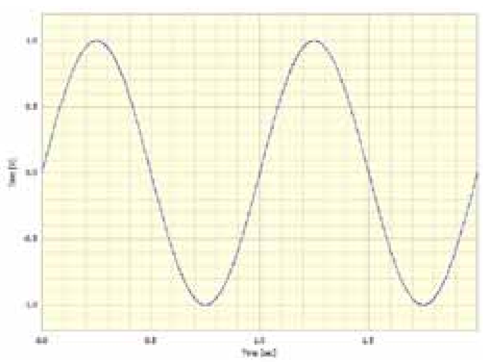


Figure 1: A Simple Sinewave

We can seen the simple sine wave is a repeating pattern, but the swept sine wave is increasing in frequency. That is, the time between the peaks is reducing. The simple sine wave has a fundamental frequency of 1Hz, but the swept sine wave has a varying frequency, starting at 1Hz and finishing at 10Hz over the 2 seconds time period.

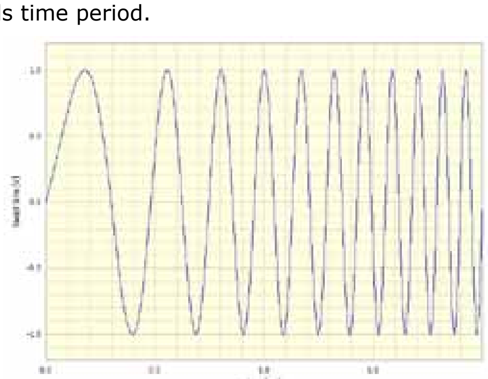


Figure 2: A Swept Sinewave

Figures 3 and 4 show the simple sinewave and swept sinewave in the frequency domain. As we can see the sinewave has one dominant frequency spike, whilst the swept sinewave shows a spread of frequencies representing the range of the sweep from 1Hz to 10Hz.

So the two types of sinewave are quite similar when viewed in the time domain. If viewing a single cycle it would be hard to distinguish them. However their frequency content is very different. Importantly, the frequency content in the swept sine wave changes uniformly across the time range of the signal.

The rest of the article will discuss the swept sine wave and the effects of certain types of filter on this swept sine wave.

Let's look at the 4 basic types of filter. Low pass, high pass, band pass and band stop. Each of these filters has different frequency characteristics.

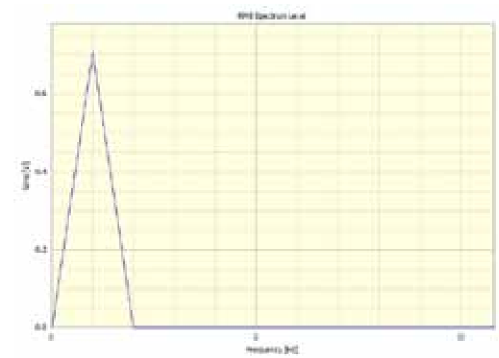


Figure 3: Spectrum of simple sinewave

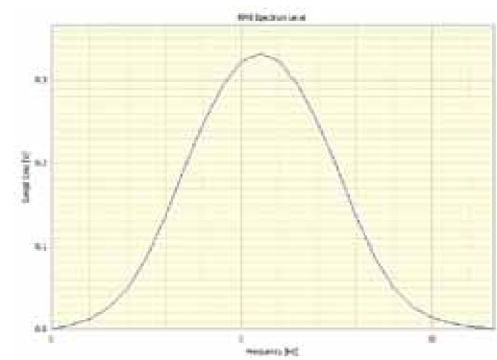


Figure 4: Spectrum of swept sinewave

Low pass filters will allow the low frequencies to pass through, but block the high frequencies. The cut off frequency is the frequency that the filter begins to attenuate the content. So a low pass filter set at 100Hz will remove the frequency content above 100Hz, but not below 100Hz. It follows that a sinewave with a fundamental frequency of 10Hz would not be affected by a 100Hz low pass filter. But a sinewave of 200Hz would be heavily affected by a low pass 100Hz filter as the frequency content above 100Hz would be removed.

High pass filters are the opposite to low pass filters. They remove the frequency content below the cut off frequency.

Band pass filters will have a low and high cut off and will pass frequencies that fall between these two limits.

Band stop filters will block the frequency content between the lower cut off and the higher cut off.

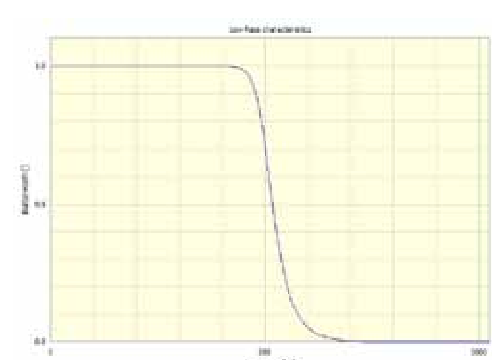


Figure 5: Characteristics of 500Hz low pass filter

We call the rate at which the filter attenuates the frequency content, the roll off rate. The filter cut off point for a low pass filter of 100Hz does not mean that the filter begins to work at 100Hz. This means that the filter will have attenuated the signals amplitude by about 30% at that point. This is known as the filter 3dB point, where the energy or power of the signal has reduced by 50% (and the amplitude reduced by a factor of 0.7071). The 'rate' of the roll off is measured in attenuation per frequency (dB per octave). This is the number of dB being attenuated per frequency octave, where an octave is a doubling of frequency.

Figure 5 shows the characteristics of a low pass filter, this example would allow the low frequencies to pass but block frequencies above 500Hz.

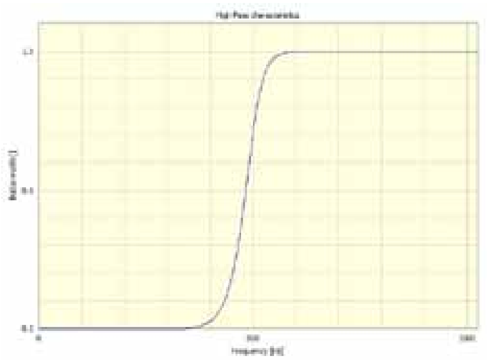


Figure 6: Characteristics of 500Hz high pass filter

The high pass, shown in Figure 6, would block frequencies below 500Hz, but allow frequencies above 500Hz.

The band pass, shown in Figure 7, would block frequencies below 250Hz, allow frequencies between 250Hz and 750Hz, then block frequencies above 750Hz.

The band stop filter, shown in Figure 8, would allow frequencies up to 250 Hz, block frequencies between 250Hz and 750Hz, but allow frequencies above 750Hz.

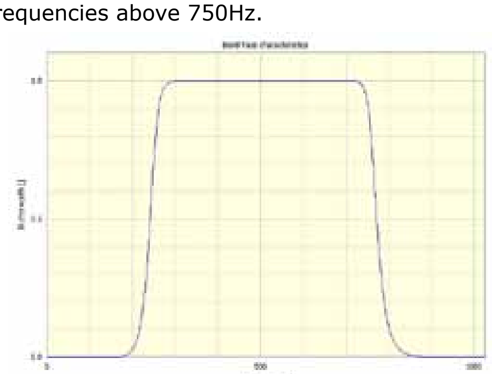


Figure 7: Characteristics of 250Hz to 750Hz band pass filter

Figure 8: Characteristics of 250Hz to 750Hz band stop filter

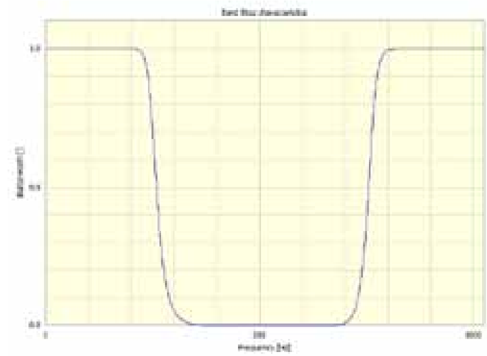


Figure 8: Characteristics of 250Hz to 750Hz band stop filter

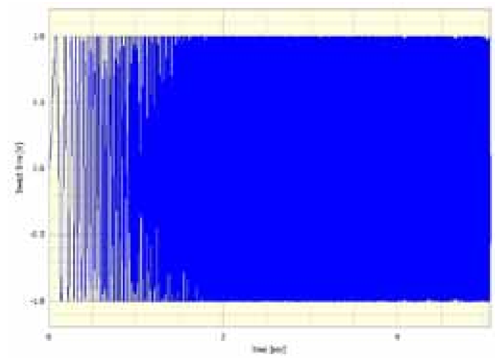


Figure 9: Swept sinewave before filtering

Which poses the next question - how would a swept sine wave be affected by these different filters?

Figure 9 shows the first 5 seconds of the swept sine wave before we have applied any filtering. This swept sinewave starts at 1Hz at t=0 seconds and increases to 1000Hz (or 1kHz) at t = 5 seconds.

Figure 10 shows the full swept sine wave after we have applied the Low pass filter. We can see how the signal is unaltered

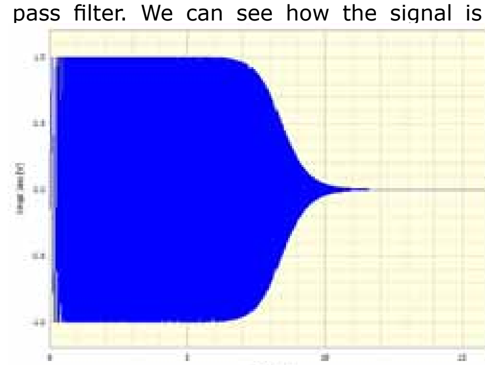


Figure 10: Swept sinewave with low pass filter applied

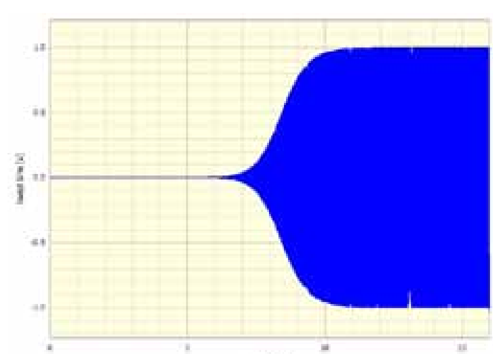


Figure 11: Swept sinewave with high pass filter applied

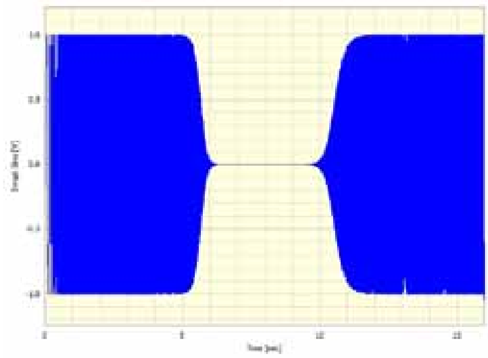


Figure 12: Swept sinewave with band stop filter applied

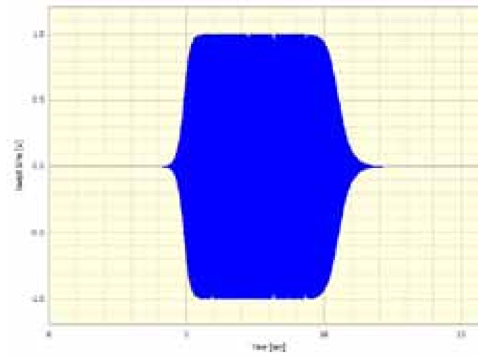


Figure 13: Swept sinewave with band pass filter applied

initially, but as the frequency approaches, and passes, the 500Hz cut off we attenuate more and more of the signal.

Figure 11 shows the full swept sine wave after we have applied the high pass filter. Here we see how the signal is attenuated at lower frequencies, but as it passes the 500Hz cut off more of the signal passes through the filter.

Figure 12 shows the full swept sine wave after we have applied the band stop filter. Clearly we can see how the filter attenuates the signal as the frequency of the swept sinewave passes through the 250Hz to 750Hz region.

Figure 13 shows the full swept sine wave after we have applied the band pass filter. Here, we see the opposite effect, where the filter only passes frequencies lying between the two cut offs.

Several properties of a filter can affect the precise form of the output. There are, for instance, many different types of filter (Butterworth, Chebyshev etc.). Also, we should consider the number of passes. This is simply the number of times we apply the filter algorithm to the signal. The more times it is applied the sharper the roll off rate. However, as well as changing the amplitude, passing data through a filter causes phase changes or delays in the output signal. The real change is frequency sensitive and depends on the number of passes, the cut off frequency and the filter type. To find out more about this and how you can use phaseless techniques to filter data, see the article "Removing Phase Delay Using Phaseless Filters" on the **Prosig Noise & Vibration Measurement Blog** at http://blog.prosig.com/2001/06/06/removing-phase-delay-using-phaseless-filtering/

Nyquist Theory and Rotational Order Analysis

James Wren

Everyone knows the Nyquist Theorem, but how does it apply to Order Analysis

Nyquist theory is generally understood, but this understanding usually relates to time sampling and the conversion to the frequency domain. Rotational order analysis and the effect of the Nyquist frequency are, however, less well understood.
Most engineers will understand the basic principle that Harry Nyquist defined and that was further developed by Claude Shannon. To re-create the frequency of interest you need at least two samples. As engineers experienced in real-world engineering know, more may well be required than that minimum.
However, when analyzing in the order domain, how many samples are required? For time domain analysis and for synchronous analysis?

Order analysis can be computed from data in the time domain or data in the synchronous domain.

So what is the relationship between the synchronous sample rate and the highest order that can be analyzed? And what is the relationship between the time sample rate and the highest order that can be analyzed?

What is the maximum order I can analyze with synchronous data?

Where the highest order is O_{max} and where N is the synchronous sample rate, in samples per revolution

So $\mathbf{O_{max} = N/2}$

So the highest order that can be analyzed is the synchronous sample rate divided by two.

The Nyquist-Shannon Theorem

Basically, the Nyquist–Shannon sampling theorem establishes a sufficient condition for a sample rate that permits a discrete sequence of samples to capture all the information from a continuous-time signal of finite bandwidth.

The name Nyquist–Shannon sampling theorem honors Harry Nyquist (1889-1976) and Claude Shannon (1916-2001). Shannon built on the earlier work of Nyquist to develop what became known as the Nyquist-Shannon theorem.

The theorem was also discovered independently by E. T. Whittaker, by Vladimir Kotelnikov, and by others. and so is sometimes known as Nyquist–Shannon–Kotelnikov, Whittaker–Shannon–Kotelnikov, Whittaker–Nyquist–Kotelnikov–Shannon and also the cardinal theorem of interpolation.

What is the maximum order I can analyze with time based data?

Where the highest order is O_{max} and where S is the time sample rate, that is samples per second and where R is the shaft speed in revolutions per second, effectively RPM/60

So $\mathbf{O_{max} = S/(2*R)}$

So the highest order that can be analyzed is the time sample rate divided by twice the revolutions per second.

Note the appearance of the twice factor, in both equations. It is not a coincidence that this factor is the same as the Nyquist value as discussed earlier. It is in fact the same relationship.

Quantifying Signals – Peak, Peak-to-Peak, & RMS metrics

John Mathey

Anytime you measure something, there are multiple ways to quantify the signal

Anytime you measure something which is changing with time, there are multiple ways to quantify the signal. For the purpose of this discussion, we will be talking about how to describe the signal in the time domain.

There are several ways to describe what the time signal is doing. Perhaps the easiest to understand are the Peak and the Peak-to-Peak measures of the signal. These are quite easily understood by looking at the graphical representation of how the signal changes with respect to time. For this purpose, we look at the graphical representation of a simple sinusoid signal.

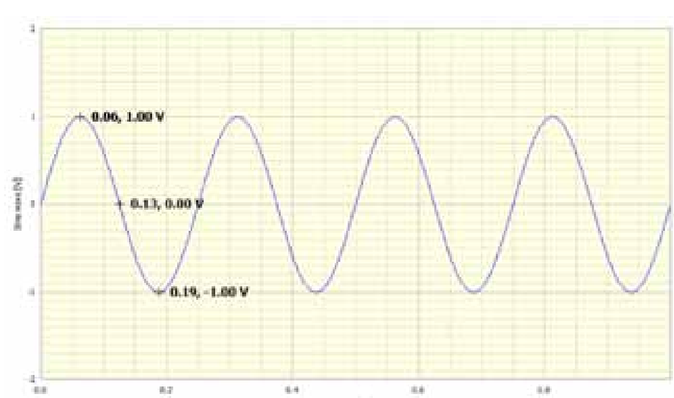


Figure 1: A simple sinusoid

Note that this is a very well behaved and repetitive signal so it is easy to determine the actual values of the signal. It is readily seen that the absolute maximum value (Peak value) of the signal over the measured time is 1.00 Volts (1.00 Vp) and the absolute minimum value of the signal is seen as -1.00 Volts, so the Peak-to-Peak value is 2.00 Volts (2.00 Vpp). In the case of this signal the Mean value is 0 (zero) Volts, but this does not always have to be the case, as is seen in the following example..

This is the same signal as before, but now it has an offset but with a mean value of 0.5 Volts, The absolute maximum value is 1.50 Volts (1.5 Vp), and the absolute minimum value is -0.50 Volts, so the Peak-to-Peak value is still 2.00 Vpp.

Another measure of signals is called RMS which means root-mean-square of the signal, also known as the quadratic mean. It is a statistical measure of the magnitude of a varying quantity. This measure gives additional information about the signal,

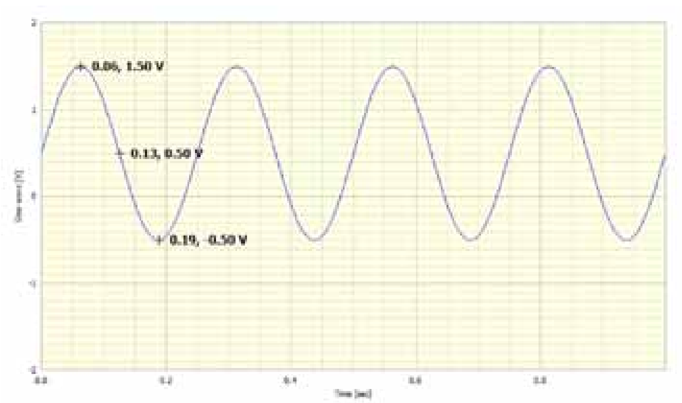


Figure 2: Sinusoid with mean of 0.5

namely the "power" of the signal. In more practical terms the RMS Voltage of a time varying signal is the equivalent DC Voltage which yields the same power.

This metric is calculated just as the name implies (RMS means the "root" of the "mean" of the "square" of the values of the signal). First the value of each point of the signal is squared, then these values are averaged over time, and then once this single number is calculated, the square root of this number is the RMS value of the signal. For discrete data points, this is mathematically calculated as:

$$RMS = \sqrt{\left[\frac{1}{n}\left(x_1^2 + x_2^2 \ldots + x_n^2\right)\right]}$$

where n is the sequential number of steps in time.

For continuous functions

$$RMS = \sqrt{\left[\frac{1}{T}\int_1^T x(t)^2 dt\right]}$$

Where T is the time over which the average is taken.

Another metric sometimes used to quantify a signal, although not as frequently, is the average. This is calculated in similar way to how the RMS is calculated, but in this case the signal values are not squared before averaging and the final number is not square rooted.

Looking at examples of how the Peak and Peak-to-Peak values are determined and how the RMS values are calculated shows that the Peak measurements are instantaneous measures whereas the RMS values are average measures. Note that all of the metrics described for quantifying a time signal are totally independent of the period or frequency content of the signal. These calculations can be applied to any time varying signal no matter how complex.

CONDITION MONITORING SOLUTIONS

PROTOR

OnLine Vibration Monitoring System

"PROTOR protects your plant and provides a future-proof solution for health monitoring

PROTOR provides a complete hardware and software solution for OnLine Vibration Monitoring (OLVMS) of rotating machines. Distributed, multichannel, data acquisition and processing subsystems are connected using standard networks to one or more database servers providing access to both real-time and historic data. Data is available locally and remotely on LANs, WANs and VPN/modems for display in graphical and numerical forms.

To help the Vibration Analyst PROTOR provided additional tools and features such as daily summary emails showing the current machine conditions, alarms activity and system status, automated report generation, long-term trending and alarm investigation.

The PROTOR P4700 range of hardware has been designed for the monitoring and analysis of vibration and associated process parameters within an industrial environment.

PROTOR is the most cost effective and efficient way of capturing, analyzing and reporting data from rotating plant. The PROTOR system protects your plant and provides a future-proof solution for machinery health monitoring.

What Are The Benefits of Vibration Condition Monitoring?

Improved Efficiency

Improved overall efficiency by increasing machine uptime.

Improved Safety

Improved safety by decreasing the risk of machine failure.

Continuous machine health information allows predictive maintenance, **avoiding and limiting machine damage**.

Reduced Capital Costs

Reduced capital costs by extending machine service life.

Decreased Servicing Costs

Decreased machine servicing costs by only repairing or replacing those parts that are damaged or worn out.

Decreased Repair Costs

Decreased machine repair costs by recognizing problems before they cause serious damage.

Reduced Downtime

Reduced machine downtime by allowing machines to be maintained while in service.

Reduced Risk

Reduced risk of unplanned shutdowns by allowing scheduled maintenance to coincide with production requirements.

PROTOR supports

- Bearing performance analysis
- Examination of critical runup / rundown modes
- Data comparison at known speeds
- Accurate and reliable trends
- Ability to set up accurate limits
- Examination of blade frequencies
- Full FFT processing of runup / rundown data
- Historical data storage
- Better control of machine rotor startup
- Remote access via modem, VPN or WAN

PROTOR can help detect

- Shaft imbalance
- Shaft bowing
- Misalignment
- Looseness of rotating elements
- Balance piston rubs
- Generator shorts
- Blade response
- Shaft rubs
- Bearing faults
- Shaft cracking
- Whirl problems
- Damaged sealing strips
- End winding vibrations
- Hub cracking

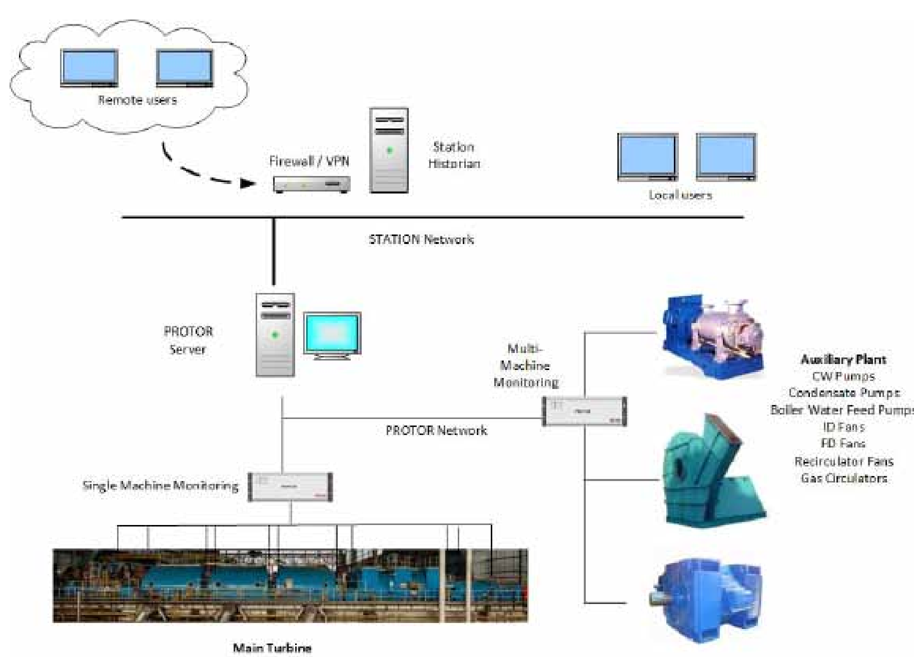
PROTOR provides comprehensive alarm checking, mimics, trends, vector plots, orbits, FFTs, waterfalls and so on, for real time and historic data.

Displays include

- Runups
- Rundowns
- Mimic diagrams
- Orbits
- Vector plots
- FFTs
- Numerical displays
- Trend plots
- Order plots
- Waterfalls
- Demand time and spectra
- Reference overlays during runup and rundown
- Combined vibration and plant process parameter plots
- Cascade and overlay X-Y plots

Other Features

- Programmable signal conditioning facilities for shaft displacement / eddy current probes, IEPE devices, velocity transducers, accelerometers and 4-20mA devices
- Individual ADC per channel for dynamic (vibration) signals or multiplexed configurations
- Synchronous sampling circuitry and analysis relative to keyphasor to provide
  - Harmonic amplitude and phase analysis for orders 1x, 2x, 3x, 4x
  - Harmonic amplitude and phase for user-selected order
  - Subsynchronous amplitude and frequency measurement
  - Overall level
  - Intra-harmonic amplitude
- Spectral Banding mode of operation (amplitude for up to five spectral bands plus overall amplitude)
- Digital input capability to define machine states such as running/stopped, onload/offload and barring/not barring
- Digital output capability for alarm indicators and watchdog
- Daily summary events
- Alarm & state change events
- Long-term trending
- Report generation
- Alarm investigation



PROTOR

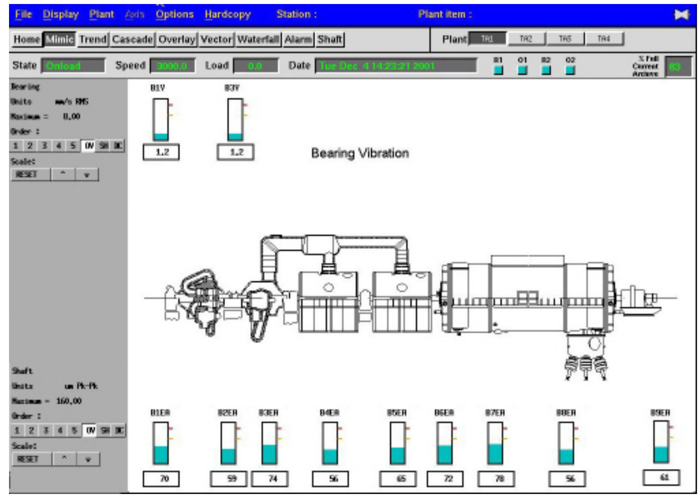
P4700 System

The P4700 is designed to be mounted close to the signal source thus reducing expensive cabling costs. Communication to the P4700 is via standard TCP/IP Ethernet. For particularly harsh environments the P4700 may be mounted in an appropriately IP-rated enclosure for protection from excessive dirt and moisture. P4700 uses 24V DC power.

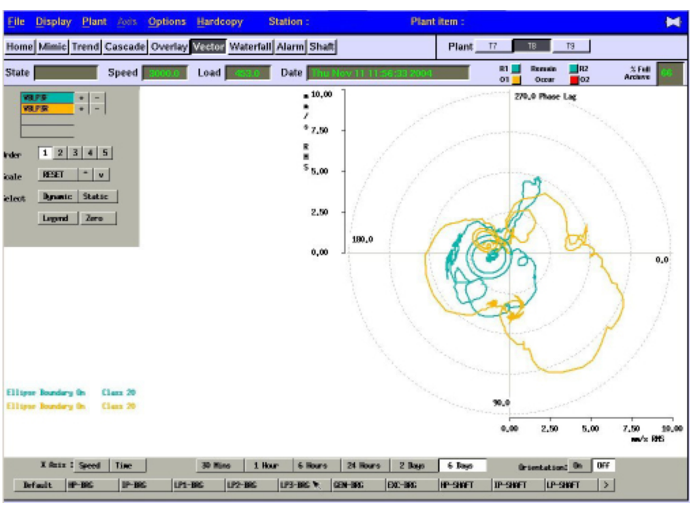
Dimensions	**Height 178mm (4U) x Width 448mm (19inch rack mounting) x Depth 361mm**
Power Requirements	**24V DC 150W or 6A @24V External rail-mount 110/220V AC to 24V DC power supply available as an option.**
Input channels	**Up to 32 channels per chassis Multiple chassis may be configured.**
Resolution	**16-bit**
Maximum Sample Rate	**20K samples/second per channel**
Input Frequency Range	**DC to 10KHz**
Input Coupling	**Programmable DC Direct , AC with 1Hz high pass filter.**
Transducer Excitation	**Programmable 24V DC power per channel.**
ICP® Transducer support	**4mA supply from 24v source**
Input Voltage	**Programmable +/- 24V, +/-10V +/- 1V +/- 100mV**
Input Impedance	**800 KOhm. Galvanic isolation available as an option.**
Maximum Common Mode Voltage	**±200V common mode compliance.**
Anti-aliasing Filter	**Programmable 8-pole Butterworth low-pass filter in range 200Hz to 20KHz.**
Phase matching	**Better than 1deg between channels. Phase compensation for filter delays performed.**
Tacho Input Range	**±24V**
Tacho Frequency Range	**0 to 120,000 RPM**
Tacho Conditioning	**Dedicated conditioning for up to 4 tachometer signals per unit. Programmable threshold, slope and re-trigger period.**
Speed Reading Accuracy	**Better than +/- 1RPM**
Synchronous Sampling	**Programmable synchronous sampling circuitry to provide 16,32,64,128 or 256 pulses per cycle.**
Constant Sampling	**Programmable constant rate sampling clock.**
Frequency Resolution	**Selectable FFT sizes of 128, 256, 512 or 1024**
Amplitude accuracy	**Accuracy of both AC and DC measurements referenced to full scale amplitude. Measurements are better than 2% fsd for amplitude and 2% in phase in all conditions apart from very low-speed or high rates of change of speed (> 50RPM/sec)**
Speed Resolution	**< 10RPM increments for speed change < 50 RPM/sec.**
AC/DC separation	**Circuitry allows DC (Gap) component and AC (Vibration) component to be measured simultaneously for displacement or proximity type transducers.**
Time Synchronisation	**Accepts synchronisation commands from PROTOR Host. Internal, battery-backed date/time storage.**
Indicators	**Front-panel LEDS for Power and Tacho trigger.**
Signal Inputs	**D-type sockets on back panel. Rail-mount screw terminals**
Power Fail Recovery	**The system is capable of re-booting and restarting normal operation following power failures and without user intervention.**
Diagnostic Functions	**Each data acquisition card within the system contains an onboard signal generator for fault diagnosis. Under software control, each channel on that card may be temporarily switched from reading its normal external input to reading the signal from the internal generator. The signal follows the same signal path as the normal vibration signal. The resultant waveform is analyzed by the system and compared to a table of expected results. Any deviations from expected within a certain tolerance are highlighted and reported to the System log.**

Some screenshots of the PROTOR system in action

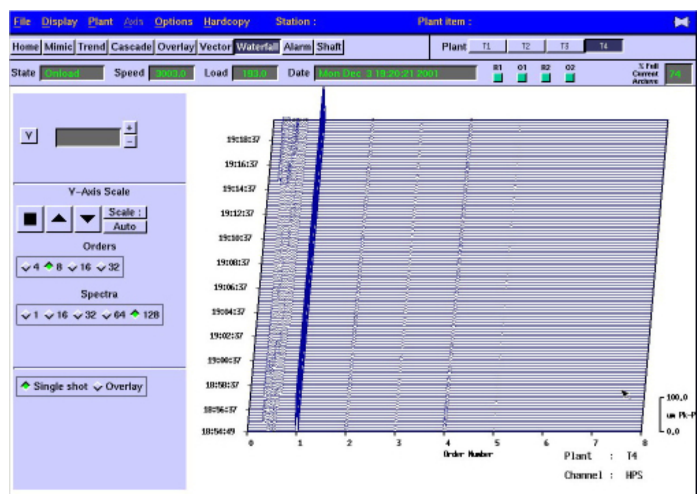
PROTOR "Home" Page



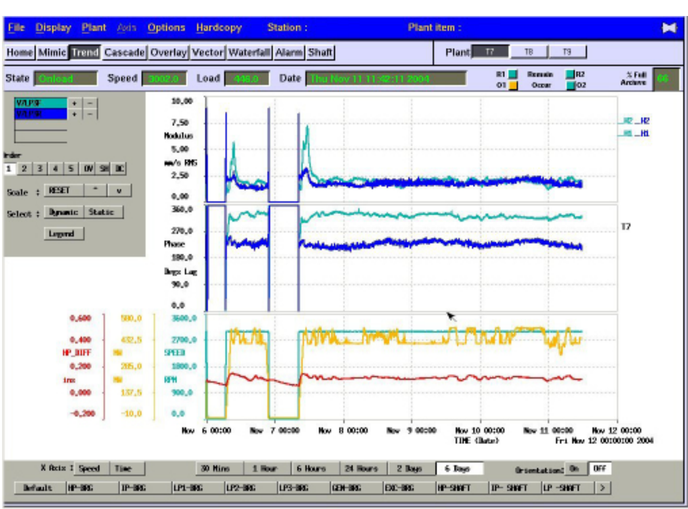
Mimic Diagram



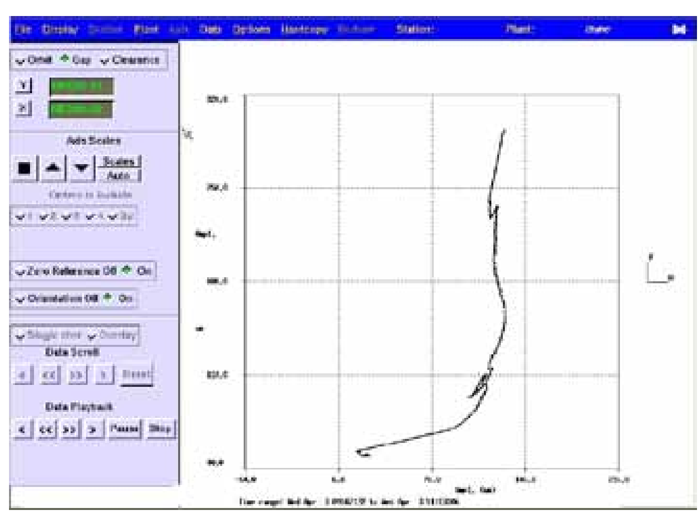
Vector Plot



Waterfall



Trend Plot



Shaft Gap

PROTOR Report Generator

PROTOR's automatic reporting tool produces detailed reports for your whole station across months or years of data with just a few clicks. You can create your own template or use one of ours. This is the ideal reporting tool for Vibration Analysts monitoring one or more stations.

Zone A	Zone B	Zone C	Zone D	N/A

Channel Name	Units	Overall
COMPEX	um Pk-Pk	42.17
COMPEY	um Pk-Pk	63.58
COMPVY	mm/s RMS	2.09
COMPVX	mm/s RMS	2.89

Basic Reports

ReportGen's basic reporting gives you an overview of the station over a time range including:

- Current alarm settings
- Current alarm states
- Alarms triggered over the time range
- Steady-state vibration levels compared against ISO 7919 or 10816
- Statistics of machine operation

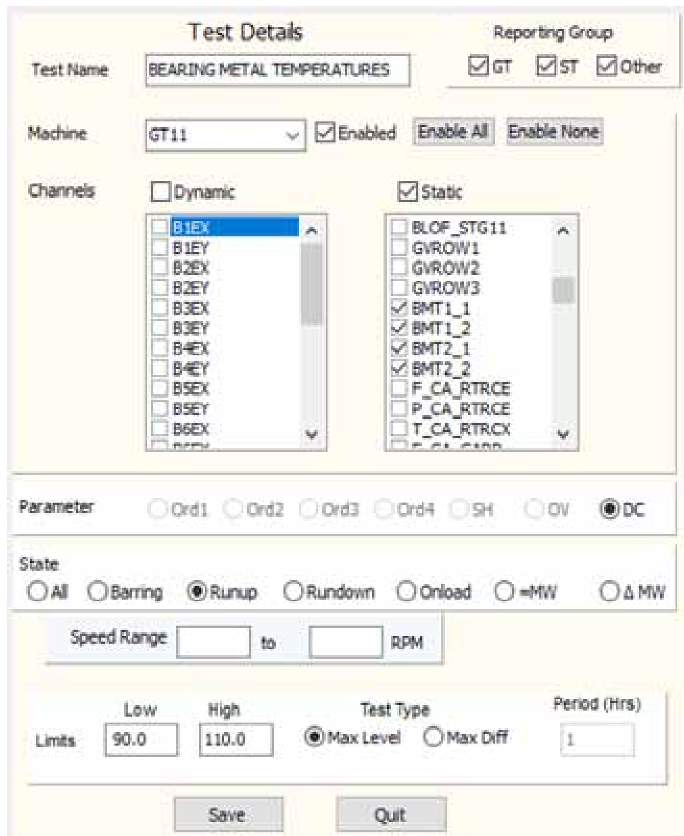
Steady-state Vibration ISO Comparison

ISO 7919 and 10816 classify steady-state vibration in four Zones. ReportGen automatically classifies the steady state vibration of the machines at your station and produces easy to read tables like the one above.

Advanced Reports – Machine Tests

The Advanced Reporting functionality of ReportGen allows you to use standard PROTOR tests appropriate to your machine or build your own custom tests. These can be used to identify common faults before they become a problem and quickly document the data and your recommendations.

Using the UI pictured to the right you can select any vibration channel, static channel and/or PI tag; specify a machine running state and a numerical test.



Prebuilt example tests exist for:

- Bearing metal temperatures
- Compressor stall
- Critical Frequencies
- Oil Whirl
- Cyclic Vibrations

You can view the results of all of your custom tests in one simple table and then expand on each one with screen shots and tables of the data.

Machine Characteristic	State	Order	Units	Low	High	GT1	GT2	GT3	GT4
BEARING TEMPS	All	DC	deg	96.5	110.0	101.9	96.6	98.6	108.4
COMPRESSOR STALL	Rundown	SH	um PkPk	160.0	320.0	364.6	315.6	353.0	499.1
TURBINE 1ST CRITICAL	Runup	O1	um PkPk	80.0	160.0	47.1	66.0	42.8	74.1
CYCLIC VIBRATION	Stable Load	O1	mm/s RMS	2.0	2.5	2.1	2.3	1.4	2.0
ROTOR THERMAL	LoadChange	O1	mm/s RMS	2.0	4.0	1.7	2.5	0.7	1.1

PROTOR-mobile System

Lightweight
Rugged
Portable
Local & remote data access
Turbine & auxiliary monitoring
Wide range of fault detection
Real-time & historic data
Choice of data visualizations

PROTOR-mobile is the ideal investigative tool for condition monitoring of rotating machinery. It provides the same extensive facilities as a standard PROTOR system but as a portable unit, rugged enough to withstand the typical power station environment.

PROTOR-mobile provides high quality, high-speed data acquisition and analysis. Data acquired synchronous to a once per revolution tachometer signal to provide high accuracy harmonic and sub-harmonic measurements or time-based for slow speed conditions or where a tachometer signal is not available.

A standard signal conditioning module allows inputs to be taken from accelerometers, velocity or displacement probes. The system can provide 24V DC excitation for proximity probes and also supports IEPE transducers. Signal conditioning parameters such as gain, AC/DC coupling and anti-aliasing filter selection are programmable.

PROTOR-mobile is a standalone unit. Access is provided either by standard monitor, keyboard and mouse ports or by Ethernet or by modem. Typically access will be via a Notebook computer running Windows and using an Ethernet connection. Connection may also be via an existing station network which would then provide local area connection even remote connection should Wide Area or VPN facilities be available. Using either connection PROTOR-mobile may be configured and set to work, thereafter the system may be left unattended to collect and store data to its internal SSD.

PROTOR-mobile extends the capabilities of the P4700 bringing the PROTOR server down into the box. It can be configured with up-to 32 inputs and four phase-reference signals (D25 or BNC). The inputs may be configured as either vibration signals (dynamics) or plant process parameters (statics). Each of the phase reference signals may be associated with an 8-channel signal group. This provides great flexibility in configuring the system for a number of different applications. In the simplest case the system may be configured to monitor a single machine using all channels relative to a single tacho. Alternatively it is possible to monitor multiple machines each with their own 8-channel blocks and individual phase-reference signals. The total channels allocated to each machine can be configured as any combination of dynamic and static channels.

PROTOR-mobile provides the same intuitive Graphical User Interface as used by the standard PROTOR system. This simply allows the user to select the type of display and the channels to be displayed by pop-up and pull-down menus using 'point and click' selection and the graphical selection of the data to be displayed.

PROTOR-mobile allows screen shots to be easily cut and pasted into standard Windows software and data to be easily exported.

Measuring Shaft Displacement

Don Davies

Measuring shaft displacement using a VCM system

Shaft displacement is an important vibration measurement for rotating machines. Shaft displacement is usually monitored by non-contact shaft displacement probes such as eddy-current probes. These probes produce a voltage proportional to the distance of the shaft surface relative to the tip of the probe. For maximum benefit, ideally two shaft displacement probes will be fitted to measure the displacement in both the horizontal and vertical directions.

The diagram in Figure 1 shows a typical arrangement.

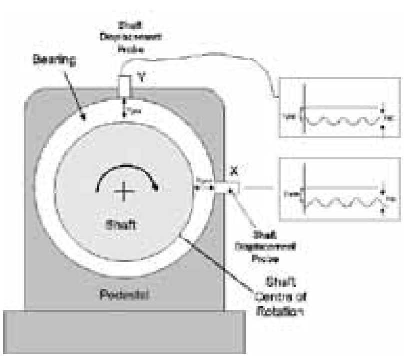


Figure 1: Eddy Current probes

This shows that the vibration signal from shaft displacement probes contains both AC and DC components. The DC component is a measure of the overall distance of the shaft from the probe, this is called the gap. The AC component is a measure of the movement of the rotating shaft about its central position. In general the DC component is large (typically -15V) with a much smaller AC component. The PROTOR data acquisition hardware includes dedicated signal conditioning which allows both the AC and DC components to be measured with high accuracy using only a single input channel.

Shaft Vibration

The AC component is usually analyzed with respect to a 'once per revolution' tachometer signal to provide measurements which are an indication of the movement of the shaft on a rotational or 'per cycle' basis. This provides information which is used to detect phenomena such as unbalance, misalignment, rotor bends, cracks and so on. For example, assume a rotor, supported by two bearings, has a bend or bow as shown below (greatly exaggerated for display purposes) then the displacement time history would be sinusoidal.

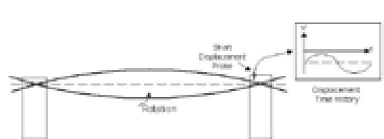


Figure 2: Rotor supported by two bearings

The PROTOR system measures the AC signal for displacement probes and performs frequency analysis on the signal with reference to the tachometer signal to identify the Overall displacement on a cyclic basis together with its constituent components such as the 1st, 2nd, 3rd, 4th and higher harmonics (both amplitude and phase), sub-harmonic (amplitude and frequency) and intra-harmonic components. These measured components are collected and stored on a regular basis and made available for real-time mimic diagrams, trend displays, vector diagrams, alert processing and also for historical analysis.

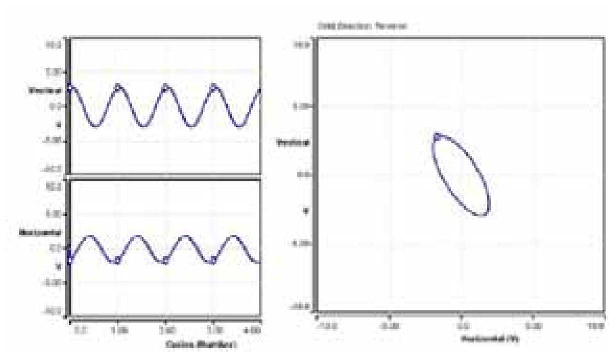
Transducer Orientation

To be of most benefit a pair of perpendicular shaft displacement probes are often used to allow measurement of the movement in both the vertical and horizontal directions.

NOTE: It is often not physically possible to mount probes in the actual vertical and horizontal planes. The PROTOR system configuration allows the actual transducer mounting position to be defined. It can then mathematically combine the contributions of a pair of probes to estimate the actual displacement in the true vertical and horizontal planes.

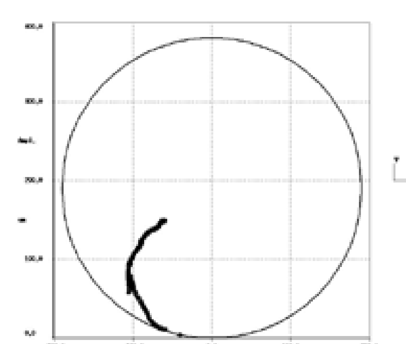
Orbit Plots

Two perpendicular shaft displacement signals may be either directly measured or determined through the orientation software. When two such signals are available then PROTOR is able to display the data in the form of a shaft 'Orbit'. An Orbit display is effectively a dynamic display of the movement of the centre of the shaft. Within PROTOR it is possible to display the 'filtered' orbits, that is the individual contributions from each of the measured orders. Alternatively you can select which orders to include in the orbit display.



Shaft Gaps

As mentioned above the signal from a shaft displacement probe also has a DC component which is proportional to the average gap between the probe tip and the shaft surface. The PROTOR system also measures and logs these components and makes them available for trending and display. If bearing clearance information is available then this may be entered and the movement of the shaft shown relative to the clearance.



Signal Conditioning for High Common Mode & Isolation

Don Davies

Dealing with issues of differing ground potentials in an industrial environment

For monitoring systems in an industrial environment special care and attention is required for both signal cables and input signal conditioning circuitry. Typical problems in this environment include long cable runs and cable routes in the proximity of high voltage sources can cause noise induction and large ground potential differences to exist. The effect of differing ground potentials between the signal source and the measurement system is of particular interest. For monitoring systems in a clean or laboratory environment then the signal source and measurement system are close together and ground or earth differences are negligible and so can be ignored.

The following notes describe some of the concepts and terminology related to these phenomena and describe ways in which these effects can be minimized by careful selection of signal cabling and signal conditioning components.

Single-Ended Inputs

With single-ended inputs a single connection is made from the signal source to the data acquisition system. The measurement made is the difference between the signal and the ground or earth. In order for the measurements to be accurate then we must ensure that the signal source is grounded (earthed) and the signal source and the acquisition system's earth have the same value. In most practical or industrial applications the ground or earths may be significantly different between the transducer source and the measurement system. Single-ended inputs are also sensitive to noise errors, in particular for long cable runs.

Differential Inputs

One way to eliminate this problem is to use differential inputs to a differential amplifier. With differential inputs, two connections are made from the signal source to the measurement system. The differential amplifier gives the difference between the two inputs, meaning that any voltage common to both wires is removed. Therefore, providing the difference in earth potential between the source and measurement system is not too large, then it does not affect measurement accuracy.

However in a number of cases especially in industrial environments where the signal source may be a long distance from the measurement system or when 'floating' inputs are used (which have no ground reference) then the difference in grounds may be significant. In these cases we need to take account of the voltage compliance range of the input amplifier and if necessary use specialist components or circuitry which removes or rejects this voltage difference.

Common-Mode

The common-mode voltage is defined as the voltage that is measured with respect to a common-mode reference point and is present on (or common to) both sides of a differential input signal. Most frequently, the common-mode reference point for a complete system is the system earth or ground. Problems arise if this common-mode voltage exceeds voltage compliance of the signal conditioning input circuitry, typically < 15V.

A solution is to use an instrumentation amplifier with a high Common-Mode Rejection Ratio (CMRR). The CMRR is a measure of how well the amplifier rejects the common-mode voltage. An ideal amplifier will have a CMRR of infinity. In practice, high-common mode amplifiers have a CMRR of around 80 to 90 dB. The higher the rejection ratio the better. The other important factor is the common mode range. This is the maximum common-mode voltage with which the amplifier can cope. Typical Common Mode Range values are +/- 200V. There are cases where extreme common-mode voltages may exist which may require further conditioning. In such cases Isolating amplifiers may be required.

Isolation

In some situations, a number of monitoring systems may 'share' signal inputs from a transducer, in this case care must be taken to ensure that the system does not affect the signal in anyway. In this case isolation amplifiers should be used such that electrical isolation is provided between the measurement system's input and its measurement circuitry. Such devices pass the signal from its input to the measurement device (ADC) without a physical connection by using transformer, optical, or capacitive coupling techniques. This ensures that there is no possibility of electrical current flowing from one measurement system to another.

PROTOR Solutions

As standard all PROTOR system are provided with high-common mode signal conditioning. For the PROTOR-4 range of hardware the programmable P4751 8-channel module provides the high-common mode characteristics.

Galvanic isolation may also be provided as an option. For PROTOR-4 the software programmable P4761 card is available.

Bearing & Gearbox Vibration Analysis (Pt 1)

Don Davies

Using demodulation techniques to study issues with critical components

Bearings and gearbox vibration are fundamental issues for rotating machines in many industrial applications. These are critical components and, as such, any failure can prove expensive in both repair cost and down-time. Because of this condition monitoring has become increasingly important over the years, usually centred around vibration measurement taken at critical locations, either continuously (online) or as part of a monitoring schedule. Vibration monitoring has become an integral part of most maintenance regimes and relies on the detection of various well-known frequency characteristics associated with this type of component. Detailed knowledge of design of the bearing or gearbox allows characteristic vibration frequencies to be calculated. However these frequencies are often masked by vibration from nearby components or by noise, sometimes making diagnosis difficult. Vibration time signatures are also often subject to both amplitude and frequency modulation which affect the resultant frequency spectra. Here we are going to look at the causes and effect of amplitude modulation in particular and how it is manifested in the frequency domain.

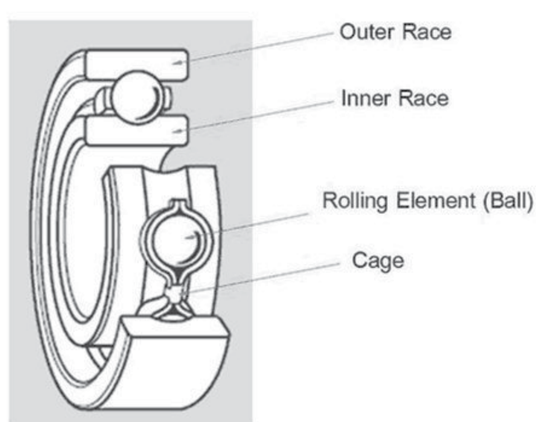


Figure 1: A typical roller bearing

Rolling Contact Bearing Features

Rolling contact bearings, either rolling ball or rolling element are used extensively in all types of rotating machine. A rolling element bearing consists of a number of balls or rollers within an inner and outer bearing ring. When faults develop within such a bearing it is often due to pitting on the surface of the elements or on the inner or outer bearing face. Standard equations exist which allow the frequency of occurrence of these impacts to be estimated from the speed of rotation and the detailed geometry of the bearing which include the inner and outer race diameters and the diameter of the individual elements. Typically we have the most common defect frequencies:

Fundamental Train Frequency (FTF)

$$FTF = \frac{f}{2}\left(1 - \frac{d}{D}cos(a)\right)$$

Ball Passing Frequency Outer Race (BPFO)

$$BPFO = \frac{nf}{2}\left(1 - \frac{d}{D}cos\left(a\right)\right)$$

Ball Passing Frequency Inner Race (BPFI)

$$BPFI = \frac{nf}{2}\left(1 + \frac{d}{D}cos\left(a\right)\right)$$

Ball Spin Frequency (BSF)

$$BSF = \frac{fD}{2d}\left(1 - \left(\frac{d}{D}cos\left(a\right)\right)^2\right)$$

where

D = pitch diameter
d = element diameter
a = contact angle
n = number of elements
f = Revolution speed (revs/sec).

Note that f is the relative speed difference between the inner and outer races. In most cases the outer race is stationary and so f is the shaft rotational speed. See Figure 2 below.

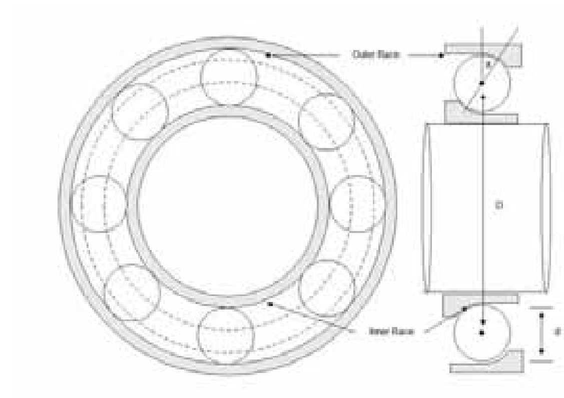


Figure 2: Bearing dimensions

Gearbox Features

There are a large number of different designs of gearbox. The simplest are known as Spur gears and are used to transfer power between two parallel shafts, usually at different speeds, as shown in Figure 3. The fundamental speeds of rotation of the shafts are defined by the ratio of the number of teeth on each gear-wheel.

fi = Input Speed
fo = Output Speed
Ni = Number of teeth on Input Gear

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No = Number of teeth on Output Gear

$$f_o = \frac{N_i}{N_0} f_i$$

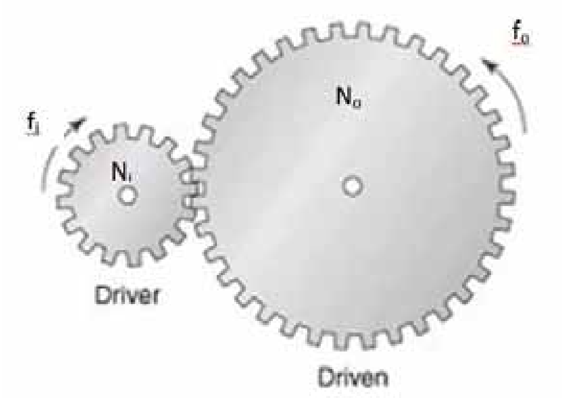


Figure 3: Spur gears

Many more complicated gearboxes, including multiple compound gear trains, follow the same principles, but the internal gear speeds are more difficult to calculate.

Gear Mesh Frequency

The main frequencies seen within a vibration spectra for a gearbox are the rotational speeds of each gear, f_i and f_o and the gear-meshing or the tooth mesh frequency, f_m. The gear-mesh frequency defines the rate at which gear teeth mesh together. The Gear Mesh Frequency is given by:

$$f_m = f_i N_i$$

or

$$f_m = f_o N_o$$

Hunting Tooth Frequency (HTF)

Another frequency which is sometimes used is the Hunting Tooth Frequency (HTF). This is found when one tooth on each gear is damaged and represents the frequency at which the two teeth contact each other. The calculation of this frequency is also dependent on the number of teeth per gear and involves finding the highest common factor (CF) between the gear ratios. For example if the input gear has 9 teeth then its factors are 1×9 and 3×3. If the output gear has 15 teeth then its factors are 1×15 and 3×5 and so the highest common factor is 3. The Hunting Tooth Factor (HTF) is given by:

$$HTF = \frac{f_m CF}{N_i N_o}$$

where CF is the highest Common Factor.

Amplitude Modulation

As discussed above, any surface defects in either a bearing or a gearbox will result in a vibration signal that will contain individual impulses due to the impacts generated when the defect comes into contact with other elements. The frequency of the impacts is dependent on design features of the component. The ability to measure and identify these frequencies will allow us to better identify these problems. In practice the signals caused by these impacts will also be modulated. This modulation could be both in amplitude and in frequency, but for the purposes of this discussion we will concentrate on amplitude modulation. Amplitude modulation is caused when the load on a bearing of a gearbox varies, typically with rotational speed. This change in load will affect the strength of the impact seen. For example, imagine a gearbox which shows signs of wear and which also has a slight bend in the shaft. The worn teeth will cause peaks in vibration to occur at the gear meshing frequency. The bend in the shaft will cause the pressure on the gear teeth to increase and then decrease during a complete revolution of the shaft. The same might apply to a bearing which is mounted horizontally. Due to gravitational forces the pressure between the element and the bearing surfaces may be greater at the bottom of the bearing rather than the top. If there is a defect on one of the elements then the impact from this may be stronger when the element rotates to the bottom of the bearing than when it is at the top. [Note that if there is a defect in the outer race and this is stationary then the impulses from this defect will not generally be subject to load variations and so will not show amplitude modulation]. The following graphs show the effect of amplitude modulation. Assume that our defect causes a pure sinusoidal output at the defect frequency, in this example 75Hz. We then amplitude modulate this signal at a lower frequency of 10Hz.

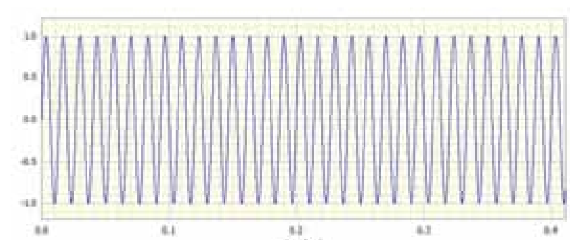


Figure 4: Main output signal at the defect frequency of 75Hz

Figure 5 shows the effects of applying amplitude modulation to this signal. The frequency of the modulation signal is 10Hz. The relative amplitude of the modulation signal, or the modulation factor, is 0.25.

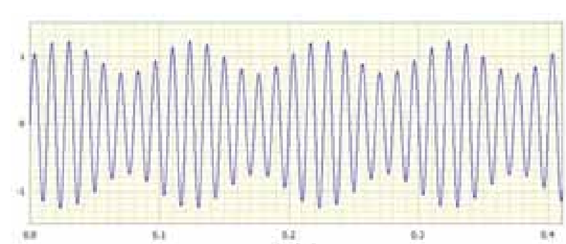


Figure 5: Amplitude modulation applied to signal

This amplitude modulation, caused by the multiplication of the higher frequency signal (f_H) and the lower frequency modulation signal (f_L), results in a signal, a(t) that is described by the standard equation:

$$a(t) = sin(\omega f_H t)\ (1 + sin(\omega f_L t))$$

This expands to:

$$a(t) = sin(\omega f_H t) + \frac{cos(\omega(f_H - f_L)t)}{2} + \frac{(cos(\omega(f_H + f_L)t)}{2}$$

That is, the resultant frequency spectrum will contain peaks at frequencies f_H, (f_H-f_L) and (f_H+f_L)Hz. The components (f_H-f_L) and (f_H+f_L) are known as 'sidebands' and are a common characteristic when performing frequency analysis on bearings and gearboxes. These are shown below in the frequency spectra of the amplitude modulated signal. Here we can clearly see the main component at 75Hz and the two side-bands at 65Hz and 85Hz.

Main component at 75Hz and two side-bands at 65Hz and 85Hz Figure 6: Main component at 75Hz and two side-bands at 65Hz and 85Hz

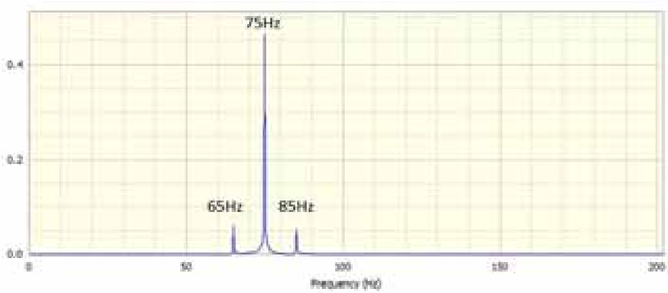


Figure 6: Main component at 75Hz and two side-bands at 65Hz and 85Hz

In most situations the main vibration signal from a defect will not be sinusoidal but often a series of impulses, repeating at the defect frequency. In the frequency domain this signal will also contain a number of harmonics of the main defect frequency. Each of these harmonics will also have side-bands. In most cases the modulation frequency will be the shaft rotational speed. Therefore when analyzing data from rolling element bearings or from gearboxes, the measurement and detection of sidebands in the frequency domain is very important. In gearboxes the gears rotate at different speeds and so there may be many sidebands present but with knowledge of the shaft speed and number of teeth per gear then detailed diagnostics are possible which can pin-point faults with specific gears and/or shafts.

Conclusion

In this discussion we have briefly looked at how we can use the detailed design information of bearings and gearboxes to look for specific fault conditions by collecting vibration information and analyzing their frequency spectra. Amplitude modulation of the vibration signatures is common and we have seen how this causes side-bands to be present in the frequency domain.

Monitoring Auxiliary Machinery

An OLVMS system isn't just for monitoring turbines...

Don Davies

The use of the PROTOR system for monitoring vibration from large rotating machines fitted with fluid-filled journal bearings such as steam or gas turbines is well understood. Vibration from these components generally falls within the main harmonics or orders of the shaft rotational speed such as 1st, 2nd 3rd or 4th harmonic. Some energy may also exist below the 1st order, called the sub-synchronous component. Most energy exists below 1KHz and so standard displacement probes or velocity transducers are generally fitted. The PROTOR system collects this data in amplitude and phase form, relative to a 'once-per-revolution' phase reference signal, as standard and allows data to be displayed in real-time as mimic diagrams, trend plots, orbit and vector displays.

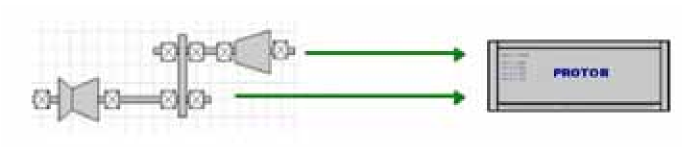
Less well known is the PROTOR system's ability to effectively monitor auxiliary items of plant such as pumps or fans. This includes rotating machines with gearboxes, rolling-element bearings, impellers and dual shaft machines. For these types of machine the vibration spectra may contain information over a wide range of frequencies that may be related to gear-mesh frequencies for gearboxes or inner or outer race frequencies for rolling element bearings. The following features are provided as standard within PROTOR and the PROTOR hardware for auxiliary plant item monitoring is exactly the same as that used for main turbines and so standard spares cover all items.

High frequency analysis

One major difference when monitoring vibration information for some auxiliary items compared with standard steam or gas turbines is the ability to monitor high-frequency content. As mentioned above, for turbines most vibration information is within the 0 to 1KHz frequency band. For high-speed auxiliary machines with gearboxes or rolling-element bearings then some frequency components may be much higher, possibly up to 10KHz. For these machines accelerometers will generally be fitted. The PROTOR P4700 system supports accelerometers as standard and will also provide a constant-current source for IEPE transducers under software control. The P4700 system contains a programmable low-pass filter and allows sampling in excess of 20K samples per second per channel.

Multi-machine configuration

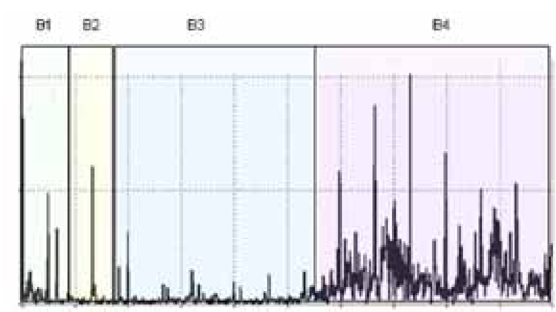
One main advantage of the PROTOR system for this type of analysis is the flexibility of the system hardware and configuration. A number of auxiliary plant equipment such as boiler feed pumps or FD and ID fans contain components running at different speeds such as a motor and a pump or a motor and a fan. A PROTOR P4700 data acquisition unit can take in up to four separate 'once per revolution' speed or phase reference signals and each 8-channel data acquisition card may be associated with any one of these speed signals.



In this example we have a LP and HP turbine each with their own phase reference signal. Signals from the LP and HP units are analyzed relative to their own phase reference. Signals from the gearbox are 'shared' and analyzed twice, once relative to the LP tacho and then relative to the HP tacho.

Spectral band analysis harmonic configuration

Another feature of the PROTOR system is the ability to configure the system to analyze and collect specific harmonics. For example, for a Gas Circulator within a nuclear power station one

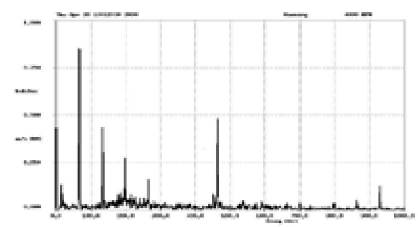


primary frequency component is related to the number of impeller blades, in this case 31. For this case PROTOR was configured to measure the 31st harmonic as standard. This component is then available alongside the other standard harmonics for display, trending and alarm checking.

Spectral band analysis

Another feature of PROTOR is the ability to configure spectral bands. These frequency bands may be set by the user for a particular machine and can be set dependent on the machine configuration around particular frequencies of interest such as gear-mesh frequencies or blade-passing frequencies. This method is used when the frequency content is well known and understood. Alternatively when the frequency content is not well known, the bands may be set for general zones of interest, say a low-frequency zone (below running speed) , a running speed zone, a general vibration zone (encompassing 2nd, 3rd and 4th harmonics) and a high-frequency zone.

Gearbox ratios



PROTOR also handles situations where only a single speed or phase reference signal is available. For example, with some gearboxes a single tachometer signal positioned on one side of the box is often the only speed reference available. In this case it is possible to define the gearbox ratio and to specify the channels associated with either side of the gearbox. For channels where the speed signal is available then normal harmonic analysis is performed. For channels on the other side of the gearbox then the speed measured by the available tachometer signal is factored by the gearbox ratio, the resultant speed is then used to determine the expected harmonic locations on this channel.

Bearing & Gearbox Vibration Analysis (Pt 2)

Don Davies

The second part of our look at demodulation techniques

In our previous article on this subject we briefly looked at how we can use the detailed design information of bearings and gearboxes to look for specific fault conditions by collecting vibration information and analyzing their frequency spectra. Amplitude modulation of the vibration signatures is common and we have seen how this causes side-bands to be present in the frequency domain.

Envelope Analysis or Demodulation

Understanding that modulation of primary fault frequencies may occur when analyzing vibration spectra from bearings and gearboxes is important when diagnosing faults. However, sometimes these effects may be difficult to see if there are strong periodic signals present due to other components or if there is high noise content.

One method often used to identify these fault frequencies and their side-bands is by performing envelope analysis. As its name suggests, envelope analysis attempts to determine the overall extremities of a signal. In analogue terms this may be performed by a simple diode to provide rectification and a low-pass filter. For digital signals there are a number of methods such as squaring the signal, to provide the rectification followed by a low-pass filter and then square-rooting.

Another method utilizes the Hilbert transform which is used to compute a complex signal from the real time signature where the imaginary component is a phase-shifted copy of the real signal. This complex signal can then be low-pass filtered. Figure 1 shows a portion a simulated impulse train which has been modulated and its envelope signal. The impulses repeat at a frequency of 75Hz and the modulation frequency is 10Hz.

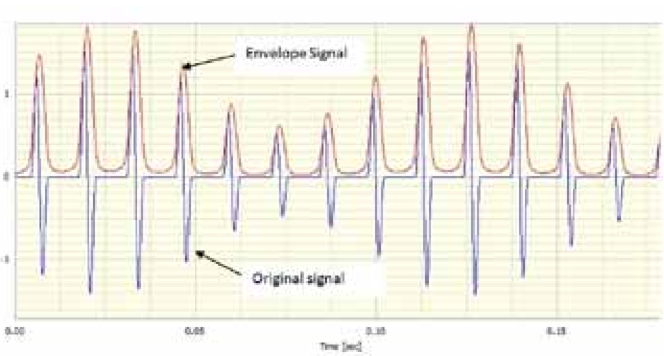


Figure 1: Simulated impulse train which has been modulated and its envelope signal

The envelope spectrum is formed by performing an FFT of the envelope time history. This is often used when the original signal has high noise content. In Figure 2 below we see the envelope spectrum from our modulated signal which clearly shows the 10Hz component relating to the rotational speed and then the impact frequency (75Hz) and its harmonics together with the side-bands. The side-bands are positioned +/-10Hz from each of the harmonics.

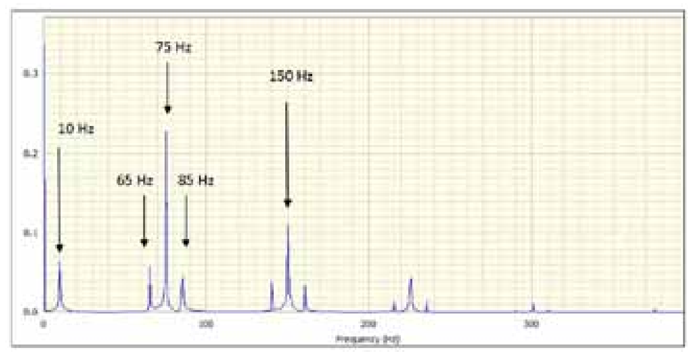


Figure 2: Envelope spectrum from modulated signal

Complex Demodulation

Another, flexible method for performing envelope analysis is called Complex Demodulation. In this method a particular frequency or frequency band is chosen and then the frequency content of the signal is shifted such that this frequency becomes 0Hz or DC. The resultant signal is then low-pass filtered to produce the signal envelope. This is shown diagrammatically below.

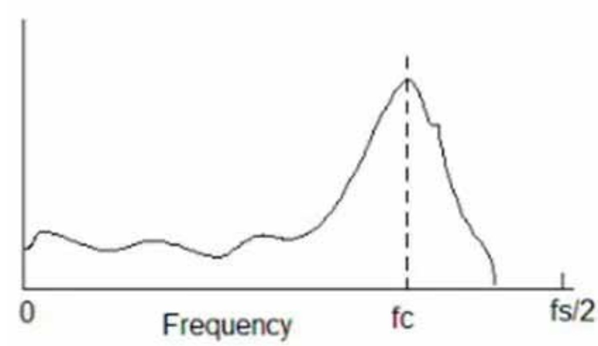


Figure 3: Half-range frequency spectrum

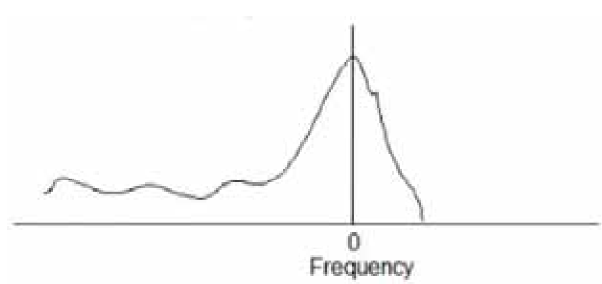


Figure 4: Half-range frequency spectrum (shifted)

Figure 3 shows a typical half-range frequency spectrum. We select the centre frequency (f_c) about which we want to perform the demodulation. In Figure 4 the frequency spectrum is shifted such that f_c now becomes 0Hz. This shift is performed my multiplying by the complex exponential:

$$e^{-i\omega t}$$

where

$$\omega = 2\pi f_c$$

Now we low-pass filter the signal. The result shown in figure 5 is a narrow-band spectrum centred on a specific frequency which highlights the modulation components.

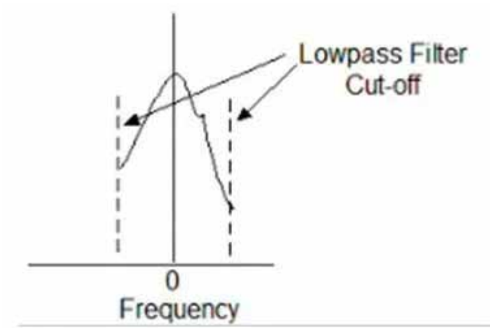


Figure 5: Narrow-band spectrum centred on specific frequency

The complex demodulation approach to envelope detection and subsequent envelope spectrum analysis can be most useful in high-noise situation. All bearings have resonant frequencies, usually at much higher frequencies than the defect frequencies we have looked at. However these resonances and its harmonics can be excited by the defect impulses. In this situation the vibration signature will consist of the resonant frequency modulated by the defect frequencies. Demodulating the signal around this resonant frequency allows the defect frequencies to be observed even in highly noisy signals. Assume our amplitude modulated signal now has a large quantity of noise added to it. The noise we added has a frequency bandwidth of 1KHz. We have also attempted to simulate a resonance at 5Khz by amplifying the modulated impulse signal in this frequency range. The time signature(Figure 6) shows that the noise dominates the signal and our defect impulses are no longer visible.

The frequency spectra of this signal (Figure 7) shows how the noise dominates up to 1KHz. However there is some evidence of harmonics of the impulse signals at higher frequencies and a slight amplification at our resonance at 5KHz.

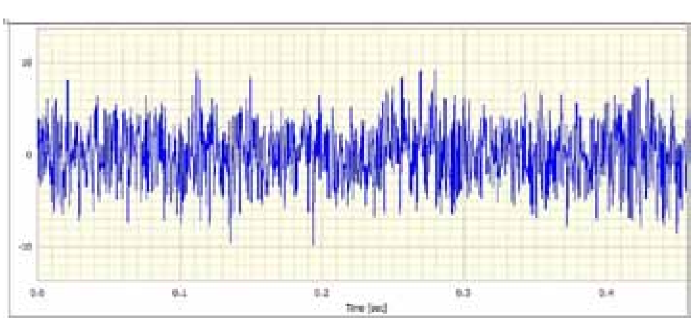


Figure 6: Signal with noise

Figure 7: Spectrum of signal with noise

If, we perform complex demodulation of the time history and request a centre frequency of 5KHz and a bandwidth of 500Hz and then form the envelope spectrum then the expected components and the side-bands again become visible as seen in Figure 8.

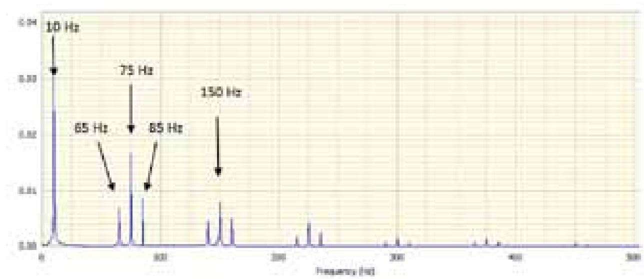


Figure 8: Complex demodulation of the time history

Conclusion

Previously, in Part 1, we briefly looked at how we can use the detailed design information of bearings and gearboxes to look for specific fault conditions by collecting vibration information and analyzing their frequency spectra.

Now, we have examined techniques such as demodulation and envelope spectra analysis which are often used to more clearly reveal any defect frequency components embedded in a vibration signature.

Understanding the Importance of Transducer Orientation

How to deal with probes that are not fitted in the horizontal & vertical planes

Don Davies

When monitoring vibration on large gas or steam turbines and generators with fluid-film bearings, the relative movement of the shaft within the bearing is typically measured by a pair of shaft proximity (eddy current) transducers. Data from these transducers is used to produce a variety of plots on a condition monitoring system including orbit displays and shaft centreline plots.

Typically these plots assume that the transducers are fitted in the vertical and horizontal direction at the bearing. However, in most practical situations this will not be possible, due to pipework or because the bearing casing is split horizontally. In these case they will be fitted at either 45 degrees from the top or the bottom of the bearing. The important fact is that the probes remain perpendicular.

Note The general convention for the naming or identification of these transducers goes back to using a dual-channel oscilloscope to view the combined XY or 'orbit' display from the resultant time history traces.

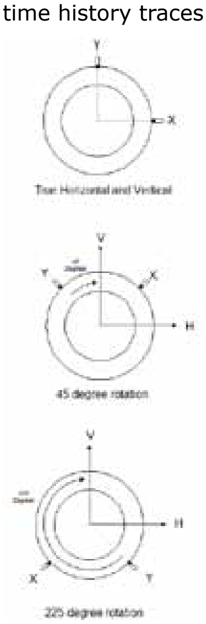


Figure 1: Orientation angles

When viewed from the 'driven' end of the bearing and for probes fitted to the top of the bearing the probe on the left-hand side of the bearing is named Y and the X is on the right. This is irrespective of the direction of rotation.

The diagram to the left shows typical transducer mounting positions and the rotation correction required to return the display to true Vertical and Horizontal. Following rotation correction the Y sensor becomes vertical and the X horizontal.

Shaft proximity probes are designed to measure along its axis only and so do not measure anything in the perpendicular direction. If the two probes were not mounted 90 degrees apart then the X and Y measurements would not be independent and any subsequent orbit plot would be skewed.

The following example shows why it is important to understand the location of the transducers when viewing vibration data. If we look at a typical Trend display from PROTOR for the variation in the 1st order amplitude and phase for a pair of shaft displacement probes against speed for a rundown (coastdown) of a Steam Turbine we can see how the orientation of the transducer affects the data we see.

In the following display (Figure 2), the blue curve is for the Y-transducer and the red curve for the X-transducer for a bearing.

If we focus on the amplitude and frequency of the main peak or resonance in the curve (Figure 3) then we can determine that

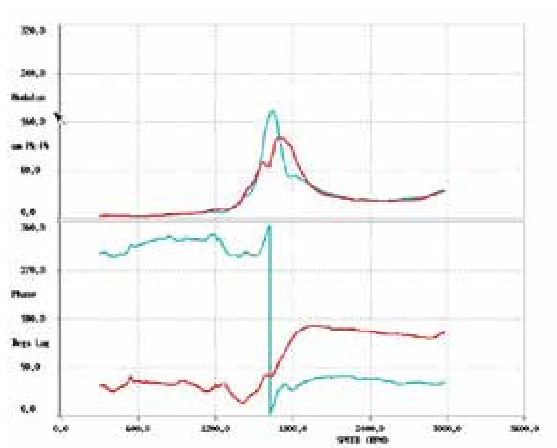


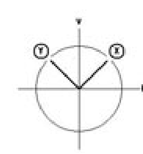
Figure 2: Trend display

the maximum amplitude for the Y transducer appears at 1640 RPM and has an amplitude of 180 um pk-pk.

Note: In the latest version of PROTOR the cursor window now has 'Next' and 'Prev' options, these move the cursor to the next or previous data scan and can be used to simply step through the measured data to find the point you want to identify.

We might assume from this that the maximum vibration amplitude experienced at the bearing would be 180 um pk-pk. However we must remember that this measurement is taken along the axis of the transducer only.

In PROTOR we are able to define the location of the pair of transducers and to then combine these into the equivalent Vertical and Horizontal measurements. If we now select the transducer orientation option to be enabled the resultant trend curves change to that shown in Figure 4.



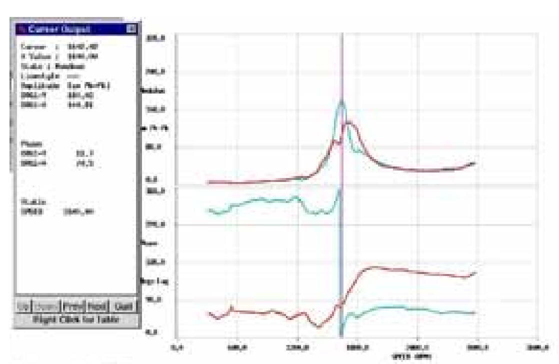


Figure 3: Trend display with cursor

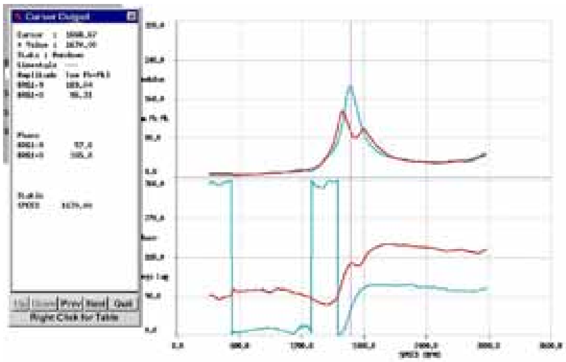


Figure 4: Trend display (with transducer orientation)

On applying the orientation translation, where the Y-Probe transforms to the vertical direction and the X-Probe to horizontal, we see that the peak amplitude in the vertical direction is now 189 um pk-pk at 1670 RPM. That is, the vibration amplitude in the true vertical direction is actually higher and at a slightly different frequency to that measured directly by the Y-probe. This identifies the true vertical critical speed for this bearing.

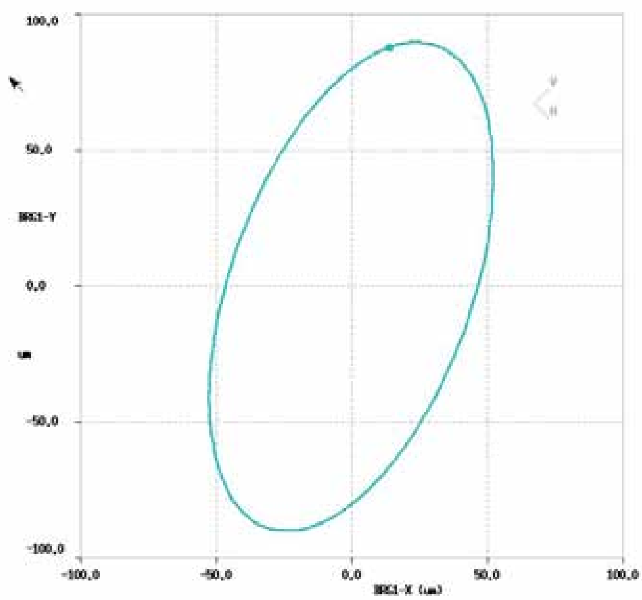


Figure 5: Orbit plot

This can also be seen by viewing the Orbit display for the pair of the transducers. As mentioned above the Orbit is the display of the two time histories collected for the pair of transducers, one represents movement in the Y-direction and the other in the X-direction. This orbit represents the movement of the centreline of the shaft within the bearing for one or more cycles. Figure 5 shows the 1st order 'filtered' orbit. Initially this is shown without any orientation correction applied, that is the vertical axis on the graph represents the vibration vector along the direction of the Y transducer.

If we now apply the orientation correction (Figure 6), the y-axis on the graph now represents the true vertical vibration and the x-axis the horizontal and the orbit has been rotated by 45 degrees.

With the Orbit display now showing the effective Vertical and Horizontal vibration and if we now step through the orbit displays for the speeds 1670 to 1640 RPM (Figure 7) we see how the orbit processes in the anti-clockwise direction as the speed

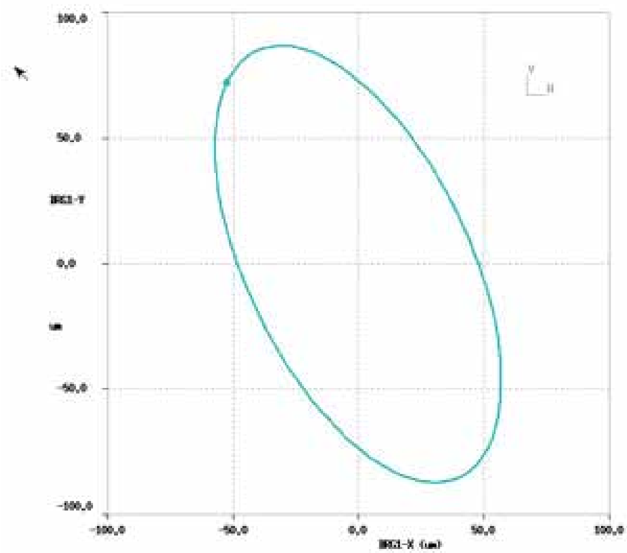


Figure 6: Orbit plot (rotated)

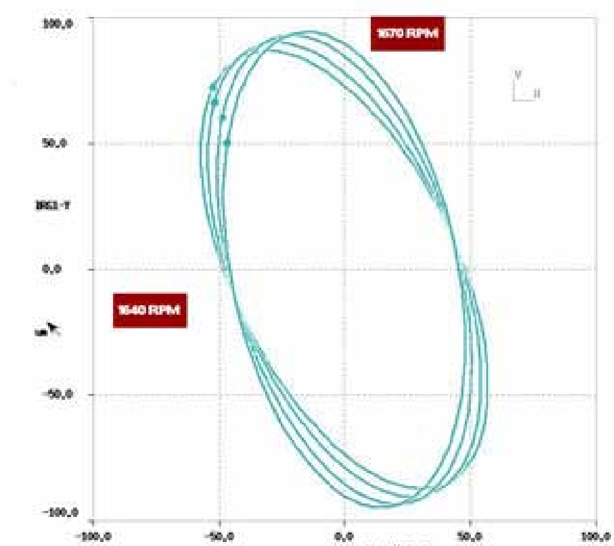


Figure 7: Orbit plot (various speeds)

decreases. At 1670RPM the orbit is at a maximum at the vertical direction but as the speed decreases the orbit major axis moves towards the axis of the Y-transducer until it reaches a maximum in this direction at 1640 RPM.

In conclusion, when viewing vibration data from a pair of shaft displacement probes always be aware of where the transducers are fitted and the conventions used for naming or labelling the signals from the transducers.

In the Real World

Pilot Exciter Bearing Failure

This failure was initially detected by a fall in first order vibration and confirmed by a corresponding increase in higher harmonics. The **PROTOR** elliptical boundary alarms detect falls in vibration as well as rises. Having detected and diagnosed a bearing failure the plant could either run to a convenient outage or quickly change preventing secondary damage.

Investigation of Road Surface Materials

The customer uses a **Prosig P8004** connected to a custom triaxial accelerometer to study tarmac surfaces. As a car moves over a road it causes a ripple in the road surface. **DATS** is used to derive displacement from the measured accelerations. The results are used to study different types of surface and changes due to humidity and temperature. The goal is to find a surface that does not flex and break, but is not too rigid.

Compressor Tie Bolt Failure

A slow rise in vibration with phase change, was seen. However, vibration levels were not high enough to trigger level alarms. Early detection by vector gradient alarm on the **PROTOR system** meant the machine was taken out of service before catastrophic failure. Investigation revealed a cracked compressor tie bolt

Flutter, Stability & Dynamic Loads

DATS hardware and software was used during the development of a high performance fighter jet. The Prosig system collected data from the aircraft telemetry system and performed Flutter, Stability and Dynamic Load analysis in near real-time durng flight tests.

Gas Turbine Rotor Disk Crack

The disk crack was seen as persistent vector change initially with a small phase change. After regular return to service **PROTOR** detected an underlying vector changed followed by exponential rise in vibration amplitude and a change in phase was observed.

In the Real World

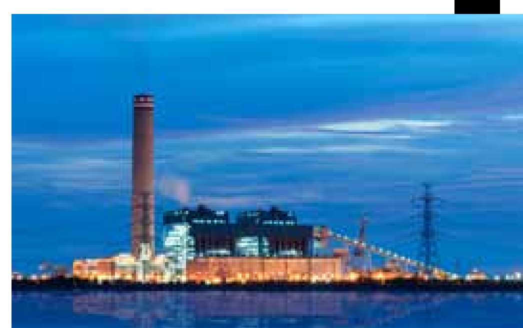
Measurement of Vibration and Pressure in Rocket Motor

The digital control lines of the firing control sequence from solid propellant rocket motors are used to control a **Prosig P8048**, which measures vibration and pressure signals. The **P8048** system is configured with a digital control module and custom acquisition software for transducer calibration, automatic data structuring and rocket test sequence measurement.

Evaluation of Vibration in Industrial Packaging Robot

A Prosig system is used to simultaneously capture CAN-bus data and vibration signals on an industrial robot. The robot is controlled by a CAN-bus and the **Prosig P8000** measures the relationship between sending commands to the robot and seeing the vibration effects caused by the displacement of the hydraulics. The combination of CAN-bus and vibration measurement make the **P8000** an ideal fit for this application.

Online Vibration Monitoring for Nuclear Power



Prosig's **PROTOR systems** have been monitoring the vibration at a particular European nuclear facility for over 25 years. The site consists of four independent nuclear reactors and the total power capacity of the site is nearly 4GW which represents around 15% of the country's power usage.

Motorcycle Helmet Compliance Testing

A weight is dropped on to a motor cycle helmet mounted on a dummy head. The acceleration of the weight and helmet is measured. Different acceleration profiles must be achieved for different test standards. The **DATS Biomechanics** software is used to check if the test has met the required profile and to verify whether the helmet meets the necessary standards.

Testing Seats Against ISO/ANSI Standards

A **Prosig P8000** system is used to measure vibrations at defined points on seats designed for commercial / industrial / agricultural vehicles. The seats are tested on a 3-axis shaker rig while suitably loaded. The **DATS Human Response** software is then used to check that the seat complies with the relevant standard.

Prosig Consultancy

Prosig provides cost effective noise & vibration solutions across a wide variety of industries. We listen and find solutions which best fit our client's needs. Quick, clear solutions are required in today's competitive environment.

Our 35+ years of client dedication has made us world leaders in solving complex noise and vibration problems. Prosig's global footprint can provide support around the world.

The same knowledge and expertise that created our hardware and software over the past 40 years are available to help solve your noise & vibration challenges.

How Do We Do It?

Modal Analysis
Rotating Machinery Analysis
Vibration Condition Monitoring
Torsional Vibration
Sound Quality Metrics
NVH & Refinement Tests
Fatigue & Durability
Hammer Impact Testing
Structural Animation
Human Response to Vibration
Crash Biomechanics
PsychoAcoustics
Source-Path and Transfer Path
Multi-plane Balancing
Flight Testing
HUMS Testing

Who Do We Do It For?

Automotive
Aerospace
Commercial Vehicles
Construction
Ship Building
Manufacturing
Acoustic Consultancy
Defence
Medical Research
Railways
Road Construction
Civil Engineering
Power Production
Steel Manufacture
Motorsport
Oil & Gas

What Do We Do?

Prosig will support you through the entire project including

- **Initial evaluation / Problem definition**
- **Defining hardware, software and technical expertise requirements**
- **Supplying hardware and software**
- **Designing and manufacturing custom hardware & software if required**
- **Performing testing, data reduction, automation and analysis**
- **Identifying potential solutions**

Host Computer Configuration for DATS Hardware & Software

In order to make full use of the facilities available, the DATS software & hardware has the following recommended system requirements.

To run the DATS software, Prosig recommend the following specification of PC hardware.

- **Intel i7 multicore processor** (or faster)
- **Microsoft Windows 10**®
- **16 GB memory**
- **PCI-E graphics adapter** (256 MB or more)
- **USB 2.0/3.0** port
- **1 GB** of hard drive space *

** This is for kit installation. Working space will be dependent on size of data files generated.*

The DATS software is supplied on a USB memory stick. Customers with a valid software support contract can download updates from prosig.com.

To use the full facilities of the DATS licensing scheme, users will need a connection to the Internet.

Other Windows® operating systems are also supported. Please contact us if you require more information.

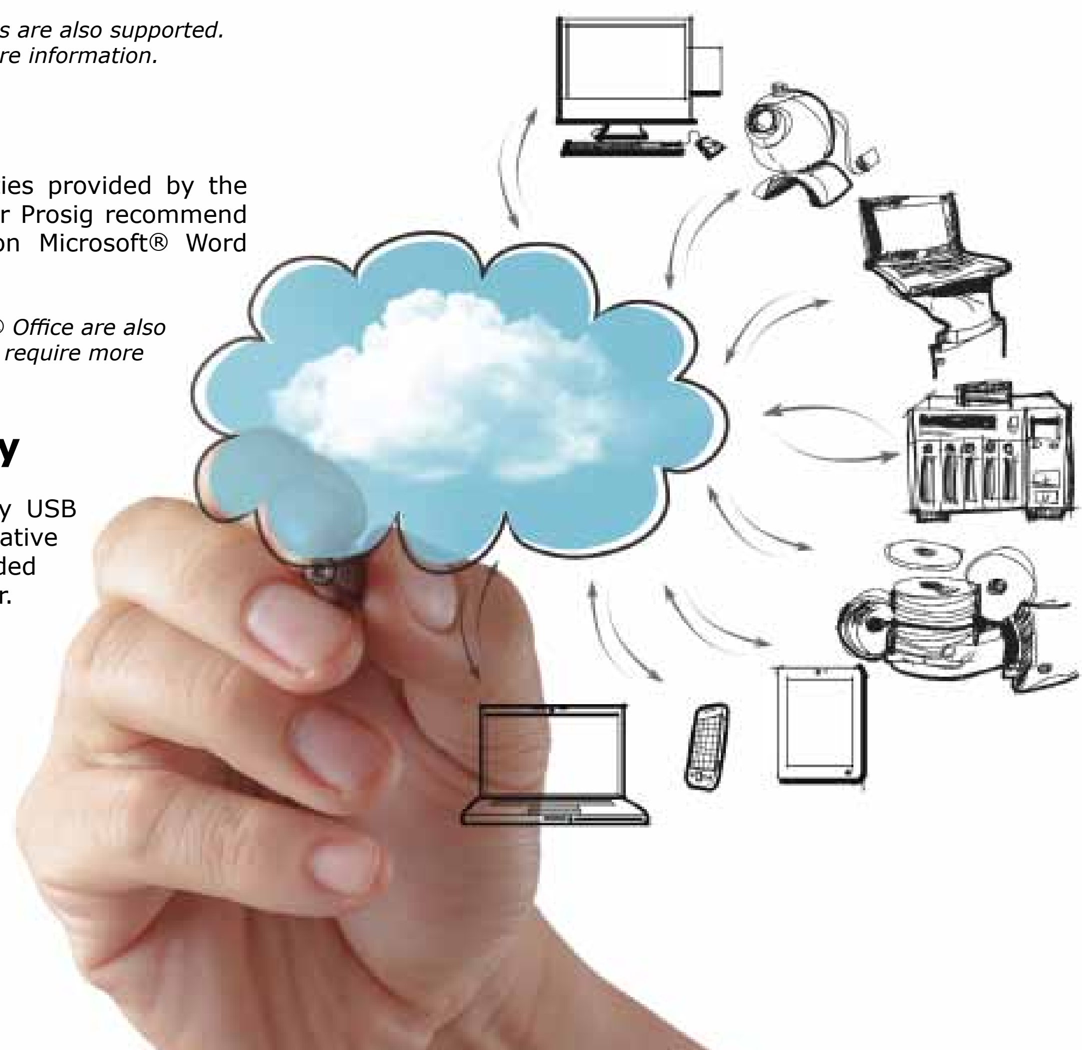
DATS Intaglio

To make full use of the facilities provided by the DATS Intaglio Report Generator Prosig recommend that the current latest version Microsoft® Word (Microsoft® Office) is used.

Some earlier versions of Microsoft® Office are also supported. Please contact us if you require more information.

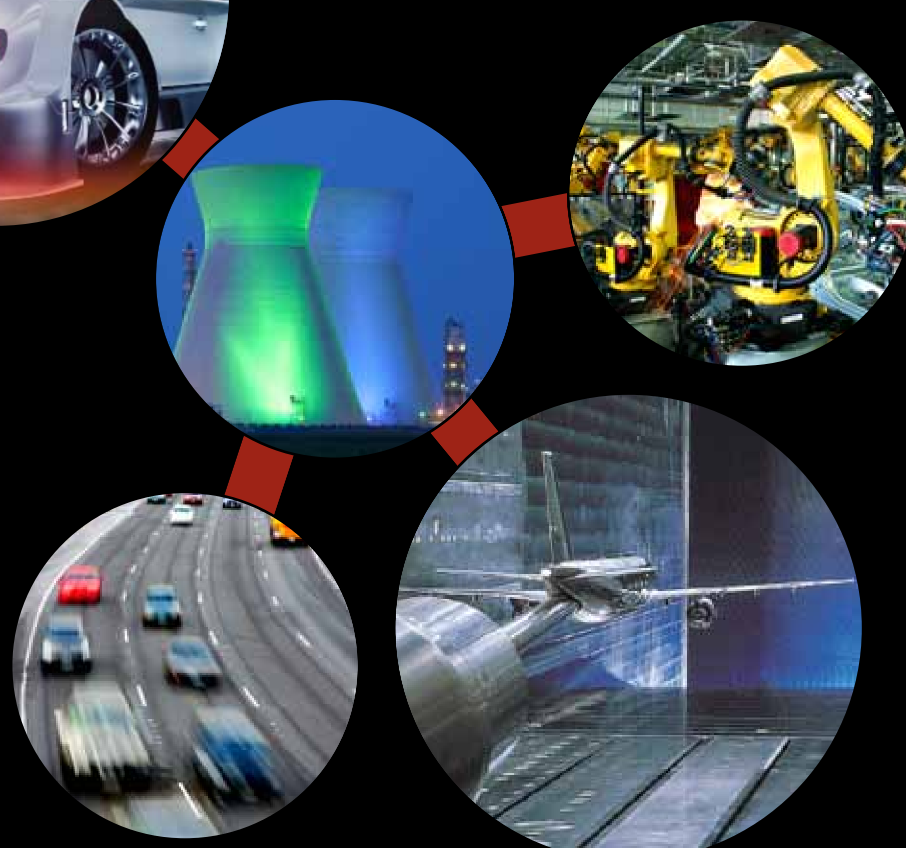
DATS Audio Replay

We recommend a good quality USB soundcard, such as the Creative Sound Blaster, and branded headphones such as Sennheiser.



"Why should you choose Prosig? We set out to help engineers, technicians and researchers obtain precise and reliable measurements. It remains our goal to make measurement simple and accurate for our partners and customers. We want you to have the best tools so that you can do the best job possible

Don Davies
Applications Group Manager
Prosig



Whilst every effort has been made to check and validate the accuracy of all information within this handbook, Prosig reserves the right to change or withdraw any product without prior notice. The information contained in this handbook may change and shall not form part of any contract.

QUALITY SYSTEM CERTIFICATION DNV·GL ISO 9001:2015

Prosig has achieved the internationally recognised ISO9001, establishing it as one of the leaders in its field. This demonstrates Prosig's commitment to customer service and quality in delivery

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Your Local Representative

The Experts In Noise & Vibration Measurement