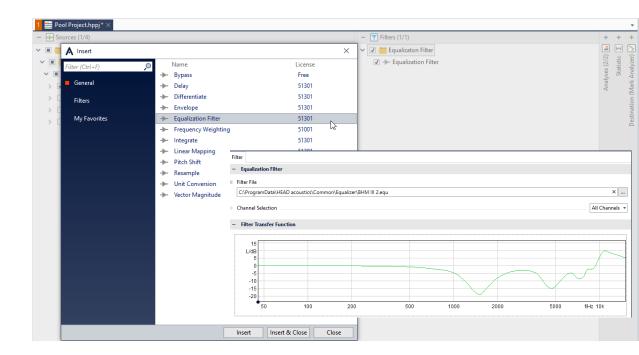


DATA SHEET



ArtemiS SUITE Signal Processing

Code 51301

ASP 301 Offline Filters

Offline Filters of ArtemiS SUITE provides filtering, editing, and pre-processing tools, that can be embedded and applied in Automation Projects, Pool Projects, and Standardized Test Projects, for example.

OVERVIEW

ASP 301 Offline Filters

Code 51301

Offline Filters makes it possible to subject input data to pre-processing, e.g., in order to better compare differently sampled signals or to be able to concentrate the subsequent analysis on a specific frequency range.

This enables, for example, a pre-processing of input signals (filtering, differentiation/integration, etc.) or the use of configurable IIR filters in parallel or serially in filter banks and filter chains in the Filter Pool of a Pool Project or with an Automation Project.

KEY FEATURES

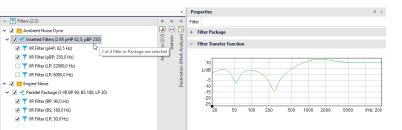
Offline Filters includes several filtering, editing, and processing tools:

- Equalization Filter
- > IIR Filter
- > FIR Filter
- Binaural FIR Filter
- > Differentiate
- > Integrate
- > Resample
- > Unit Conversion
- > Vector Magnitude
- > Linear Mapping
- > Envelope
- Pitch Shift
- › Delay

The filters and tools can be used in Pool Projects (APR 010 is required), Automation Projects (APR 050 is required), Standardized Test Projects (APR 220 is required), and Metric Projects (APR 570 is required)

APPLICATIONS

Pre-processing operations for an adjustment or modification of various signals



DETAILS

Equalization Filter

The Equalization Filter enables it to perform a channel selective equalization of an input signal. In the Filter Pool of a Pool Project, for example, the Equalization Filter can be used to compensate equalizations (Diffuse Field, Free Field, or Independent of Direction) in recordings by means of a BHM binaural head microphone or an HSU head-shoulder unit.

IIR Filter

The IIR Filter enables it to define a recursive IIR filter and apply it to an input signal. In the Properties tool window, several options, such as the filter kind, order, or type, cutoff frequency, transfer function, tracking options, ... are available for individual configurations.

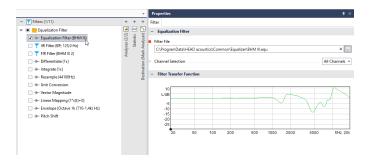
FIR Filter

The FIR Filter enables it to load an FIR filter stored in an HDF, DAT, FFT, or EQU file and apply it to an input signal. This is useful, for example, to add the correct recording equalization to recordings made without equalization and to save them. Transfer functions and other spectra can be used as filter data sets. For 2D filter files, the properties stored therein are only displayed but cannot be changed directly. Instead the editing of the FIR Filter Definition is available therefore.

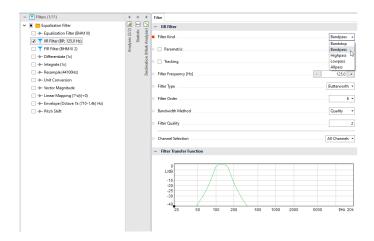


The Binaural FIR Filter can be used to generate a two-channel playback signal from a single-channel input signal by means of a binaural FIR filter. This enables users, for example, to transform a microphone recording into form comparable to an artificial head recording.

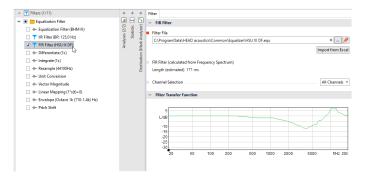
The Binaural FIR Filter is only available with an Automation Project.



Equalization Filter







FIR Filter

Differentiate

The Differentiate filter differentiates the input signal over the time. Users can select, if all channels or selected channels are filtered. The number of differentiation steps to be executed can be selected, too.

Integrate

The Integrate filter integrates the input signal over the time. Users can select, if all channels or selected channels are filtered. The number of integration steps to be executed can be selected, too.

For the correct interpretation of the results after the integration of an acceleration, the context of the recorded measure can be additionally taken into account.

Resample

The Resample filter is used to perform a resampling of the input signal, e.g., to reduce the amount of data and thereby save disk space, to match the sampling rate requirements of other devices, and to improve the comparability with other files.

Unit Conversion

With Unit Conversion users can convert the units used in an input signal into another system of units. Thereby users can, for example, have a length be converted from a non-metric system into a metric one. For a quick selection, several physical quantities listed in the Quantities and Formats Options are available for a quick selection. In addition, users can also execute any other conversion.

Vector Magnitude

With Vector Magnitude users can calculate the vector magnitude of several channels of an input signal.

For example, during the usage of triaxial acceleration sensors, the signal is measured in three spatially orthogonal directions simultaneously. Each direction is then assigned to a particular channel in the data set. Via the vector magnitude, the resulting overall absolute value of all directions at a recording point can be calculated. One application for this is, for example, the determination of the "Vibration Total Value" according to ISO 8041, which describes the human response to vibration.

Linear Mapping

With Linear Mapping users can apply simple linear mappings on single or all channels of an input signal. Users can select, if all channels or selected channels are filtered.



Differentiate

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TIR Filter (BP; 125,0 Hz)		Channel Selection	Selected Channels *
FIR Filter (HSU III DF)	Anal Destination (Mark	Selected Channels	3, 4
Differentiate (2x)			
🗹 🕩 Integrate (2x)	iteu	Count	2
Resample (44100Hz)	Desti	Integrate Acceleration to	Vibration Velocity *
Unit Conversion			
Vector Magnitude		Highpass Mode	Absolute *
Linear Mapping (1*s(t)+0)		Absolute Frequency [Hz]	10
Envelope (Octave 1k (710-1.4k) Hz)		- About require (re)	
Pitch Shift			-0

Integrate

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FIR Filter (HSU III DF)	Anal		(Mark.	Sampling Rate (Hz)	44100 -
Differentiate (2x)			n (i)		4000 ^
Integrate (2x)			natio	Stretch/Shrink Time Signal	4096
✓ → Resample (44100Hz)			Destination	Playback Speed Factor	6000
Unit Conversion					6400
Vector Magnitude					8000
→ Linear Mapping (1*s(t)+0)					8192 10240
					11025
					12000
					12800
					16000 16384
					20480

Resample

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	Integrate (2x)		inatio					mm/s^2 b	
	Resample (44100Hz)		Dest					µm/s^2	
	Unit Conversion		-					ft/s^2	
	->- Vector Magnitude							in/s^2	
					Use	r Conversions			
	— Linear Mapping (1*s(t)+0)					Source Unit	Factor	Target Unit +	
	Envelope (Octave 1k (710-1.4k) Hz)						1	Target Unit + X	
	Pitch Shift								J

Unit Conversion

-	Filters (1/11)	+ + +	Filter
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	IIR Filter (BP; 125,0 Hz)	yses Sta Ana	
	FIR Filter (HSU III DF)	Anal	Vector Channels 3
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	Resample (44100Hz)	Dest	
	Unit Conversion		
	🗹 🕕 Vector Magnitude		
	□ → Linear Mapping (1*s(t)+0)		
	Envelope (Octave 1k (710-1.4k) Hz)		
	🔲 🔶 Pitch Shift		

Vector Magnitude

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IIR Filter (BP; 125,0 Hz)	Sta		
FIR Filter (HSU III DF)	Anal	Offset	1
Differentiate (2x)		() uo	
Integrate (2x)		Offset Output = 2,5 * Input + 1 (Sample by Sample) Channel Selection	
— Resample (44100Hz)	1	Channel Selection	Selected Channels *
Unit Conversion			
Vector Magnitude		Selected Channels	6
✓ → Linear Mapping (2,5*s(t)+1)			
Envelope (Octave 1k (710-1.4k) Hz)			
🗌 🔶 Pitch Shift			

Linear Mapping

Envelope

The Envelope calculates the envelope of a bandpass-filtered input signal. The envelope is the curve which connects the maxima of a periodic oscillation. If the signal is modulated in sinusoidal form, the envelope is a sine wave. With an unmodulated sine wave signal, the envelope is a straight, level line.

Pitch Shift

The Pitch Shift filter changes the pitch of an input signal without affecting the signal length.

Delay

The Delay filter delays single or all channels of an input signal. Users can select, if all channels or selected channels are filtered.

- Tilters (1/11)	+ + +	Filter	
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IIR Filter (BP; 125,0 Hz)	yses Sta Ana	baild type	Standard Danid
FIR Filter (HSU III DF)	Anal	Bands	Octave *
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Integrate (2x)	Anal Destination (Mark	Row	B +
Resample (44100Hz)	Desti	Band Number	5
Unit Conversion			1k (710-1.4k) Hz
Vector Magnitude			
— Linear Mapping (2,5*s(t)+1)		Envelope Lowpass [Hz]	200
🗹 🔶 Envelope (Octave 1k (710-1.4k) Hz)			
🗌 🔶 Pitch Shift		V Lock Sampling Rate	

Envelope

- 🝸 Filters (1/11)	+	+	+		
r 🔳 💳 Equalizaton Filter		М	Ŋ	Pitch Shift	
	Analyses (2/2) 🖿	Statistic Z	Destination (Mark Analyzer) 🗾	Pitch Shift Factor Time Stretch Factor	1
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¶ Filters (1/12) ■ Qualization Filter → Equalization Filter (BHM III) ▼ In Filter (BH 12:30 Hz)			(Mark Analyzer) 🗸	Delay [ms] Zelay [ms] Zelay Core at StartEnd	
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			(Mark Analyzer) 🗸	Delay [ms] Zelay [ms] Zelay Core at StartEnd	

Delay

Required: APR Framework (Code 50000) and/or: HEAD System Integration and Extension (ASX) programming interfaces



Contact Information

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