# NOISE & VIBRATION MEASUREMENT HANDBOOK
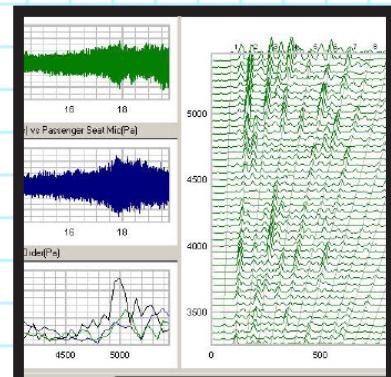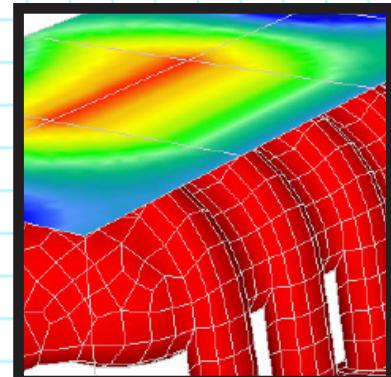
PROSIG Data Acquisition & Analysis Tools
## Third Edition

SOFTWARE PRODUCTS
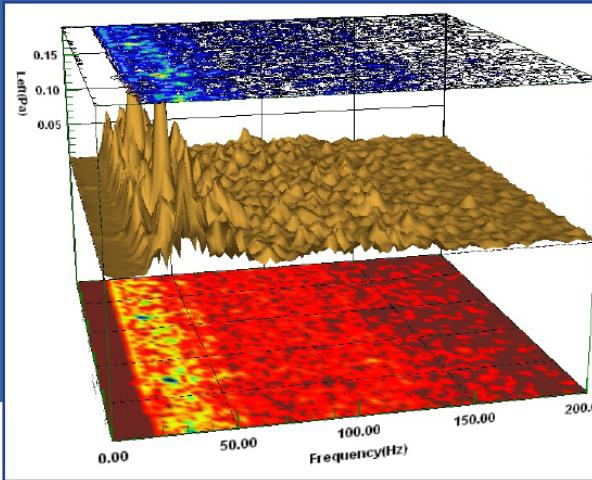
DATS.toolbox

A comprehensive set of tools for the noise 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 hardware and software, DATS is fully integrated with the Prosig P8000 series hardware. The data acquisition software
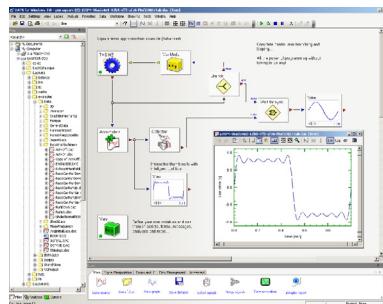
Graphical interactive and automated capture, analysis & reporting

in DATS contains everything needed to view, calibrate, monitor, and store your data. As well as capturing data using the P8000, 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.
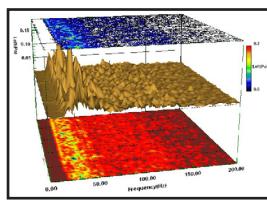
Extensive 2D & 3D graphics styles

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, colormaps, 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
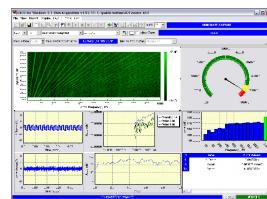

Realtime graphical displays during capture & almost instant post capture results

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.
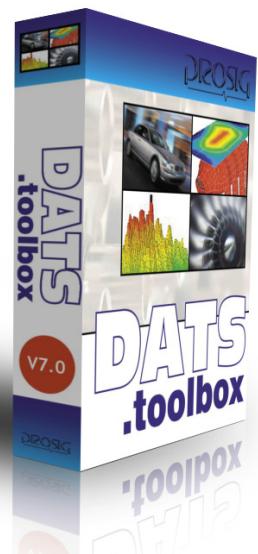
Data Acquisition (Capture)
2D & 3D Graphs
Data Arithmetic
Frequency Spectra
Time Domain Analysis
Digital Filtering
Curve Fitting
Shock Spectra
Data Generation & Synthesis
Probability Analysis
Maths, Statistics & Calculus
Data Import & Export
Built-in Report Generator
Worksheet Automation
Script (Macro) Automation

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 Analyser
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)

Training & Support

Condition Monitoring

Software

Hardware

System Packages

DATS.toolbox Software

01-55-622   DATS Professional software. Includes Intaglio reporting suite, DATS BASIC Script support and P8000 acquisition software.

The refinement of vehicles and rotating machines with respect to noise and vibration is central to creating a successful product. Not just making less noise, but also making the "right" noise can be important in the **overall success of any manufacturing endeavor**.

Using the depth and power of the DATS.nvh software suite, it is possible to determine which sources are causing the problem.

Any speed, or pulsed signal, can be used by the **extensive tacho analysis** module to analyze the signal as a spectrum waterfall. This shows the frequency content of signals as the speed changes. The skilled engineer is then able to distinguish quickly between those phenomena that respond at fixed frequencies such as **structural modes of vibration**, from those which are harmonically related to one of the rotating parts of the assembly.

Order analysis picks out those parts of the spectrum that are harmonically related and displays the data on an order axis. Slices and **order cuts** make the location of problematic sources or frequency coincidences much easier.

As these tests are often carried out on test tracks (in the case of vehicles) no two measurements will be exactly the same. **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 amount of data into standardized frequency bands that reflect the human response to noise.

There are a large number of **sound quality metrics** that can be used to further quantify noises in a way that reflects more accurately the **psychoacoustic response** of the customers. Not only the noise and vibration from the engine and powertrain, but also the squeaks and rattles can all be assessed in one objective measurement.

DATS.nvh contains all of the functions and tools available in DATS. toolbox and adds all of the features mentioned above. You can enhance the DATS.nvh package with the following options...

- **Hammer Impact Analysis Software**
- **Rotating Machinery Analysis Suite**
- **Structural Animation Software**
- **Modal Analysis Suite**
- **Time-Frequency Analysis Suite**
- **Source Path Analysis Suite**
- **Multiplane Balance Suite**
- **Sound Quality Audio Replay**

DATS.nvh

V7.0
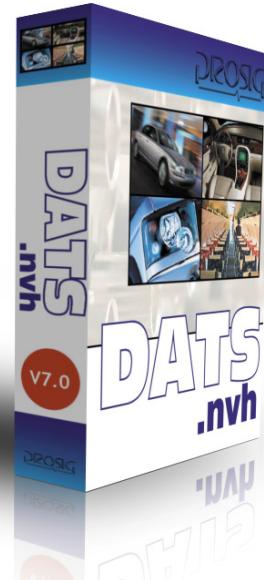
Waterfalls
Order Tracking
Sound Quality Metrics
Tacho Analysis
Waterfall Averaging
A, B, C, D Weighting
Frequency Spectra
1/nth Octave Analysis
Worksheet Automation
Script (Macro) Automation

DATS.nvh Software

01-55-1003   DATS.nvh software. Includes DATS.toolbox package and DATS NVH Analysis Suite

The analysis of rotating machinery is central to refinement activities in many fields including automotive, aerospace, power generation and general industry. It enables 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. With variable speed machines, it is a considerable challenge to reduce noise and vibration to acceptable levels. DATS.rotate contains a complete set of tools for analyzing the **sources of vibration and noise**.

One of the main tools for rotating machinery analysis is classical **waterfall analysis**, which produces frequency spectra related to the speed of rotation. The software includes comprehensive **tacho conditioning software** so that a high quality speed signal can be produced. The waterfall can optionally be produced with a phase component relative to the tachometer. This can be useful in **balancing applications**. Waterfalls can also be averaged. The software allows many different visualizations of the waterfall. Data can be displayed versus frequency or order and this is often useful in the investigation of coincident frequencies. **Order cuts and frequency cuts** can also be displayed; this gives the result of the analysis at a single order, frequency or speed. Orders can be extracted as integer ratios for the analysis of engines and motors, or at fractional orders, more useful for the analysis of gearboxes. Band-pass filtering and **envelope analysis** can be carried out for **bearing analysis**.

In addition, data can also be viewed and analyzed in the angle domain. This allows the data to be easily analyzed using a Discrete Fourier Transform to extract the orders directly, even if the speed varies dramatically during a cycle. Data that is synchronously sampled in this way can be **averaged across cycles** in the **angle domain** thus reducing noise.

Other special analyses are provided to investigate phenomena such as **torsional vibration** and **shaft twist**.

The acquisition software has additional displays available on the realtime page for use with DATS.rotate. These include order tracking, order snapshot, waterfall, speed v. time curve and needle gauge.

DATS.rotate contains all of the functions and tools available in DATS.toolbox and adds all of the features mentioned above. You can enhance the DATS.rotate package with the following options...

- **NVH Analysis Suite**
- **Multiplane Balance Suite**

DATS
.rotate

Waterfalls
Order Tracking
Tacho Analysis
Waterfall Averaging
Time Domain Analysis
Frequency Domain Analysis
Order Domain Analysis
Shaft Twist
Torsional Vibration
Worksheet Automation
Script (Macro) Automation
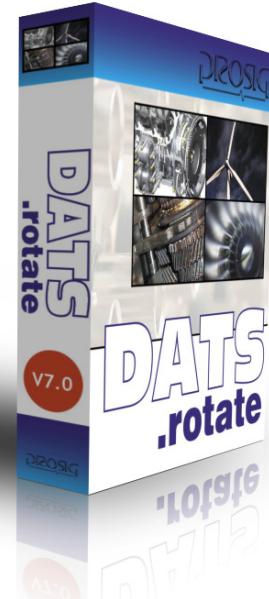
DATS.rotate Software

01-55-1004	DATS.rotate software. Includes DATS.toolbox package and DATS Rotating Machinery Analysis Suite

Training & Support
Condition Monitoring
Software
Hardware
System Packages

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+1 248 443 2470 (USA)
+44 (0)1329 239925 (UK)
or contact your
local representative
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DATS.fatigue provides a comprehensive set of programs for the **fatigue & durability test engineer**. The DATS software allows all of the fatigue analyses to be easily automated for efficient data classification.

Fatigue analysis may be broadly categorized under the following headings:

- **Cycle Counting**

- **Component Data (Material Selection)**

- **Life Prediction**

The first, and one of the most important stages is to collect a sufficiently long signal representative of the strain imposed on the structure. Typically signals are recorded from strain gauges sited near to a critical part of the component. After the signal is captured, using a Prosig P8000, the data manipulation stage begins by **finding the peaks and troughs** (turning points) within the data. This turning point data, together with the material selected from the database, may then be analyzed with one of the many life prediction functions. From the combination of the turning points and the material data the component's life prediction can now be estimated.

Fatigue life estimation is divided into two classes both of which are based upon an increment of fatigue occurring for every stress-strain cycle.

The two basic analysis categories are...

- measured stress range versus number of cycles to failure curves, (S-N curves)

- strain life and critical location methods based upon cyclic strain history

These two approaches assume that the material has effectively fatigued when a crack is initiated. This type of analysis is most appropriate for use with highly stressed parts such as a car axle where the time between crack initiation and final failure is short.

DATS.fatigue contains all of the functions and tools available in DATS.toolbox and adds all of the features described above.
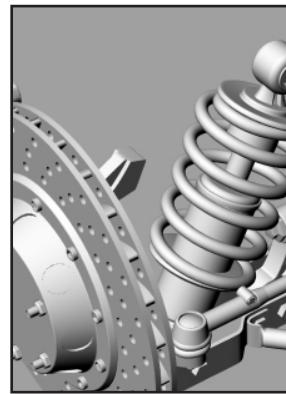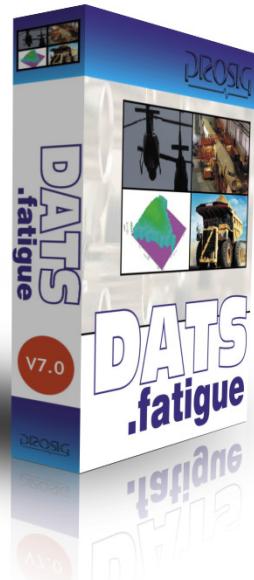
Rainflow Counting
Life Prediction
Stress Life
Weld Life
Strain Life
S-N Curves
Є-N Curves
Materials Database
Materials Editor
Worksheet Automation
Script (Macro) Automation

DATS.fatigue Software

01-55-1005   DATS.fatigue software. Includes DATS.toolbox package and DATS Fatigue Life Analysis Suite

The DATS.acoustics package is a complete and comprehensive set of functions for the study of acoustic phenomena.

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.

DATS.acoustic contains all of the functions and tools available in DATS.toolbox and adds all of the features mentioned above. You can enhance the DATS.acoustic package with the following options:-

- **Psychoacoustics Analysis Suite**
- **Source Path Analysis Suite**
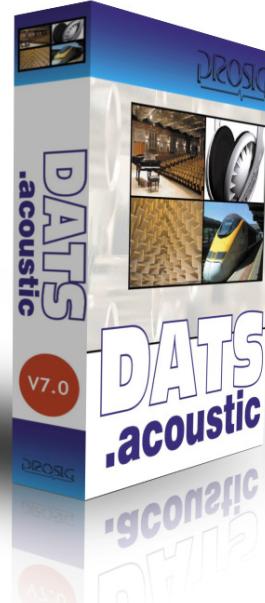
1/N Octave Filters
Sound Power
Sound Intensity
Sound Level Meter
Transmission Loss
Room Acoustics
Two Microphone Impedance Tubes
Worksheet Automation
Script (Macro) Automation

DATS.acoustic Software

01-55-1006	DATS.acoustic software. Includes DATS.toolbox package and DATS Acoustics Analysis Suite

Training & Support | Condition Monitoring | Software | Hardware | System Packages

DATS NVH Analysis Suite

- **Waterfalls**
- **Order Tracking**
- **Sound Quality Metrics**
- **Tacho 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. $1/n$th 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.
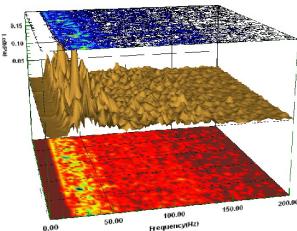
Analysis functions included

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 by Zwicker Diffuse (ISO532B)
Loudness by Zwicker Free (ISO532B)
Loudness by Zwicker Diffuse (Vehicle Biased)
Loudness by Zwicker Free (Vehicle Biased)
Loudness by Stevens (ISO532A)
Loudness Versus Time
Speech Articulation Index (ANSI S3.5 1969)
Speech Articulation Index (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

DATS Rotating Machinery Analysis Suite

- **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.
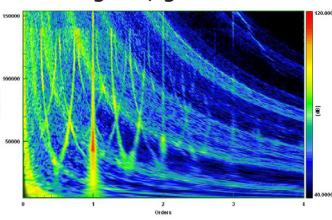
Prosig acquisition software has additional realtime displays for use with Rotating Machinery Analysis.

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

DATS NVH Analysis Suite

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

DATS Rotating Machinery Analysis Suite

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

DATS Hammer Impact Analysis Software

- **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 reject double impacts and overloads
- use of Prosig remote keypad

A pre-test Wizard assists the user in setting up trigger levels and response window weighting factors.

An initial display of Time Histories and FFT spectra enables the input signal to be checked before the Transfer Functions are calculated. Vibration data can be displayed as Inertance, Apparent Mass, Mobility, Impedance, Receptance or Obstructance. The latest measurement is shown with the accumulating average. A reference target transfer function can also be superimposed.

The measurements can be saved in numerically sequenced datasets and the user has full control over whether just the H1 or H2 is required for each response, or whether any of the associated spectra (H2, Coherence, ASD, CSD) or the time histories are to be saved.

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

DATS Structural Animation Software

- **Frequency & Time Based Animation**
- **Operating Deflection Shapes**
- **Full 3D Views**
- **Sophisticated Model Editor**
- **Built-in Hammer Acquisition Support**
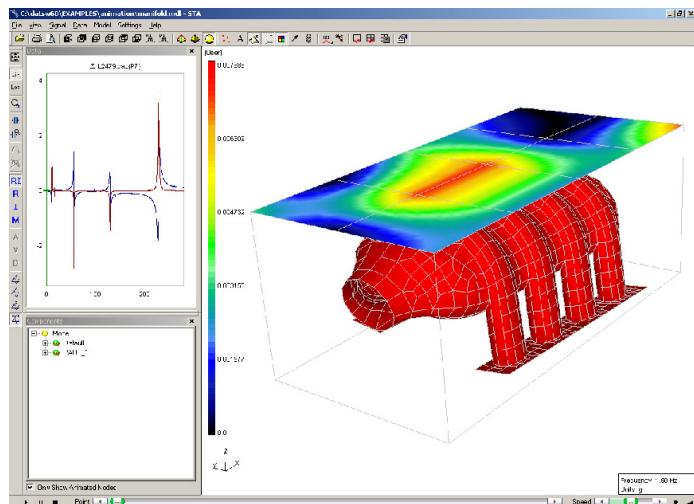
Frequency Animation uses the magnitude and phase of Frequency Response Functions (FRFs), FFTs, or cross spectra at each measurement position on the structure to reveal the motion at different frequencies.

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 Hammer Impact Analysis Software

01-55-627	Hammer Impact analysis software for P5000/P8000 (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)

PROSIG

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or contact your
local representative

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DATS Modal Analysis Suite

- **Experimental Modal Analysis (EMA)**
- **Operational Modal Analysis (OMA)**
- **Modal Parameter Identification**
- **Alternative Curve Fitting Algorithms:**
  SDOF
  MSDOF & MDOF (Frequency Domain)
  ERA-DC (Time Domain)
- **Stability Diagram**
- **FRF Synthesis from Modal Parameters**
- **Forced Response Prediction**
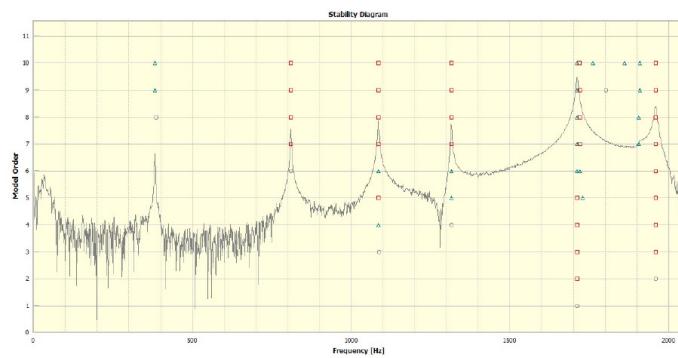
The DATS Modal Analysis suite is provided for analysts who want to determine Modal Frequencies, Damping Factors & Modal Amplitudes from either measured frequency response functions, impulse response functions or from response-only data. A variety of frequency and time domain identification methods are 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 Modal Analysis Suite

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

DATS Sound Quality Audio Replay Software

- **Playback & Compare Multiple Signals**
- **Synthesis of Audio by Filtering**
- **Filter Frequencies or Orders**
- **Vary Filter with Speed**
- **Import from WAV**
- **Export to WAV & WMA**
- **View Sound Quality Metrics**

The DATS Sound Quality Audio Replay (SQAR) package allows a user to listen to and analyze audio data. The replay software displays various data views (time series, orders, sound quality metrics, speed) with simultaneous audio output for listening. A click on one of the graphs will jump straight to a particular place or a couple of clicks will make the replay loop around a small section.

Multiple signals are loaded to form a playlist. While listening a simple click will switch between different signals. If the data is speed based the software will switch between signals at the same speed.

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. All filters have an optional speed profile. This allows varying amounts of filtering or gain to be applied to different parts of the signal according to the speed at that point.
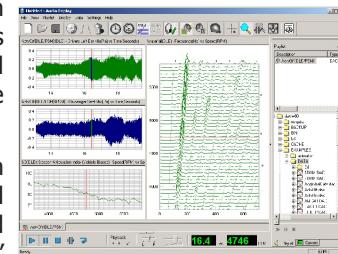
Features included

Visualizations	Sound Quality Metrics
Time Histories	**Loudness**
Time-Speed Curve	Zwicker Free
Order Plots	Zwicker Diffuse
Waterfall Plot	ISO532B & Vehicle Biased
Waterfall Color Map	Stevens (ISO532A)
Sound Quality Metrics	**Speech Articulation Index**
Real-time Speed Readout	ANSI S3.5 1969
Filters	Vehicle Biased
Order Pass	Composite Rating Performance Value
Order Reject	High Frequency Factor
Butterworth Frequency (Band Pass)	Speech Interference Level
Butterworth Frequency (Band Reject)	Spectral Balance
Filter Attenuation versus Speed	

DATS Sound Quality Audio Replay Software

01-55-717	Sound Quality Audio Replay software. (Requires 01-55-622 DATS.toolbox)
04-55-782	Prosig Sound Quality Audio Replay add-on consisting of a pair of Creative Inspire 2.1 speakers, Creative Soundblaster Audigy 2 external USB sound card, a pair of Sennheiser HD437 headphones (all items subject to availability)

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 Sound Mapping Software

- **Sound Pressure Mapping**
- **Sound Intensity Mapping**
- **Sound Power Mapping**
- **Graphical Overlays**
- **Color & Contour Maps**
- **Full Grid / Model Editor**
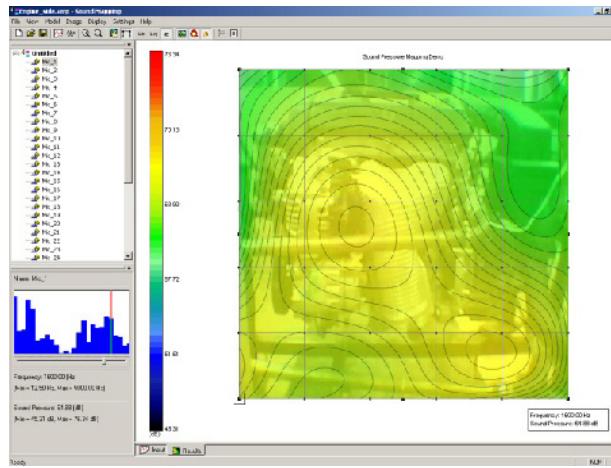
The DATS Sound Mapping software package consists of two main facilities:

The first takes sound pressure measurements from a grid of microphones (or a single microphone moved between different grid points). The measurements are then used to create either a two-dimensional map of the sound emitted by the test piece as pressures or to create a two-dimensional map of the sound power. Both options display results as color or contour maps.

The second facility takes data from an intensity probe at a series of grid points across a test piece, this data is then used to create a sound intensity color or contour map to allow visualization of the sound fields. This data can also be used to create a sound power color or contour map.

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.

Features included

Output Types
Intensity [W/m2]
Intensity Density [(W/m2)/Hz]
Power [W]
Pressure [dB] or [dBA]
Display Types
Intensity Map
Contour Map
Graphical Overlays

Bandwidth Types
Narrow Band Spectrum
Third Octave Spectrum

DATS Human Response Biodynamics Suite

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

DATS Sound Mapping Software

01-55-636   Sound Mapping Software (Requires 01-55-622 DATS.toolbox)

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Training & Support
Condition Monitoring
Software
Hardware
System Packages

DATS Multiplane Balancing Software

- **Dual Tacho Support for 4x4 Vehicles**
- **Zero Imbalance**
- **Specified Imbalance**
- **Split Weight Calculation**

Multi-plane Balancing analyzes the baseline vibrations and then the vibrations resulting from adding a trial mass (inertia) at each balance plane in turn. The software guides the user throughout the entire process. These measurements enable influence coefficients to be calculated. A least squares, optimized, multi-plane balancing algorithm, using a singular value decomposition algorithm (SVD), forms the mathematical heart of the software. An option to add bias is included.
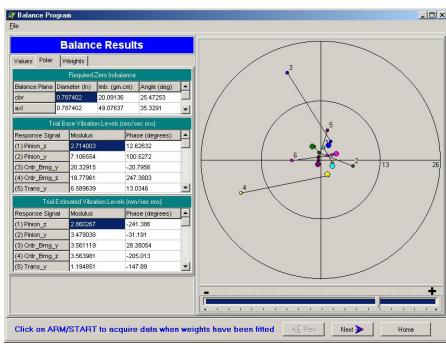
Once the influence coefficients are known the software predicts the masses required to achieve a zero imbalance or some user specified imbalance. Test runs at these conditions may be made giving the actual weights added to compare the predicted and achieved results.

The user interface is designed to automate repetitive testing, taking the user through the necessary steps to perform the baseline capture and trials capture. Shaft configurations, containing such details as balance plane name, radius, preferred trial weight and split weight locations, may be set up and saved.

A pre-balance run-up allows identification of the ideal shaft speed to perform balance. Displays include first order components for each measurement position with individual and average values. The split weight graphical display shows the required shaft positions.

Diagrams of standard shaft configurations are provided to enable easy set up of vehicle balancing applications.

Features included

Multi-plane balancing	Primary Solution
Dual Tacho support for 4x4 vehicles	Refinement by Least Squares
Split Weight calculation	Minimisation
Specified Imbalance	Synchronous Resampling
Zero Imbalance	Data Quality Checks
Bias Factor Selection	User Defined Level of Averaging
SVD (Single Value Decomposition)	Standard Driveline Configurations

DATS Multiplane Balancing Software

01-55-731 Multiplane Balancing Software (Requires 01-55-622 DATS.toolbox)

DATS Acoustic Analysis Suite

- **1/N Octave Filters**
- **Sound Power**
- **Sound Intensity**
- **Sound Level Meter**
- **Transmission Loss**
- **Room Acoustics**
- **Two Microphone Impedance Tubes**

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.

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)
Speech Articulation Index (Vehicle Biased)
Composite Rating Performance Value
High Frequency Factor
Preferred Speech Interference Level
Spectral Balance
Sound Power
Sound Power from Sound Pressure (ISO3744, ISO3745)
Sound Power from Sound Intensity (ISO9614-1)
Sound Level Meter
rms S time constant
rms F time constant
rms I time constant (peak)
rms selectable time constant
Leq with S time constant
Leq with F time constant
Leq with I time constant (peak)

Leq with selectable time constant
LN Measure with S time constant
LN Measure with F time constant
LN Measure with selectable time constant
SEL with S time constant
SEL with F time constant
SEL with selectable time constant
Misc
Nth Octave
Nth Octave RMS Output
Nth Octave Filter
Third Octave
Microphone Calibration
Spectrum Weighting (A,B,C,D)
Sound Intensity
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

DATS Acoustics Analysis Suite

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

DATS Psychoacoustics Analysis Suite

- **Loudness**
- **Sharpness**
- **Roughness**
- **Fluctuation Strength**
- **Prominence Ratio**

In simple terms, psychoacoustics is the study of the relationship between the physical measures of sound, amplitude and frequency and the human perception of them.

The DATS psychoacoustics software extension provides a host of functions for the objective description of subjective sounds.
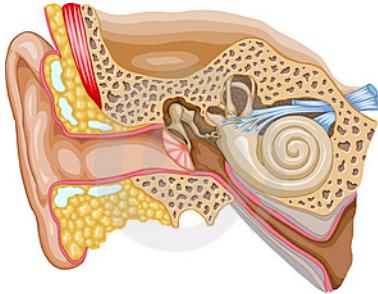
Hearing is not purely wave propagation from one medium to another, it is a sensory and perceptual event. When a person hears a sound, it arrives at the ear as a wave travelling through the air, but within the ear it is transformed into neural signals by a number of mechanisms. These nerve pulses then travel to the brain where they are perceived. Hence for many problems in acoustics it is advantageous to take into account not just the mechanics of the environment, but also the fact that both the ear and the brain are involved in a person's listening experience.

Human hearing can be compared to a spectrum analyzer - the ear resolves the spectral content of the pressure wave. This even includes phase information, which provides a significant part of the directional sensation of sound.

One good example of the power of psychoacoustics is listening to crackly, hiss-filled, vinyl records; the listener soon stops noticing the background noise, and enjoys the music, despite the presence of the hiss in the audible sound. A listener who does this often appears to forget about the noise altogether and may not be able to tell after listening if there was noise present. This effect is called psychoacoustical masking.

Analysis functions included

Loudness
Sharpness
Roughness
Fluctuation Strength
Tonality
Prominence Ratio

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

DATS Psychoacoustics Analysis Suite

01-55-997	Psychoacoustics 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.
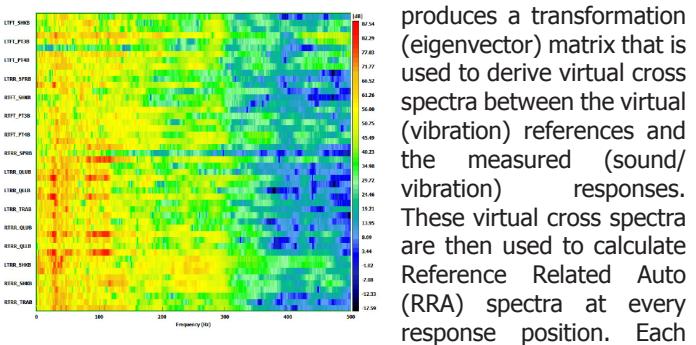
Full Transfer Path Analysis requires not only data at the Source and Response locations, but also frequency response (FRF) functions referenced to the attachment points of the vibration isolators (anti-vibration mountings). The software estimates the dynamic forces present at the isolators and determines 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)

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DATS Time-Frequency Analysis Suite

- **Wavelets**
- **Born-Jordan**
- **Wigner Ville**
- **Zhao Atlas Marks**

Two simple questions are often asked in Time Series Analysis. The first is - When did it happen? The second is - What frequency is it? These two questions can be answered by standard analyses. However, a third question - What frequency was it when it happened? - is less easily answered.
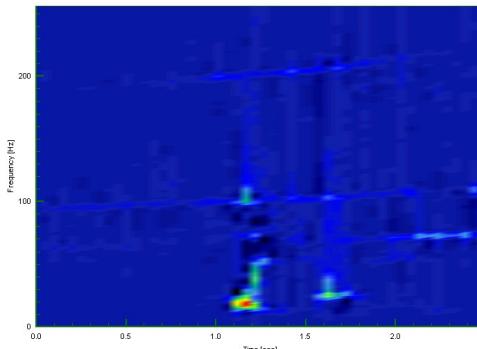
In the Time-Frequency Analysis package, Prosig have been able to implement a number of different algorithms including Wigner Ville, Atlas Zhao Marks, and Born Jordan, which all give slightly different emphasis to features of the signal.

Analysis functions included

Born-Jordan
The BORN-JORDAN module uses the Born-Jordan transform to analyze the time history signal into successive sections providing an estimation of frequency content as a function of time.

Wigner Ville
The DATS WIGNER module uses the Wigner Ville transform to analyze the time history signal into successive sections providing an estimation of frequency content as a function of time.

Zhao Atlas Marks
The 'Zhao Atlas Marks' distribution analyzes a time history into successive sections to provide an estimation of the frequency content as a function of time. It enhances the time and frequency resolution and suppresses the cross terms.

Also includes Wavelet Analysis

Mother Wavelet Generation
Wavelet Transforms
Wavelet Reconstruction
Wavelet De-noising
Wavelet Filtering

DATS Fatigue Life Analysis Suite

- **Rainflow Counting**
- **Life Prediction**
- **Stress & Weld Life**
- **Strain Life**
- **S-N Curves & ϵ-N Curves**
- **Materials Database & Editor**

The Fatigue Analysis package provides a comprehensive set of programs for the fatigue test engineer. DATS worksheets and Scripts can be used to automate the testing and analysis process from data capture to report preparation.
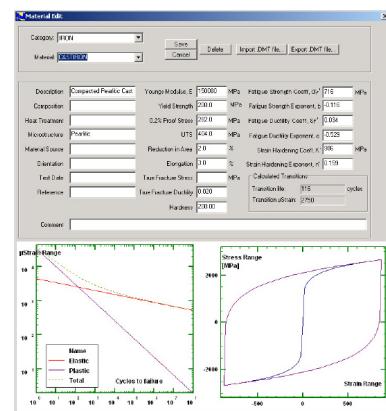
Analysis functions included

General
Materials Editor
Peak and Trough Count
Principle Strain
Principle Stress
Resolve Rosette into x,y directions
Rainflow Counting
Peak and Trough Count
Downing and Socie
ASTM 1049
Construct Histogram
Histogram Matrix Print
Histogram Matrix to Excel
Stress Life
Stress Life Fatigue Prediction
S-N Fatigue Curve Generation
Stress Life Load Sensitivity

Weld Failure Probability
Weld Life (Detailed)
Weld Life (All Classes Report)
Weld Life Load Sensitivity
Strain Life
Peak and Trough Extract
Life Prediction Cycle by Cycle Detail
Life Sensitivity with Load
Life Sensitivity with Notch Factor
Life Prediction from Histogram Dataset

DATS Time-Frequency Analysis Suite

01-55-804	Time-Frequency analysis suite (Requires 01-55-622 DATS.toolbox)

DATS Fatigue Life Analysis Suite

01-55-625	Fatigue Life Analysis suite (Requires 01-55-622 DATS.toolbox)

Standard Octave Bands

The "standard" centre frequencies for 1/3 Octaves are based upon the Preferred Numbers which date from around 1965 (see reference British Standard BS2045:1965).

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 $2^{(3/[10N\log 2])}$. 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.

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

Preferred Values 1Hz to 10Hz, 1/24th Octave

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

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

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

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

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

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

If we use the base 2 method and find the centre frequency of the third octave 10 steps below 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.

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

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

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is the AI value. This may be expressed as shown below.

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.
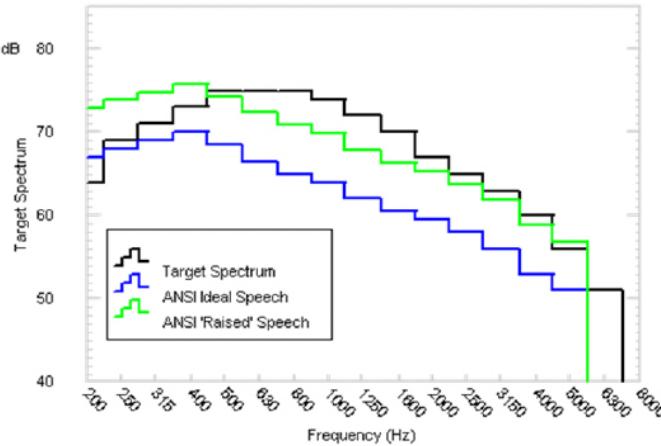
Figure 1: Standard example AI noise spectrum

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 below 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 principle 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.
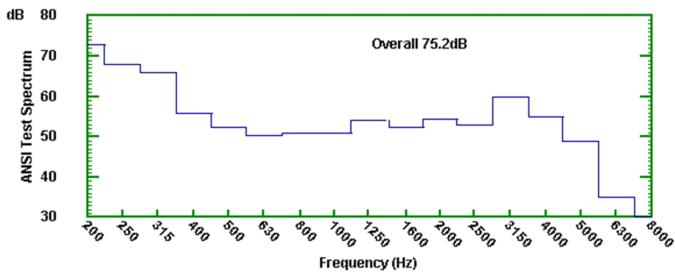

Figure 2: Third octave background noise level

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 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 and C Weighting

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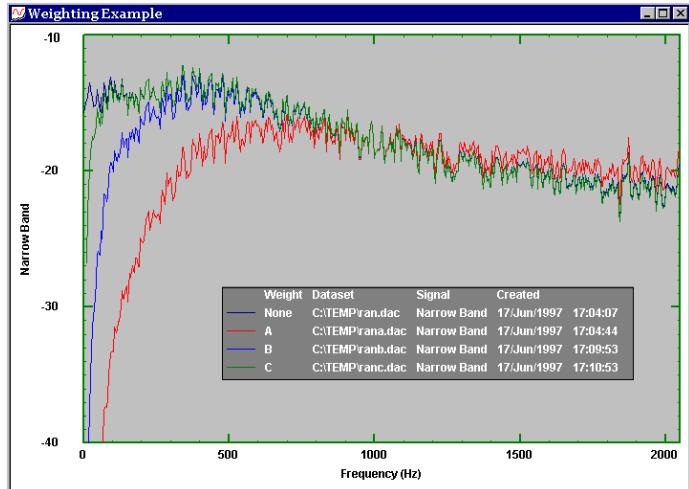
Fig. 1 : Example of A, B & C weighting

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.

Audio Equalisation Filter & Parametric Filtering

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

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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 our, 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
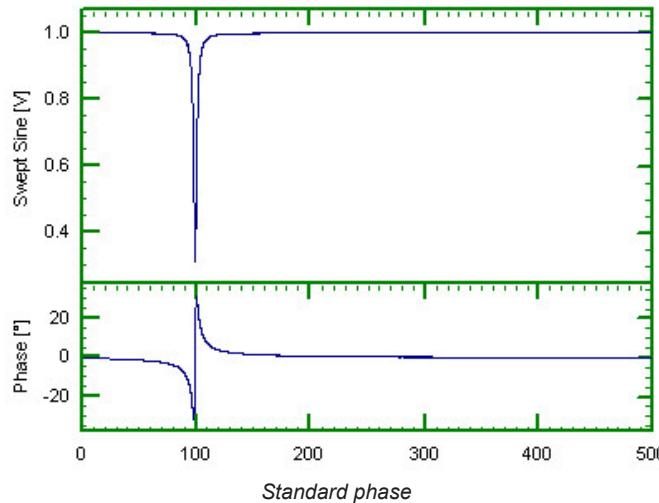
The basic component of an equalizing filter is an All Pass filter used in a feedback loop. Equalizing filters could be based upon standard filters such as Butterworth, Chebyshev and similar. However, experience in audio reconstruction suggests that a more "rounded" filter characteristic is better for 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 below. 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

"Phaseless"

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.

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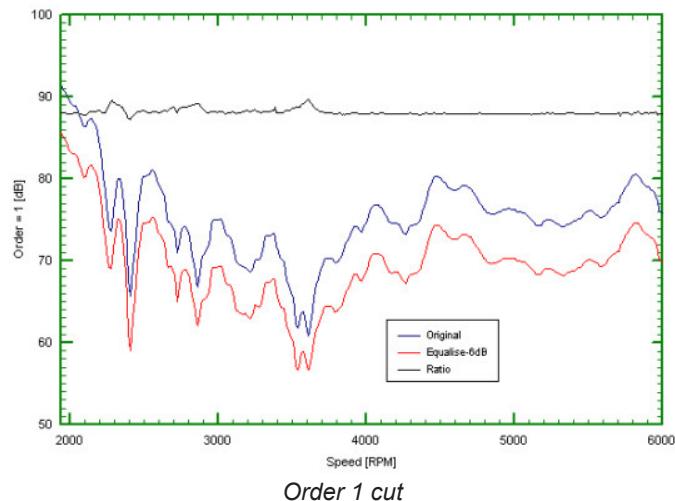
The 6dB reduction is clearly seen. The ratio (black line) between the two order cuts should be a constant, which it is except at rapid rates of change of the order. Even there the variations are generally within +/- 1dB.

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Fourier Analysis - The Basics & Beyond

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.
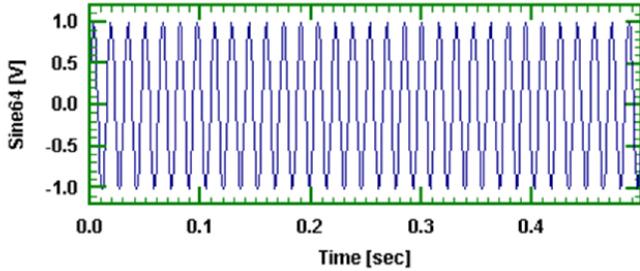

Figure 1 Sine wave, 64Hz, unit amplitude, zero phase
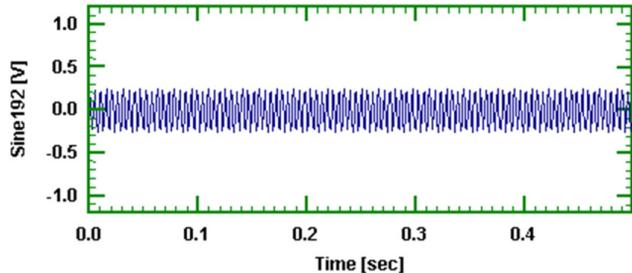

Figure 2 Sine wave, 192Hz, 0.25 amplitude, 30° phase
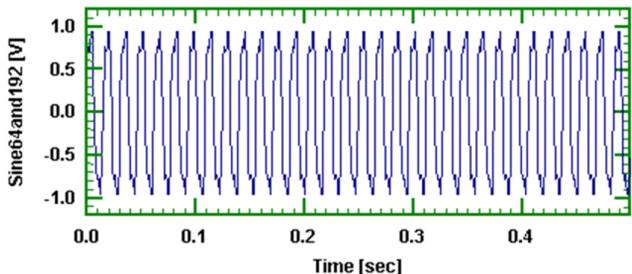
Figure 3 Combined Sine waves

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.
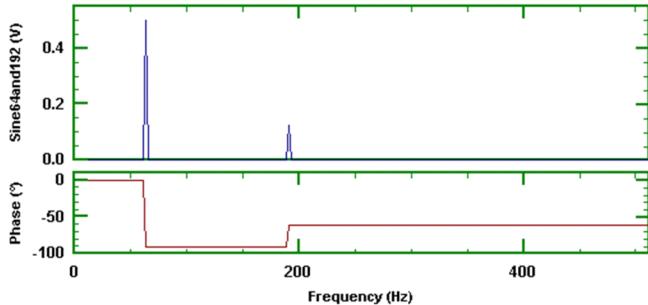

Figure 4. FFT of 64Hz & 192Hz signals

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

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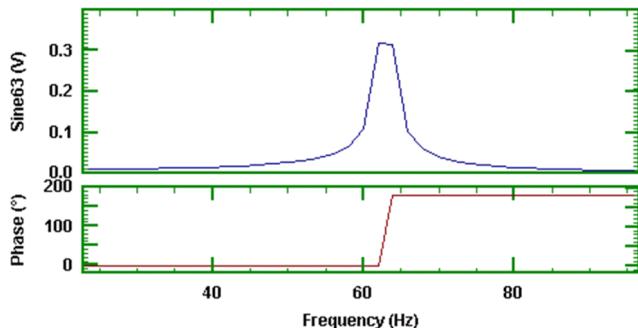
Figure 5 FFT of 63Hz, a "non-exact" frequency

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.
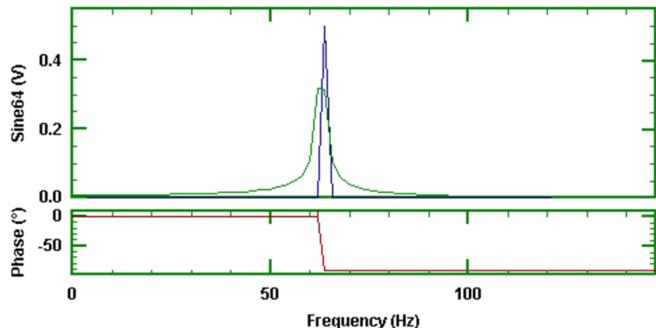

Figure 6. Overlay FFT of 63Hz and 64Hz signals

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 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.
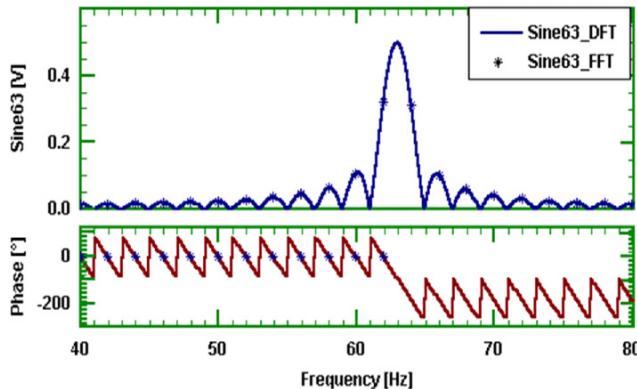

Figure 7. DFT analysis of 63Hz Sine wave

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

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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 assist 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.
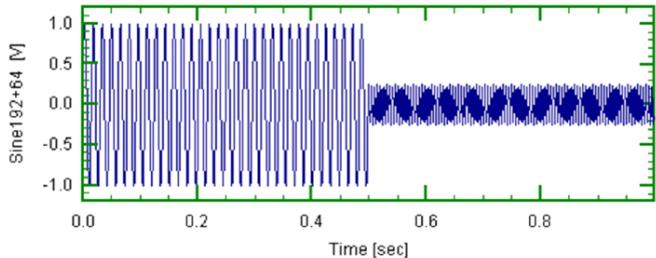

Figure 8. Two sines joined

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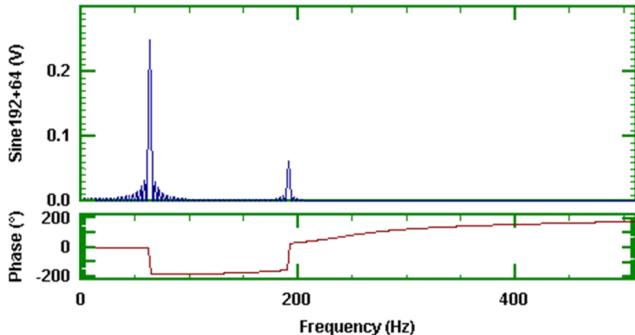

Figure 9 FFT of two joined sinewaves

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.
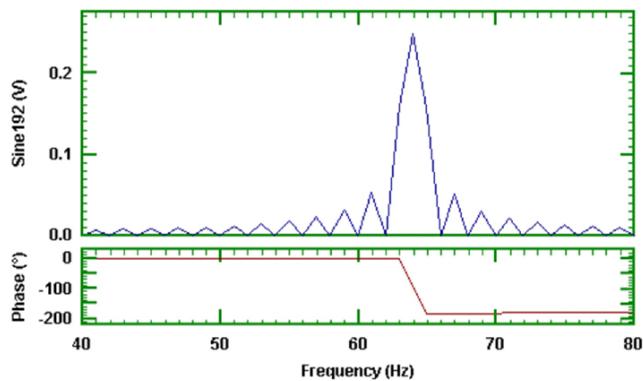

Figure 10. FFT (part) of joined signals

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.
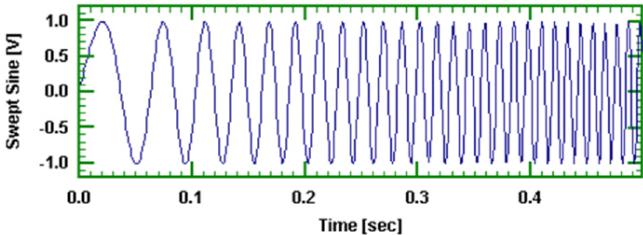

*Figure 11 Swept sine, unit amplitude, !0 to 100Hz.*
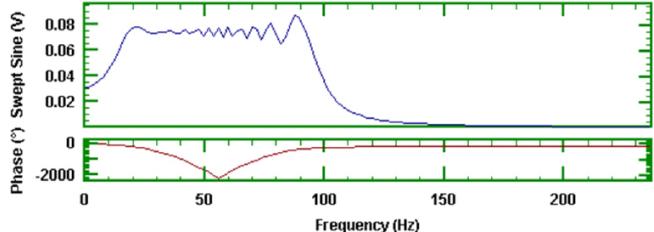
Figure 12 FFT of Swept Sine

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

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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.

DR COLIN MERCER

The articles in this section are written by Dr Colin Mercer, Technical Director, of Prosig. You can find many more articles, as well as application stories and news articles, on the Prosig Signal Processing Blog at http://blog.prosig.com. Colin Mercer is the Technical Director of Prosig and has prime responsibility for signal processing and its applications. He was formerly at the Institute of Sound and Vibration research (ISVR), Southampton University where he founded the Data Analysis Centre. He is a Chartered Engineer and a Fellow of the British Computer Society.

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