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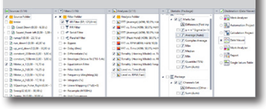
ArtemiS SUITE
ASM 01

DATA SHEET

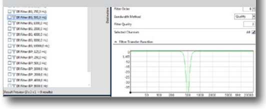
ArtemiS SUITE Basic Analysis Module (Code 5001)

Module for analysis in Pool Projects and with Automation Projects

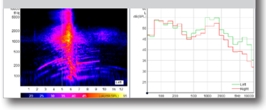
Pool Project



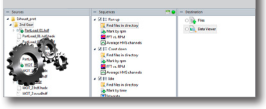
Preprocessing



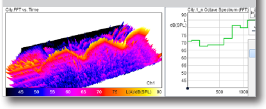
Mark Analyzer



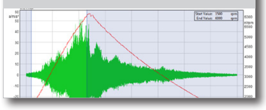
Automation Project



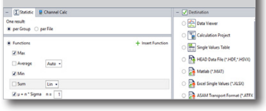
Analyses



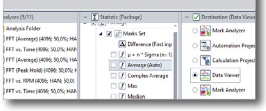
Mark Editor



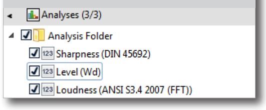
Calculation Project



Statistics



Single Values



Overview

The Basic Analysis Module provides basic analysis functions, filters, statistical operations, display options, etc.

Users can perform a wide range of measurement and analysis tasks interactively in a Pool Project or perform repetitive examination tasks by means of Automation Projects.

Furthermore, tools like the Mark Analyzer and the Mark Editor allow users to examine sounds interactively and to cut time-domain signals.

Features

Pool Project

- Pool structure (Source, Filter, Statistic, Destination Pools, Result Preview)
- Data processing based on the cross product logic principle
- Pre-processing operations, analyses (including single value calculation for 2D and 3D analyses), statistical calculations
- Frequently used, custom-configured Pool items can be saved as Pool Favorites
- Pool items can be consistently defined for multiple users (Team Favorites)
- Output of results in the Data Viewer, in the Mark Analyzer, in a Report (requires the Reporting Module ASM 02), etc.

Automation Project

- Creation and execution of Automation Projects for repetitive workflows without user interaction
- Convenient creation of linear processing chains
- Filters, analyses, statistical calculations, cutting of marks, etc.
- Channel-specific analyses (e.g., separated by airborne and structure-borne sound)
- Parametrization of processing elements via one central location
- Tolerance Check
- Automated output of results as a file, in a Data Viewer, or in a Report (requires ASM 02)

Calculation Project

- Processing of time-domain and analysis data with statistical functions

Mark Editor

- Manual or numeric cutting of time-domain data in a Pool Project by time or RPM
- Simultaneous cutting of multiple recordings by RPM
- Detection of RPM ramps and easy jumping between multiple RPM ramps within a signal

Pre-processing operations

- Pre-processing of input signals (filtering, differentiation/integration, frequency weighting, etc.)
- Non-recursive FIR filter, IIR filters, filter bank (parallel IIR filters), filter chain (serial filter elements); combination of multiple filter banks and chains is possible.

Analysis

- Basic analysis functions (FFT, Level vs. Time/RPM, third and octave analyses, Power Spectral Density, Reverberation Time, Distortion, Specific Loudness, etc.)

Statistical operations

- Processing of time-domain and analysis data with common statistical functions (minimum, maximum, mean, median, etc.)

Mark Analyzer

- Tool for simultaneous listening, analyzing and filtering (requires the Advanced Playback Module ASM 11) of time-domain data
- Use of the playback spot (requires ASM 11) to determine a suitable playback section in the diagram
- Horizontal and vertical cuts for 3D analyses
- Direct diagram export to PPTX or PDF format or as an image (PowerPoint or Adobe Acrobat need not be installed)

Single value calculation

- Calculation of single value parameters (minimum, maximum, average, percentile, sum, limits, Vibration Dose Value)
- Export of single value results to XLSX format (Excel need not be installed)

Mark Editor

The Mark Editor allows marks to be cut based on time or revolution speed (analog or digital channel). The mark limits can be conveniently adjusted with the mouse or entered numerically. The Mark Editor automatically finds the correct mark limits matching the desired RPM values and allows the user to switch easily between different RPM ramps within a signal.

For simultaneous cutting of multiple marks based on revolution speed, a table view shows the respective limits by means of a bar graph display.

Pre-processing operations

Input data can be subjected to pre-processing, e.g., in order to achieve better comparability of files recorded at different sampling rates, or to restrict the planned analysis to a certain frequency range.

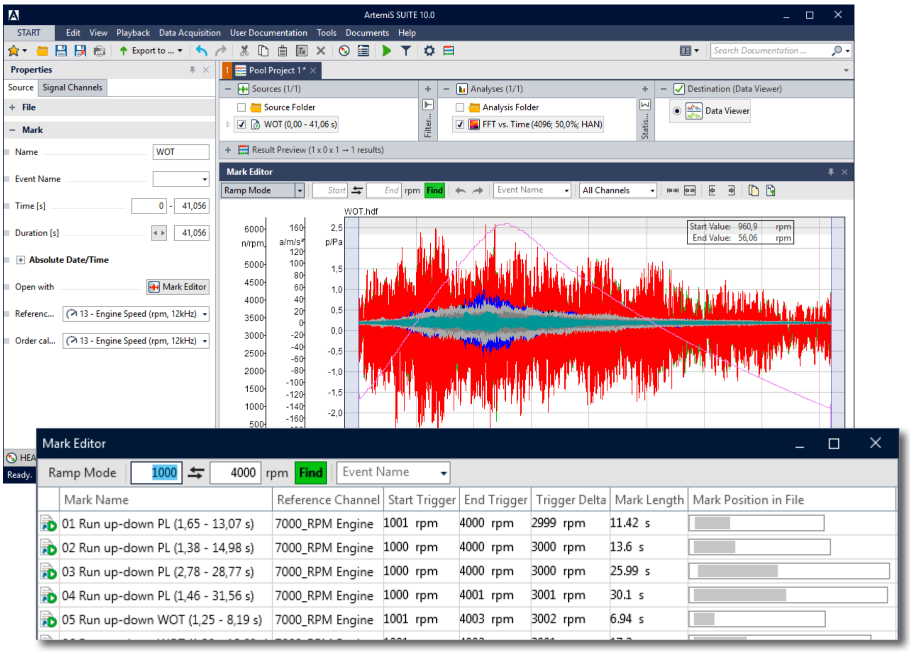
Besides individual IIR filters, the Pool Project also allows multiple configurable IIR filters to be applied in parallel in one or several filter banks. Configurable filters can also be connected in series in one or several filter chains.

Analyses

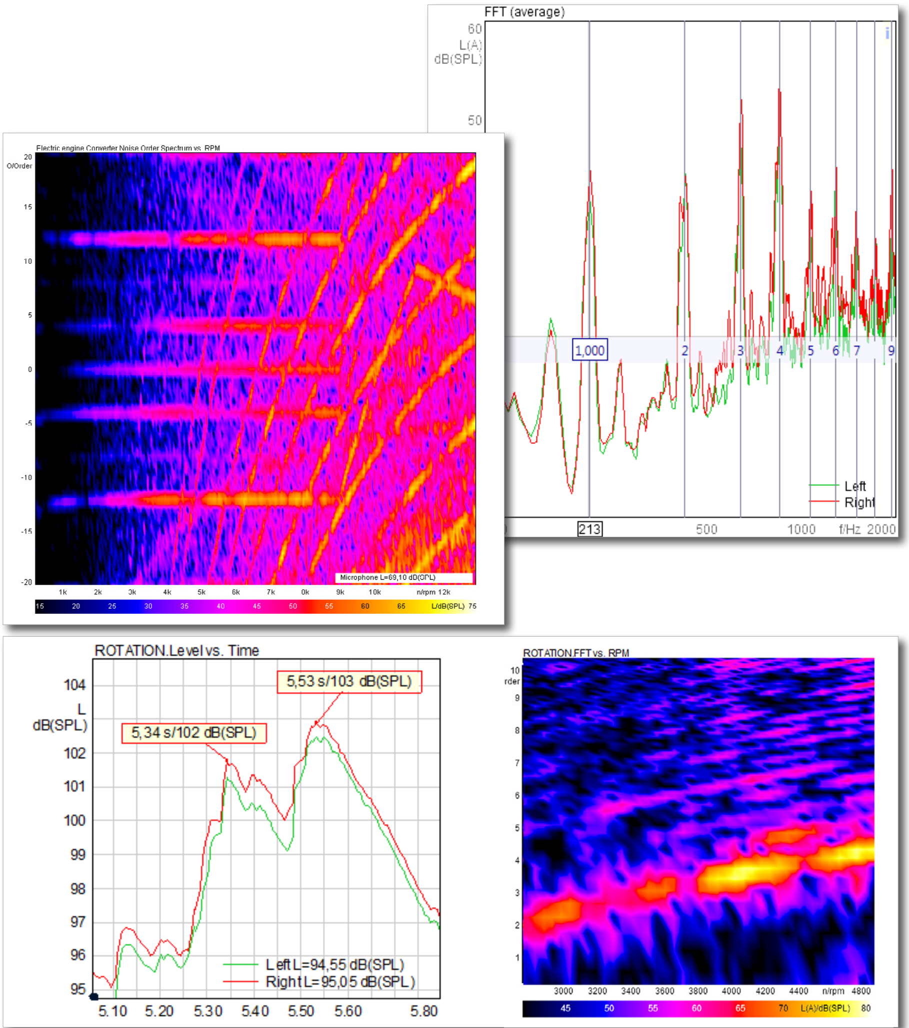
ASM 01 provides basic analysis functions for examining input data.

- FFT vs. Time, FFT (average), FFT (peak hold), FFT vs. RPM
- Level vs. Time, Level vs. RPM
- Signal vs. RPM
- 1/n Octave Spectrum (FFT), 1/n Octave Spectrum (FFT) vs. Time, 1/n Octave Spectrum (FFT) (peak hold)
- Specific Loudness
- Order Spectrum vs. Time, Order Spectrum vs. RPM
- Power Spectral Density vs. Time, Power Spectral Density (average), Power Spectral Density (peak hold), Power Spectral Density vs. RPM
- Reverberation Time, Reverberation Time vs. Band
- Harmonic Distortion, Harmonic Distortion vs. Time, Harmonic Distortion vs. Frequency
- Bypass
- Single Value Analyses: Level, Loudness, Sharpness, Metric, from Documentation, Vibration Dose Value

The analysis functions of ArtemiS SUITE can be customized to specific requirements of the user via the properties tool window.



The Mark Editor, an easy-to-use tool for time-based or RPM-based cutting of mark limits, can be opened via the context menu of a time-domain signal in a Pool Project.



Various examples of analyses with ASM 01: Order Spectrum vs. RPM with offset, FFT (average) (top), Level vs. Time, and FFT vs. RPM (bottom).

Statistics

ASM 01 allows time-domain data and analysis results to be evaluated with various statistical calculations.

For example, several marks or channels can be included in a calculation to determine an average, a maximum, a minimum, etc.

Mark Analyzer

The Mark Analyzer is used for interactive analysis and playback of time-domain data from within a Pool Project. The Mark Analyzer allows signals to be played back and analyzed at the same time, thus allowing a combined analysis with user's eyes and ears.

The Advanced Playback Module ASM 11 allows time-domain data to be filtered in real time, i.e., the effect of the filters can be heard immediately when playing the signals.

Diagram

The diagram is used in the Data Viewer and in the Mark Analyzer to display 2D and 3D data sets. Extensive editing possibilities make the diagram itself a flexible tool. Various cursors are available, which allow information to be attached to curves, abscissa and ordinate values and harmonics to be read, or single value results to be determined for any section of the diagram.

Detailed diagram settings can be configured in advance for a Mark Analyzer or a Data Viewer.

Single value calculations

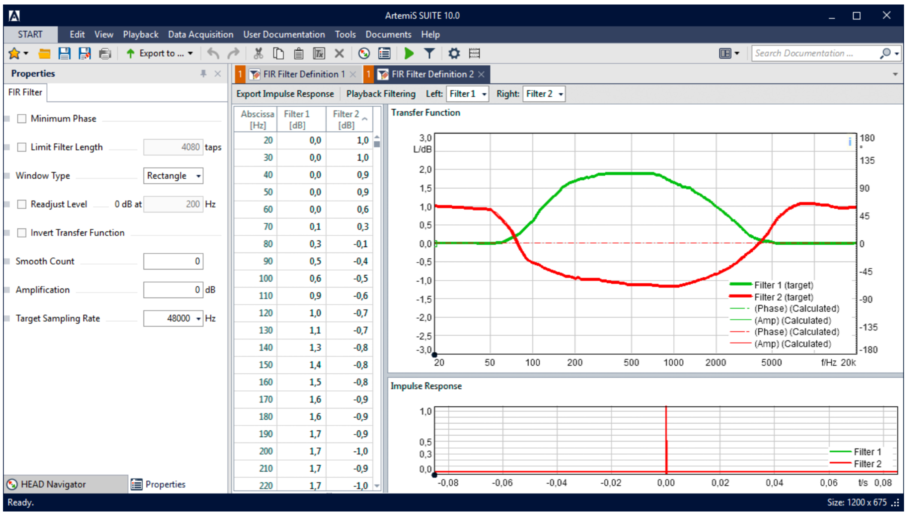
Using ASM 01, single values (min, max, average, ...) can be calculated from 2D and 3D analyses. Furthermore, special single value analyses (1D analyses) can be added to the Analysis Pool. Single value calculations allow an easy comparison or ranking of analysis results, for example.

The single value results as well as the results of tolerance checks, which are possible as well, can be displayed in the diagram, in a Single Values Table with a user-defined column layout, or in a Report (requires the Basic Report Module ASM 02).

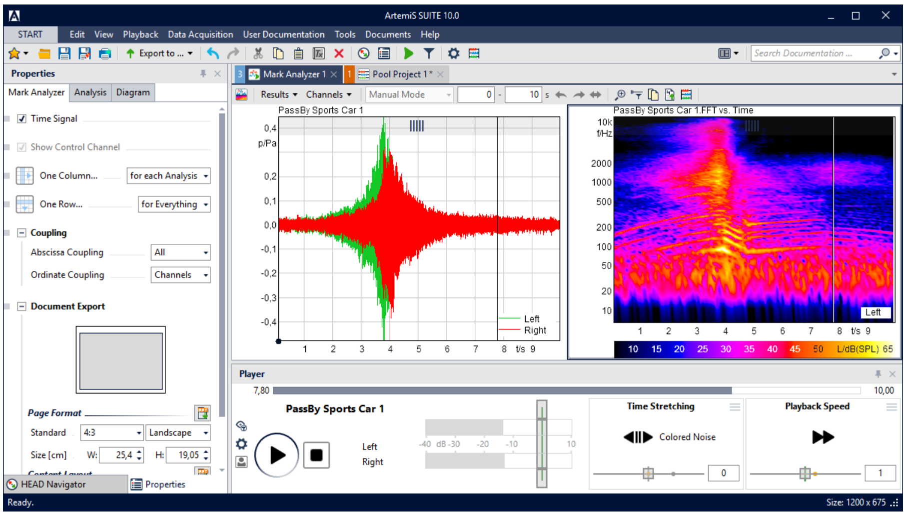
Calculation Project

A Calculation Project performs statistical evaluations of already existing analysis results. As a special feature, an adjustable smoothing function is available, e.g., for creating scatter bands.

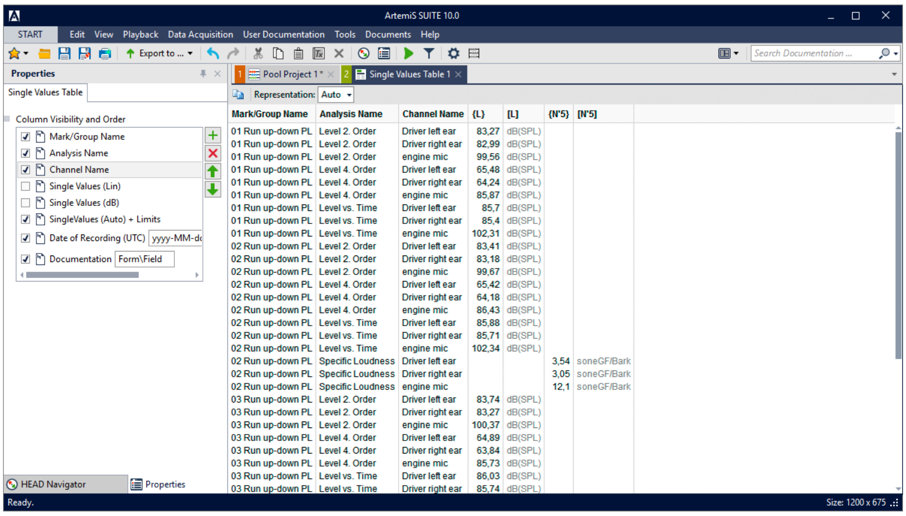
The Channel Calculation ASM 29 extends the Calculation Project so that each channel can also be processed individually via a script.



The FIR Filter Editor allows the creation and editing of transfer functions.



The Mark Analyzer is used for interactive analysis of input data. All parameters of the Mark Analyzer as well as all mark, filter, and analysis parameters an analysis result is based on can be customized.



The Single Values Table is easy to use and allows the visibility and order of the columns to be customized. The sorting order can be specified in several levels (e.g., first by name and then by analysis result).

Project structures

A key feature of ArtemiS SUITE is its project-oriented workflow structure based on the pool principle.

The individual workflow steps and the processing sequences are represented in a transparent manner, making them easily operable by the user. Various functions, such as multi-selection, text search when selecting elements, or sorting and filtering options in the Source Pool, facilitate work even with large amounts of data.

Pool Project

A Pool Project consists of five clearly structured pools. All time-domain signals and tools for sound analysis are compiled in these pools. Users configure their projects interactively and keep track of everything even with complex tasks.

For the calculation, marks and channels can be sorted and activated in the Source Pool, the required filters, analysis functions, and statistical methods are specified in the Filter, Analysis, and Statistics Pools, and the display and export options for the output of the results are configured in the Destination Pool.

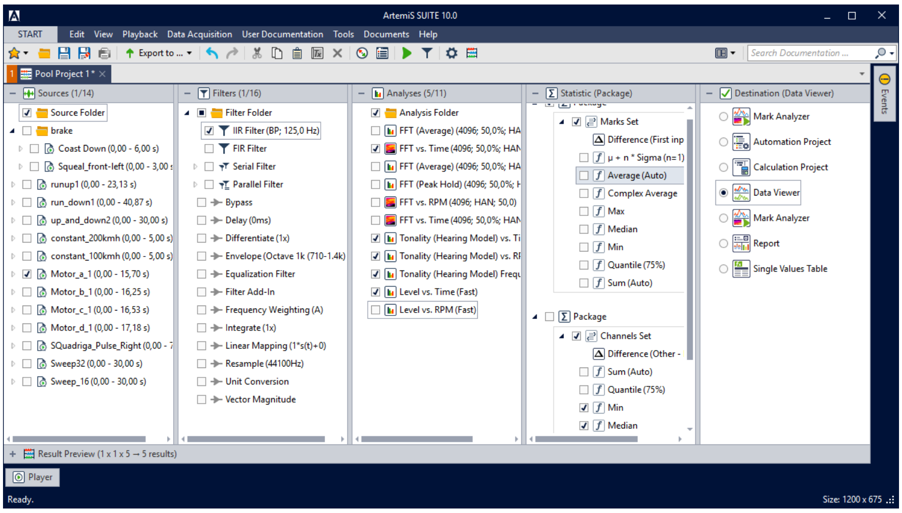
Automation Project

Automation Projects are ideal for measurement and analysis tasks that need to be performed in a repetitive way without user interaction.

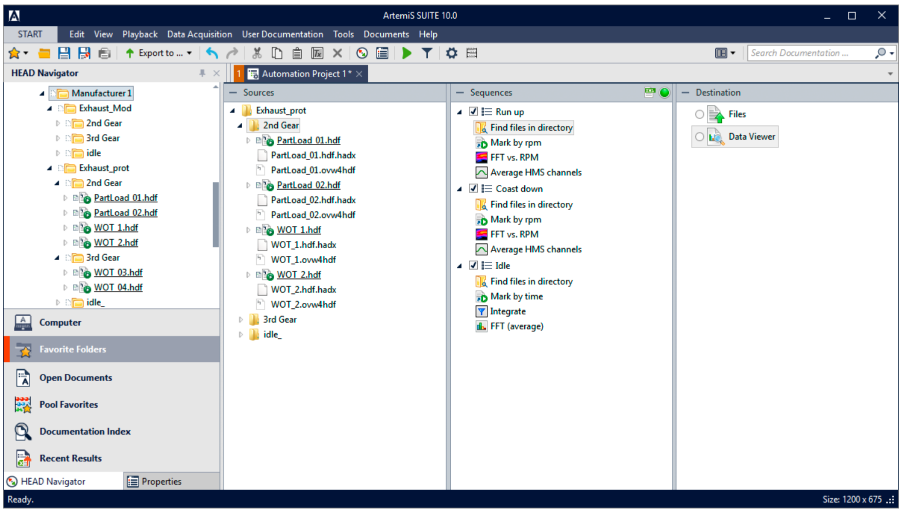
The Automation Project consists of three pools. It can either be created by the user or generated by ArtemiS SUITE from an existing Pool Project. The first pool contains the data to be analyzed, whereas linear processing chains of custom-configurable elements are defined in the second pool. These chains can contain functions like selection, cutting, filtering, analysis, calculation of a single value, import/export, etc.

For example, an Automation Project allows a channel-specific (separated by airborne and structure-borne sound) analysis of time-domain data by means of two processing chains: one for the processing of airborne sound channels (A-weighting and FFT analysis) and one for the processing of structure-borne sound channels (integration instead of frequency weighting and FFT analysis).

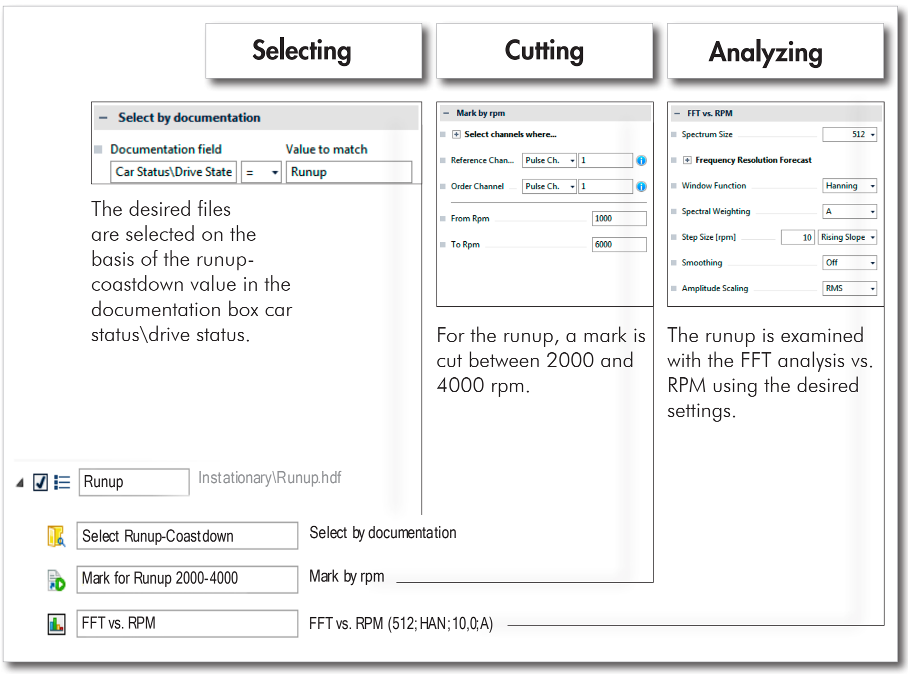
The output of the results is configured in the Destination Pool: output to a new file, a Data Viewer, or a Report (requires ASM 02).



The pool structure reflects the major steps in the execution of a project: data acquisition, processing (e.g., filtering), analysis, and statistical post-processing.



Data to be analyzed in loaded into the first pool of an Automation Project, processing chains are specified in the second pool, and the representation of the results is configured in the third pool.



Example of a processing chain with three elements: The desired files are selected, then cut according to the specifications, and finally the resulting marks are analyzed with the third element.

Data Pool
- Clear presentation of an unlimited number of marks/channels
- Comfortable channel selection, e.g., according to physical quantities, sampling rates, etc.

Calculation Project
- Postprocessing or further processing of input signals from the Destination Pool

Analyses
- FFT vs. Time/(average)/(peak hold)/vs. RPM
- Level vs. Time/vs. RPM
- Signal vs. RPM
- 1/n Octave Spectrum/(peak hold)/vs. Time
- Specific Loudness
- Order Spectrum vs. Time / vs. RPM
- Power Spectral Density (average)/(peak hold)/vs. Time/vs. RPM
- Reverberation Time/vs. Band
- Harmonic Distortion/ vs. Time/vs. Frequency
- Bypass
- Single Values (2D/3D): Loudness/ Sharpness/Level/from Documentation/Metric/ Vibration Dose Value

Statistics
- Average, Median
- Min, Max
- Difference, Sum
- Quantile
- μ + n*σ

Mark Analyzer
- Individually editable diagram
- Playback
- Direct export: PPTX, PDF, PNG, JPEG, TIFF, GIF

Mark Editor
- Cutting marks by time
- Cutting marks by RPM

Filter
- Recursive filter (IIR)
- Non-recursive filter (FIR)
- Serial filter (filter chain)/ Parallel filter (filter bank)
- Resample
- Differentiate, Integrate
- Frequency weighting/ Equalization filter
- Linear mapping/ Vector magnitude
- Unit conversion
- Delay
- Bypass

Tolerance Check
- for a violation of upper and lower threshold values

Available functions and elements for an Automation Project with ASM 01

Creation of an Automation Project
- From a Pool Project

Interactive execution of an Automation Project
- Via the respective Automation Project

Sequence Bundle
- All sequences contained in a sequence bundle are calculated in parallel

Tolerance Check
- for a violation of upper and lower threshold values

Analyses
- FFT vs. Time/(average)/(peak hold)/vs. RPM
- Level vs. Time/vs. RPM
- Signal vs. RPM
- 1/n Octave Spectrum/(peak hold)/vs. Time
- Specific Loudness
- Order Spectrum vs. Time / vs. RPM
- Power Spectral Density (average)/(peak hold)/vs. Time/vs. RPM
- Reverberation Time/ vs. Band
- Harmonic Distortion/ vs. Time/vs. Frequency
- Bypass
- Single Values (2D/3D): Loudness/Sharpness/Level/ from Documentation/ Metric/Vibration Dose Value

Statistics
- Average, Median
- Min, Max
- Difference, Sum
- Quantile
- μ + n*σ

Miscellaneous
- Point Map Data Set
- Point Map Rasterization
- Cut 2D from 3D (Rescale to Hz)
- Linear/Spectral smoothing
- Data Reduction (3D ⇒ 2D)/(3D ⇒ 3D)
- Reset abscissa

Automation Variables
- Parametrization of processing elements

Filter
- Recursive filter (IIR)
- Non-recursive filter (FIR)
- Resample
- Differentiate, Integrate
- Frequency weighting/ Equalization filter
- Linear mapping/Vector magnitude
- Unit conversion
- Delay

Cutting of recordings
- Generating marks by time or by RPM
- Creation of sections from one mark
- Generating freely configurable triggers (e.g., threshold or extremum of time signals, analysis results or filtered signals)

Execution of an Automation Project with ASM 04 and ASM 05/ASM 06

ASM 04

ASM 01 allows the integration of an existing Automation Project in the Flow Control of HEAD Recorder (with ASM 04).

The Automation Project is opened by the Flow Control so that it can be edited interactively.

ASM 05/ASM 06

Using ASM 05 and ASM 06, the functionality of Automation Projects can be extended (see data sheet ASM 05/ASM 06).

Technical Data

Filter Pool

IIR Filter

Filter Kind:	Bandstop/Bandpass/Highpass/Lowpass/Allpass
Parametric:	Bandpass/Highpass/Lowpass
Variable Amplification:	Selecting the HDF file containing the desired variable amplification information
Amplification:	Amplification for parametric filters inside the active filter area within a range of ±48 dB
Tracking:	Selectable
Tracking Order:	Selecting the desired order
Tracking Offset [Hz]:	Selecting a fixed offset to the tracking order
Filter Frequency [Hz]:	Selecting the desired center or cutoff frequency
Filter type:	Butterworth/Bessel/Chebyshev
Ripple [Chebyshev]:	Specification of the desired ripple in the range from 0.01 dB up to 3 dB
Filter Order:	2/4/6
Bandwidth Method:	Filter Quality/Hz/Bark/Order/1/n Octave
Selected Channels:	Filtering all channels/Selected channels
Equalization Filters:	Filters contained in an EQU file can be loaded

Non-recursive Filter FIR

FIR Filter File:	Selectable
Window:	Rectangle/Hanning
Filter Length Limit:	Selectable
Minimal Phase:	Selectable
Window Shift [ms]:	Selectable
Normalize 3D Filter:	A transfer function 1 (0 dB) can be referred to an adjustable position
Readjust Level:	Selectable
Invert Transfer Function:	Selectable
Smooth Count:	Selectable
Amplification [dB]:	Selectable
Selected Channels:	Selectable
Filter Transfer Function:	Display of the transfer function in a diagram (no 3D filters)

Filter Bank

Any number of filters can be configured as bandstop, bandpass, highpass, lowpass and allpass; the individual filters are applied in parallel.

Filter Chain

Any number of IIR filters can be configured as bandstop, bandpass, highpass, lowpass and allpass; the individual IIR filters are applied in series.

Frequency Weighting

Frequency Weighting:	None/A/B/C/D/G/Wd/Wk/Wc/We/Wj/Wf/Wh/Wb/Wm
Channel Selection:	All channels/All airborne channels/All vibration channels/Selected channels

Integrate

Integrate:	All channels/Selected channels
Count:	Number of integration steps to be executed
Highpass Mode:	Off/Relative/Absolute
Highpass Frequency [Hz]:	Selecting the desired cutoff frequency (relative/absolute)

Differentiate

Differentiate:	All channels/Selected channels
Count:	Number of differentiation steps to be executed

Delay

Delay [ms]:	Positive and negative values
Allow zeros at start/end:	Selectable
Selected Channels:	Filtering all channels/Selected channels

Linear Mapping

Factor:	Selecting the desired multiplier
Offset:	Selecting the desired offset
Selected Channels:	Filtering all channels/Selected channels

Vector Magnitude

First Channel:	Selecting the channel index from that on the magnitude of the number of channels defined at Vector Channels shall be calculated
Vector Channels:	Selecting the number of channels used to calculate one vector magnitude channel
Bypass:	The vector magnitude is added as additional channel after the vector channels, the original vector data are preserved

Unit Conversion

Conversion of the measurement units used for the input signal to a different unit system. This includes all units supported by ArtemiS SUITE.

Resample

Auto Select Audio Sampling Frequency:	Selectable
Sampling Rate [Hz]:	Selecting the desired destination sampling rate
Stretch Time Signal:	Stretch factor to alter the pitch of the signal

Analysis Pool

FFT (average)/FFT vs. Time (peak hold)/FFT vs. Time/ FFT vs. RPM

Spectrum Size:	16 – 2^{23}
Window Function:	Rectangle/Hanning/Hamming/ Blackman/Bartlet/Kaiser-Bessel 8, 10, 12, 14, 16/Flat-top/Gauss 8, 16, 32
Spectral Weighting:	None/A/B/C/D/G/Wd/Wk/Wc/ We/Wj/Wf/Wh/Wb/Wm/Equal Loudness/Integrate (1x) – (2x)/Differentiate (1x) – (2x)
Overlap:	Selectable
Smooth:	Off/Octave – 1/24th Octave (Intensity Averaging/dB Averaging)
Amplitude Scaling:	RMS/Peak
Max. Nbr of Time Values:	Selectable
Store DC:	Values at f = 0 and f = sampling rate/2 are stored along with the result
Phase Calculation:	Calculation of a complex spectrum/ Reference channel
Slope:	Auto Detect/Rising/Falling/Angle/ Rotation
Step Size [rpm, ...]:	Selectable
Frequency Resolution:	Table with resulting frequency rate (in Hz) for common sampling rates
Cuts:	Extracting of 2D curves from the three dimensional spectrum (Cut Mode: First Abscissa/Second Abscissa/Free selectable cuts)

Specific Loudness

Calculation Method:	DIN 45631 / ISO 532-1 / ISO 532-3 / ANSI S3.4-2007 (FFT / FFT/3rd Octave)
Combine Binaural Channels (1, 2):	Selectable
Soundfield:	Free/Diffuse
Frequency Scale:	Hz/Bark/Hz/ERB
Spectrum Size:	16 – 2^{23}
Window Function:	Rectangle/Hanning/Hamming/ Blackman/Bartlet/Kaiser-Bessel 8, 10, 12, 14, 16/Flat-top/Gauss 8, 16, 32
Overlap:	Selectable

1/n Octave Spectrum/1/n Octave Spectrum (peak hold)/1/n Octave Spectrum vs. Time

Method:	FFT Synthesis
Band Resolution:	Octave/3rd Octave/1/6 – 1/96 Octave
Row:	A/B
Spectral Weighting:	None/A/B/C/D/G/Wd/Wk/Wc/ We/Wj/Wf/Wh/Wb/Wm/Equal Loudness
Band Border Frequency:	Nominal/Octave/Decade
Spectrum Size:	16 – 2^{23}
Window Function:	Rectangle/Hanning/Hamming/ Blackman/Bartlet/Kaiser-Bessel 8, 10, 12, 14, 16/Flat-top/Gauss 8, 16, 32
Overlap:	Selectable
Max. Nbr of Time Values:	Selectable
Cuts:	Extracting of 2D curves from the three dimensional spectrum (Cut Mode: First Abscissa/Second Abscissa/Free selectable cuts)

Level vs. Time/Level vs. RPM

Spectral Weighting:	None/A/B/C/D/G/Wd/Wk/Wc/ We/Wj/Wf/Wh/Wb/Wm
Time Weighting:	Fast/Slow/Impulse/Manual (0.01 – 60000 ms), Rectangle (0.01 – 60000 ms)
Time Constant [ms]:	Selectable
Downsampling:	Selectable
Step Size [rpm, ...]:	Selectable
Slope:	Auto Detect/Falling/Rising

Signal vs. RPM

Step Size [rpm]:	Selectable

Bypass

Method:	Display of a time signal without previous analysis

Order Spectrum vs. Time/Order Spectrum vs. RPM

Window Function:	Rectangle/Hanning/Hamming/ Blackman/Bartlet/Kaiser-Bessel 8, 10, 12, 14, 16/Flat-top/Gauss 8, 16, 32
Spectral Weighting:	None/A/B/C/D/G/Wd/Wk/Wc/ We/Wj/Wf/Wh/Wb/Wm
Amplitude Scaling:	RMS/Peak
Frequency Offset [Hz]:	Off/Fixed [Hz]/Channel
Spectral Resolution [Order]:	Selectable
Width Definition:	Off/Order/Frequency/Frequency Factor/Bark
Width:	Selectable
Spectral Range [Order]:	Minimal Order – Maximal Order
Phase ref. to:	Off/Channel/Order/Pulse
Step Size [ms]:	Selectable
Time Weighting:	Fast/Slow/Impulse/Manual (0.01 – 60000 ms), Rectangle (0.01 – 60000 ms)
Time Constant [ms]:	Selectable
Order Algorithm:	Variable DFT Size/RPM-sync. Resampling/Time Domain Averaging
Reference Channel:	Selectable
Frequency Offset [Hz]:	Selectable
Cuts:	Extracting of 2D curves from the three dimensional spectrum (Cut Mode: First Abscissa/Second Abscissa/Free selectable cuts)
Slope:	Auto Detect/Rising/Falling

Power Spectral Density (average)/Power Spectral Density (peak hold)/Power Spectral Density vs. Time/Power Spectral Density vs. RPM

Spectrum Size:	16 – 2^{23}
Window Function:	Rectangle/Hanning/Hamming/ Blackman/Bartlet/Kaiser-Bessel 8, 10, 12, 14, 16/Flat-top/Gauss 8, 16, 32
Spectral Weighting:	None/A/B/C/D/G/Wd/Wk/Wc/ We/Wj/Wf/Wh/Wb/Wm/Equal Loudness/Integrate (1x) – (2x)/Differentiate (1x) – (2x)
Overlap:	Selectable
Smooth:	Off/Octave – 1/24th Octave (Intensity Averaging/dB Averaging)
Step Size [rpm, ...]:	Selectable
Slope:	Auto Detect/Rising/Falling/Angle/

	Rotation
Amplitude Scaling:	RMS/Peak
Frequency Resolution:	Table with resulting frequency rate (in Hz) for common sampling rates
Cuts:	Extracting of 2D curves from the three dimensional spectrum (Cut Mode: First Abscissa/Second Abscissa/Free selectable cuts)

Reverberation Time/Reverberation Time vs. Band

Time Constant [ms]:	Selectable
Decay Start [dB]:	Selectable
Decay Range [dB]:	Selectable
Show Correlation:	Selectable
Band Resolution:	Octave/1/3 Octave/1/6 Octave
Row:	A/B
Representation:	Reverberation Time/Level Original/ Level Regression
Frequency Range [Hz]:	Selecting the lower and upper cutoff frequencies
Frequency-dependent Time Constant:	Selectable

Harmonic Distortion/Harmonic Distortion vs. Time/Harmonic Distortion vs. Frequency

Spectrum Size:	$16 - 2^{23}$
Overlap:	Selectable
Frequency Range [Hz]:	Selecting the lower and upper cutoff frequencies
Reference:	First Harmonic/Signal Power/All Harmonics
Results:	THD/THD+N/S/N/Sum of Harmonics/Single Harmonic
Frequency Scale:	Linear/1/1 Octave/1/3 Octave/1/6 Octave

Level (Single Value Analysis)

Spectral Weighting:	None/A/B/C/D/G/Wd/Wk/Wc/ We/Wj/Wf/Wh/Wb/Wm
Remove DC:	Selectable

Loudness (Single Value Analysis)

Loudness Method:	DIN 45631 / ISO 532-1 / ANSI S3.4-2007 (FFT / FFT/3rd Octave)
Soundfield:	Free/Diffuse
Scale:	Phon/Sone
Spectrum Size:	$16 - 2^{23}$
Window Function:	Rectangle/Hanning/Hamming/ Blackman/Bartlet/Kaiser-Bessel 8, 10, 12, 14, 16/Flat-top/Gauss 8, 16, 32
Overlap:	Selectable

Sharpness (Single Value Analysis)

Sharpness Method:	Aures/von Bismarck/DIN 45692
Loudness Method:	DIN 45631 / ISO 532-1 / ANSI S3.4-2007 (FFT / FFT/3rd Octave)
Soundfield:	Free/Diffuse

Single Value From Documentation

Documentation items with numeric values can be selected from the User Documentation.

Vibration Dose Value

Spectral Weighting:	None/A/B/C/D/G/Wd/Wk/Wc/ We/Wj/Wf/Wh/Wb/Wm
Group by DOF Point:	Selectable
Limits:	Quantity/Unit

For all Analyses from ASM 01

Representation Settings:	Individual scaling of the axes in the analysis result
Add Tolerance Scheme:	Display of tolerance curves with tolerance test of the analysis result

Single Values from 2D and 3D Results

Only Single Values as Result:	Selectable
1st/2nd Abscissa Range:	Selectable
Default:	Previous versions of ArtemiS SUITE
Minimum:	Selectable
Maximum:	Selectable
Percentile:	Selectable
Advanced (Sums):	Average/Min/Max/Percentile

Definition of threshold values for whose compliance the determined single values shall be tested for.

Quantity:	Selectable
Unit:	Selectable

Recast 2D Abscissa

Transforming the abscissa of a two-dimensional data set versus time or RPM.

Abscissa Range:	Selectable
Step Size:	Selectable
Optional Manual Configuration:	Interpolation/Aggregation

Requirements

- ArtemiS SUITE Basic Framework (Code 5000)