Contents

Algorithmix® Analyzer PlugIn

[Overview](#) [Parameter](#) [FFT Setting](#)

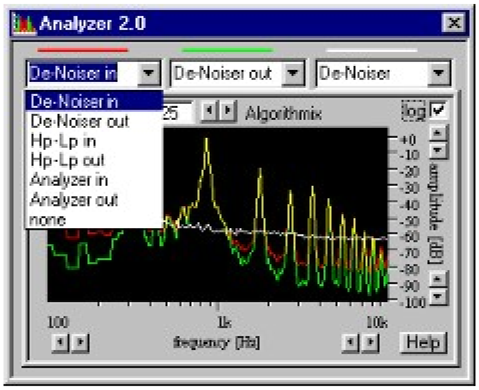

Fig.1: Screenshot of **Analyzer PlugIn** window.

Help file last modified: December 21, 1997.

Overview

The **Analyzer PlugIn** is intended as an additional graphical tool for the **Algorithmix®** frequency-domain PlugIns. It can display the frequency spectrum for two independent channels (red or green) and a frequency characteristic (filters, noise profiles in white) simultaneously. The channels to be displayed can be chosen from a drop-down box containing all available signal sources within the PlugIn chain.

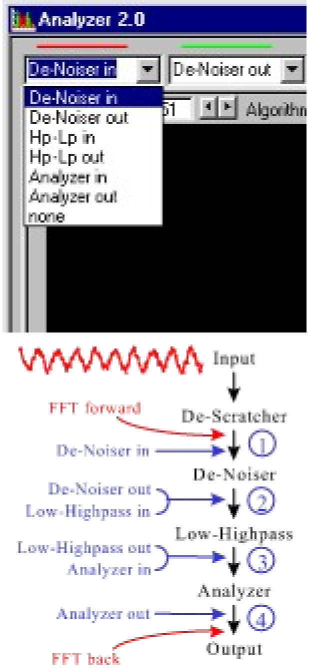

Fig.2: Signal sources available for the **Analyzer PlugIn** (left) and the corresponding points in the processing chain (right), including **De-Scratcher PlugIn**, **De-Noiser PlugIn**, **High-Lowpass PlugIn** and **Analyzer PlugIn** loaded into the **PlugIn Station**.

The PlugIns, like the **De-Scratcher PlugIn** or the **DC-Removal PlugIn,** work in time domain. Therefore they are executed before the signal is Fourier transformed into frequency domain (Point 1 in Fig.2). Thus all PlugIns working in the frequency domain (De-Noiser, Low-Highpass, Analyzer) are executed after any time-domain PlugIn. Each input channel of the **Analyzer PlugIn** can be connected with any input or output belonging to any frequency-domain PlugIn. Because the PlugIns are chained, note that some outputs are identical to some inputs and can be chosen alternatively (e.g., De-Noiser outputs the same signal as Low-Highpass in). So the six signal names included in the list in Fig. 2 (left) actually belong to four signal points of the frequency domain effect chain.

In addition to the two independently selectable signals in the display of the **Analyzer PlugIn**, the filter characteristic of the **High-Lowpass PlugIn**, or the noise profile used currently in the **De-Noiser PlugIn,** can be displayed (in white). This feature is very helpful for successful parameter set-up for the frequency-domain PlugIns. Other useful features are:
The frequency as well as the amplitude scale can be zoomed by clicking the corresponding arrow buttons.

- The amplitude axis can be switched between linear and logarithmic.
- The *decay* parameter allows the set-up of the fallback time.
- The Analyzer window can be blown up to full screen size for accurate measurements.

Help file last modified: December 21, 1997.

Parameter

To visualise the frequency spectrum in the processing chain at input or output of a frequency-domain PlugIn loaded into the PlugIn Station, select the corresponding signal source from the drop-down list belonging to the red or green display channel respectively.

Note that the Analyzer window will remain black as long as no signal is played back.

The white display channel may be set to the *filter characteristic* of the **Low-Highpass PlugIn**, or to the *noiseprofile* currently used in the **De-Noiser PlugIn.** These curves can be displayed simultaneously with any two spectrum lines pre-chosen from the drop down lists.

When loading the **De-Noiser PlugIn** prior to the **Analyzer PlugIn**, the Analyzer will be automatically set up to display the input (red) and output (green) spectrum of the **De-Noiser PlugIn** as well as the currently selected *noise profile* (white).

To zoom in or out of the currently displayed frequency range, click on the arrow buttons below the frequency scale. The same applies for the amplitude range; zooming in or out is achieved by clicking on the arrows to the right of the dB scale.

To switch between the *logarithmic* and *linear* amplitude scale, click the appropriate checkbox at the upper right corner of the window (*log*).

The *decay* parameter controls the fallback time of the display. Its optimal value has to be in accordance with the usage of the **Analyzer PlugIn**. For higher decay times, the frequency spectrum display behaves more inertly or smoothly; for the lower values the time resolution is better, but the display becomes too restless. With *decay* equal to zero, each transformed FFT block (FFT Setting) is displayed separately without any smoothing of displayed values. In general, the higher values are convenient for average spectrum measurements, while the lower if short transients have to follow. So, depending on your purpose, use the display and choose a convenient value for the *decay* parameter.

Help file last modified: December 21, 1997.

FFT Setting

To begin spectral modification of the audio material to be processed, the signal has to be transformed from time domain to frequency domain. In the current version of Sound Laundry, this is done with a Pentium-optimized fast Fourier transform (FFT) using a modified radix-4 algorithm. For maximum flexibility, two FFT-specific parameters *Blocksize* and *Overlap* are available for the user. If working with an older PC compatible computer, the FFT process can be switched to mono mode with the *mono* button to save on computing power.

The FFT settings in *FFT Properties* window are valid for all frequency-domain PlugIns that were switched to active mode. The appropriate window opens automatically when loading the first frequency-domain PlugIn..

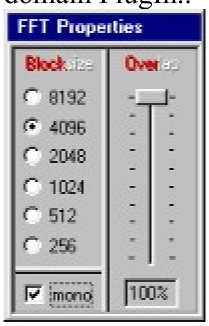

Fig.3: FFT Properties.

The *blocksize* parameter (expressed in audio samples) controls the size of the audio blocks used for the Fourier transform. As the largest possible setting of the FFT *blocksize* depends on the I/O buffer size (FFT-blocksize <= I/O buffersize), an adjustment to the I/O buffer size in the I/O settings dialog of the **PlugIn Station** may be necessary prior to setting up large FFT buffer sizes.

The larger the FFT *blocksize*, the more frequency bands are available for carrying out filter functions in frequency domain. Thus a larger FFT *blocksize* usually improves the audio quality and a smaller one increases the risk of audible artifacts. Note that the larger *blocksize* also increases the CPU load.

The time resolution of the applied filter function, however, becomes worse when increasing *blocksize*. For a static filter, as in the **High-Lowpass PlugIn,** this phenomenon is not critical. For filters dynamically changing their parameters according to the input signal (like in the **De-Noiser PlugIn**), a proper balance between *blocksize* and time resolution may be quite an important issue.

If a high repetition rate of the displayed frequency spectrum is preferred, it may be necessary to lower the FFT *blocksize* while accepting a reduced frequency resolution.

Unlike the PlugIns that apply filter functions in the frequency domain (e.g. **De-Noiser PlugIn**, **High-Lowpass PlugIn**), the *overlap* parameter is not crucial when just displaying the frequency spectrum of an audio signal. However, due to the FFT setting, it is common for all frequency-domain PlugIns that the most critical requirements must be considered if some PlugIns are chained together. In some cases it is useful to apply different FFT settings adjusted to specific signal circumstances and perform certain processing steps separately one after another with respectively optimized FFT settings .

To avoid discontinuities of the audio signal at the limits of the FFT blocks used for filtering in the frequency domain, the use of the window overlapping process is indispensable. The amount of overlapping is controlled by the *overlap* parameter. An *overlap* of 100 means that only half of the audio data is new in succeeding FFT blocks, thus each audio sample is transformed twice.

One of the most important components in the audio FFT filtering process regarding sound quality is the cross-fading operation that must be performed after inverse FFT (see Fig. 4). With **Algorithmix**® technology, the typical time-domain artifacts are successfully suppressed.

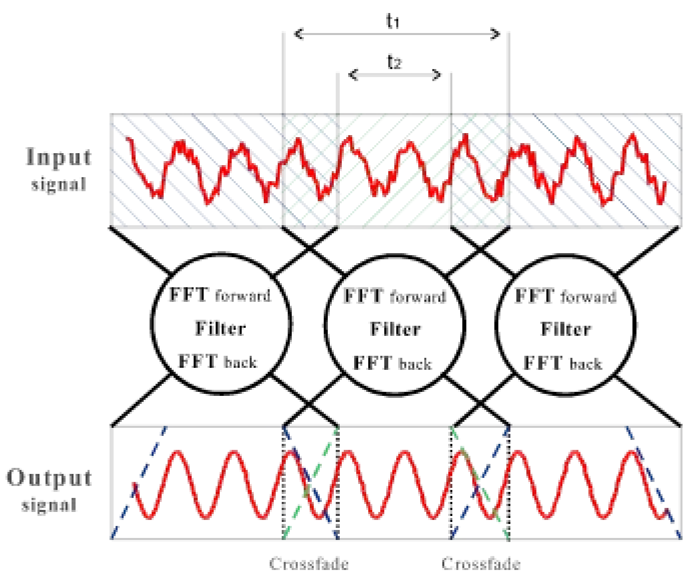

Fig. 4: Principle of the overlapping process used by the FFT in **SoundLaundry 2.0**™.

The process of overlapping used for the Fourier transform is illustrated in Fig. 4. The *overlap* parameter can be calculated from the times t1 and t2 defined in figure 4 as :

$$\text{overlap [\%]} = \frac{t1 - t2}{t1} \cdot 100 .$$

Increasing the *overlap* from 10% to 100% almost doubles the CPU load. Thus reducing the *overlap* is an effective way to save CPU power. For the final mix, however, *overlap* should be set to 100% in any case. 100% overlap prevents signal distortion, guaranteeing the lowest THD+N factor.

For taking advantage of the **SoundLaundry**™ real-time performance on slower systems, or if you're using mono files anyway, the FFT process can be switched to *mono* mode by selecting the appropriate checkbox in the *FFT properties* window.

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