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Fig.1: Screenshot of **PlugIn Station** main window.

Help file last modified: January 13, 1998.

Overview

The **PlugIn Station** is the host for the **Sound Laundry™** PlugIns from **Algorithmix®**. It is responsible for handling the input and output of .WAV audio and for loading and chaining the audio processing PlugIns.

The **PlugIn Station** includes a sophisticated .WAV file player (*Play Station*) with advanced functions like differing playback speeds, reverse playback, loop mode, real-time playback of **Algo*Rec*™** and MPEG layer3 compressed files.

The CPU Load display allows permanent monitoring of the CPU usage for all currently activated PlugIns and helps to prevent system overloads. The bargraph Level Meter with a numerical peak-level display helps you avoid potential overloads in processed .WAV files.
In general, the **PlugIn Station** supports three main .WAV audio Processing Techniques:

- wave file to (processed) wave audio output (real-time).
- wave file to a processed wave file (off-line).
- wave audio input to processed wave audio output (live, real-time).

The **PlugIn Station** supports auto-playback if started by dragging a sound file from e.g. the Explorer window and dropping it on the desktop icon of the **PlugIn Station**.

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Device I/O

The *I/O Options* dialog allows selection of an *input* and *output device*, the preferred *sampling frequency* used for real-time operations, as well as some general preferences for I/O handling and *priority* set-up.

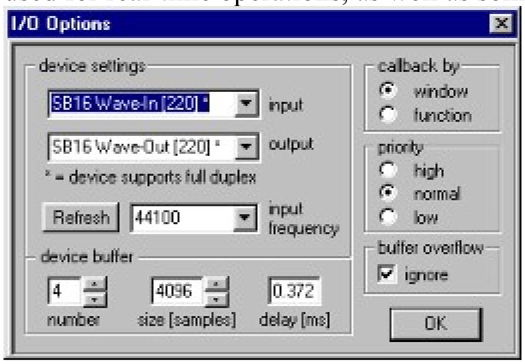

Fig. 2: *I/O Options* dialog

device settings

The *input* and *output* boxes show the available *wave device* for *input* and *output*. If only one sound card is installed in the computer, there is usually only one input and one output device available for selection. Please note that devices currently opened by other programs are not shown in the list. If any devices become available again after closing previously open program, use the *refresh* button to make these devices appear in the list. A star (*) behind the device name in the box indicates full duplex capability of the sound device, thus you may use this device for Live Processing requiring input and output of wave audio simultaneously.

The *input frequency* box allows selection of the sampling frequency used for live operation. During playback of a .WAV file, the appropriate frequency is selected automatically (is taken over from the .WAV file).

device buffer

The *device buffer* setting controls the number and size of buffers used for input and output of wave data. The corresponding delay resulting in increased system response time to PlugIn parameter changes appears in the *delay* field. In general, the *buffer size* should be increased if drop-outs occur or playback stutters.

callback by

This parameter refers to internal handling of the audio devices during wave audio input and output. The *callback by window* selection should work on all systems, but could cause playback interrupts if one of the windows is grabbed and not moved for a period longer than the delay caused by the device buffer settings.

Callback by function overcomes this problem, but due to an incorrect handling of the *callback* function by some NT audio drivers, great care should be taken when using this selection if running the **PlugIn-Station** on Windows NT 4.0 ™. With some audio drivers the program may even crash and have to be ended manually from the task manager.

priority

The *priority* setting affects the CPU time spent for graphically displaying all loaded PlugIns. The .WAV file playback itself, however, is not influenced by this parameter, because playback is controlled by the hardware interrupt, which always has higher priority than those controlling graphics.

buffer overflow

Buffer overflows occur if the audio stream is interrupted for more than the length of the I/O-buffer by e.g. excessive hard disk activity or other CPU-consuming tasks running on the computer. Sound Laundry automatically detects and reports buffer overflows by pausing playback or live-processing. In some situations, however, it may be useful to disable the automatic detection of buffer overflows. This can be done by checking the *ignore* checkbox in the *buffer overflow* setting.

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Input

For .WAV files containing two audio channels (stereo) there are four possibilities for input routing:

- *stereo* => both channels are processed separately.
- *mono* => both channels are merged in a mono signal before processing.
- *left* => only the left channel is used and copied to both outputs (left and right) after processing.
- *right* => only the right channel is used and copied to both outputs (left and right) after processing.

The advantages are obvious: both stereo channels can be acoustically compared to each other (keeping sound in the middle of the stereo image), mono compatibility can be tested, and the CPU Load can be reduced by an approximate factor of 2 (e.g., for preview or testing purposes on slow computers) without separating stereo .WAV file before processing.

For original mono .WAV files, this parameter is automatically set to *mono*.

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Output

There are two possible selections in this section (Fig.1):

- *bypass all* => all PlugIns are bypassed.
- *difference* => the only difference in sound introduced by the activated PlugIns is audible (difference = input signal - output signal).

The *bypass all* check box can be used for a quick *before/after* comparison between the original and processed audio signal. The *difference* feature supplies invaluable help for the fine-tuning of some PlugIn parameters. It makes the part of audio signal being removed by active PlugIns audible. This is extremely helpful when working with the **De-Scratcher PlugIn** or the **De-Noiser PlugIn**.

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PlugIn Center

The *PlugIn Center* (Fig.1) handles the communication between the **PlugIn Station** and the PlugIns currently installed. There are four slots available for plugging and chaining real-time PlugIns coming from **Algorithmix®** or third parties. After starting the **PlugIn Station**, all slots are free, which is indicated by the default *none* in all slots. All PlugIns installed in the system before starting the **PlugIn-Station** are automatically available in the slot drop-down lists. Every selected PlugIn becomes enabled and added to the processing chain. Its place in the execution sequence depends upon two factors:

- type of the PlugIn (pre-FFT time-domain / frequency-domain / post-FFT time-domain).
- slot number the PlugIn is selected into.

The *type* is a fixed parameter for each PlugIn and can be found in the documentation of the respective PlugIn. For example, the **DC-Removal PlugIn** and **De-Scratcher PlugIn** are *pre-FFT time-domain* PlugIns, whereas the **De-Noiser PlugIn** and the **Analyzer PlugIn** are *frequency-domain* PlugIns. The **Parametric EQ PlugIn** is an example for a *post-FFT time-domain* PlugIn.

The general rule is that *pre-FFT time-domain* PlugIns are always executed prior to *frequency-domain* PlugIns. The *post-FFT time-domain* PlugIns are executed after *frequency-domain* PlugIns. Within one group (*pre-FFT time-domain, frequency-domain, post-FFT time-domain*) PlugIns are executed according to the ascending slot number, e.g., a PlugIn in the slot number one is executed before a PlugIn in slot number 2, etc. If looking at the screenshot in Fig.1 the PlugIns are executed in the following order**: De-Scratcher PlugIn, De-Noiser PlugIn** and finally **Analyzer PlugIn**. This order remains the same if the **De-Scratcher PlugIn** would be selected in slot 4 (and not like in Fig.1 in slot 1), because a *pre-FFT time-domain* PlugIn (in this case **De-Scratcher Plug-In**) is always executed before any *frequency-domain* PlugIn.

When a PlugIn is selected for a slot, the corresponding window opens automatically. Every PlugIn window can be closed manually without unloading the PlugIn itself. By pushing the *open* button, a closed widow can be reopened anytime with the previous parameter status.

In Off-Line Processing mode, all PlugIn widows are automatically closed and the *open* button is disabled during the processing period.

The *use* check box switches a particular PlugIn *on* or *off* without unloading it. This is especially useful in combination with the *difference* feature (see Output) for checking the influence of a single PlugIn in the processing chain.

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CPU Load

The *CPU load* indicator displays the percentage of CPU time spent for hard disk operations, fast sampling-rate conversion (during cue-mode), input and output handling, and algorithm calculations.

The CPU time spent for graphic output (like the *level meter* or **Analyzer PlugIn** graphical activities) is not included in the *CPU load* measurement.

This was done to limit the *CPU-load* indication to time-critical operations only. All the graphical activities normally scale their CPU demand appropriate to its availability.

If the *CPU load* indicator shows more than 80% usage for longer than few seconds, there is a risk of drop-out during .WAV file playback, especially if other tasks are activated.

In an exceptional case, it may happen that for *CPU load* close to 100% the playback cannot be normally stopped by pushing the *stop* button in the *Play Station* window. In this case, it may be necessary to end the **PlugIn-Station** manually by pressing /*Ctrl-Alt-Del*, selecting the *PlugIn-Station* in the task list, and finally pushing the *End task* button.

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Level Meter

The *level meter* shows the left and right output level of the wave audio stream after processing by all the activated PlugIns. The green region of the *level meter* bargraph covers the levels from *-infinity* dB to -12 dB, the yellow region from -12 dB to -3 dB, and the red region from -3 dB to 0 dB.

In addition, the numerical *peak level* indicator, working independently for both channels, holds the highest peak level measured in the time period from playback beginning to the current time point. This allows level fine-tuning for the PlugIns having the capability of level amplifying (like equalizers, compressors, limiters, or maximizers). The numerical peak values can be reset any time by pushing the *reset* button.

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Processing Techniques

The **PlugIn Station** can process audio material in three different modes:

 Wave file to (processed) wave audio output (Real-Time Processing)

 Wave audio input to processed wave audio output (Live Processing)

 Wave file to new processed wave file (Off-Line Processing)

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Real-Time Processing

The *real-time processing* mode of the **Algorithmix PlugIn Station** allows real-time .WAV file processing. It means you can hear the processed file and its reactions to the parameter changes during playback. In the current version, up to four PlugIns can be chained and applied to the audio input at the same time. Due to the acoustical control option, the *real-time processing* mode is very useful for optimal parameter set-up and is always recommended before storing processed .WAV files in the Off-Line Processing mode.

The *Play Station* being an advanced .WAV file player contains the standard features plus several advanced functions intended to support the fine-tuning adjustment of PlugIn parameters, especially when working with critical audio material.

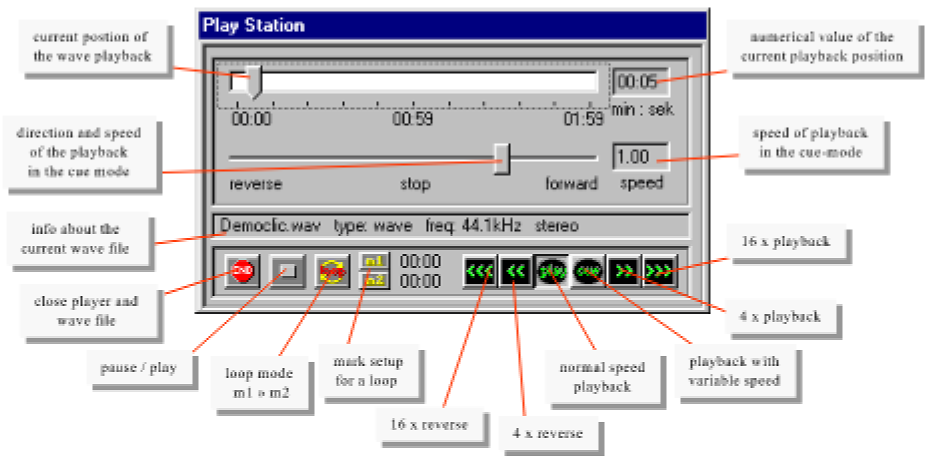

Fig.3: Overview of the player window that opens after loading a .WAV file.

After opening, the *Play Station* always starts working in the *normal* mode, e.g., *forward* playback with normal speed. The *pause/ play* button toggles between both functions, whereas the *end* button stops playback and closes the Play Station.

The upper slider can be used to *forward* or *rewind* to a particular position of the .WAV file played back. The position of the lower slider is relevant only for playback in *cue mode* (the *cue* button must be pushed) and controls the playback *speed* and *direction*.

Setting up two *marks* by pushing the corresponding buttons *m1* and *m2* during playback and then pushing the *loop* button introduces the *loop mode*. This means that the part of the .WAV file between the marks is played back in a loop. If the markers *m1* and *m2* are set to the same position (like the initial position 00:00) the whole selected .WAV file is continuously played back in loop.

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Live Processing

The *live processing* feature of the **Algorithmix PlugIn Station** provides a unique opportunity to apply real-time effects to an audio signal connected to an *input device,* and to hear the results at the *output device* without generating any intermediate .WAV file. If the same *device* is used for input and output, a full duplex driver must be installed for the sound card being used.

As in *real-time processing*, depending on how much CPU time is available, you can chain as many PlugIns between the input and output device as you like. If the activated PlugIns overload the CPU, a warning is issued and the live processing is stopped. Whenever this occurs, either reduce the CPU time needed for FFT processing (see the appropriate PlugIn description), or reduce the number of loaded PlugIns.

According to the *device buffer* setting in the device *I/O Options* dialog (see Device I/O), a delay is introduced between audio input and output. The smaller the *device buffers,* the less the *number of device buffers, and* the shorter the delay introduced. Note that too small buffers increase the risk of buffer overflow. While using some particular sound cards, the lowest setting of the *buffer size* (2048) does not work properly. Usually there is a trade-off between low delay times (down to 100 ms with some sound cards) and drop-out free operation.

The live processing can be stopped by clicking on the *STOP* button in the window popped up at the lower left corner of the **PlugIn-Station** main dialog.

A very attractive application of the *live processing* mode is the real-time cleaning of old vinyl or shellac records. Just connect your turntable (sometimes you need a proper preamplifier) to the input of your sound card and the output of the sound card to your Hi-Fi set. Load the **De-Scratcher PlugIn** (**De-Noiser PlugIn** is also recommended) and you hear the restored sound of your favor oldies.

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Off-Line Processing

The *off-line processing* mode is usually used for the final mix after all parameters have already been set up using the *real-time processing* mode. Upon pressing the *off-line calculation* button, the user has to specify two filenames in the subsequently appearing *file selector* boxes: one for input file and one for output file. Beginning with version 2.01 of the **PlugIn-Station**, the input file may be either in .WAV format or in a compressed **Algo*Press***™ format. However, the output is always in .WAV format.

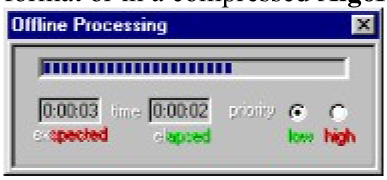

Fig. 4: *Off-line processing* window that opens after selection of an input and output file.

Before the dialog for the *off-line processing* pops up, all PlugIn windows are closed and the switches in the **PlugIn-Station** main window are locked, in order to prevent parameter changes during file processing. The *off-line processing* dialog allows selection of two *priority* levels: *low* and *high*. If there are no CPU consuming jobs running beside the **PlugIn Station**, no significant difference will be noticed between the *low* and *high* priority level selection. If other important tasks are running parallel to the **PlugIn Station** (e.g., fax or e-mail transmission, online banking, etc.) low priority set-up is recommended. In this case the *off-line processing* uses only available CPU time staying after all other normal priority tasks have been served.

The *off-line processing* can be canceled any time by closing the *off-line processing* dialog. However, in such cases no output .WAV file is generated.

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System Requirements

The system requirements mainly depend on the computational power needed by the PlugIns simultaneously used for audio processing. The **PlugIn-Station** itself easily runs on a PC 486-33, although with this CPU none of the frequency domain PlugIns is available for real-time processing.

To run the **De-Scratcher** and **De-Noiser PlugIn** simultaneously, the following minimal system is required:

Processor: Intel Pentium™ 133 MHz or AMD K6 200 MHz

Operating System: Microsoft Windows95™ or Windows NT 4.0™

Sound: Windows™ compatible 16-bit / 44.1kHz sound card

Graphic: SVGA 800x600, 256 colors

HDD: EIDE, 2 MB free for installation

Additional: Record player / pre-amplifier / cabling for digital mastering of your own records (or other audio equipment dependent on what you want to do)

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In general, you should not encounter any significant problems when using the **Sound Laundry2.0™** audio processing system. For users who want to achieve the best possible sound quality, the following remarks may be helpful:

- The better the quality of audio source material you want to process, the better the results achievable with today´s digital audio processing techniques. This means you should take great care if digitizing your analog audio material. Use a good-quality sound card (there are not too many of them) or, if possible, an external A/D converter (e.g., from a DAT recorder) linked digitally to your computer (needs a sound card with digital input). This approach is much more expensive, but it has a considerable advantage: the sometimes tremendous HF-interferences of your computer with the analog audio signal coming to a built-in sound card can be drastically reduced.

- Most multimedia boards are equipped with low-cost analog-to-digital converters that tend to produce a so-called *DC drift*. This results in the loss of dynamic range up to 30 dB and strong distortion for low-level signals. Use the **DC-Removal PlugIn** to remove the *DC-offset*.

- Unlike analog tape recorders there is no headroom above 0dB in digital recording equipment. This means that overdriven material sounds distorted and normally cannot be “repaired” in post-processing. On the other hand you should not give away available dynamic range; any unused 6dB corresponds to the loss of 1 bit. Therefore try to keep the loudest parts of your recording level just below 0 dB.

- Old analog records you wish to restore may contain clicks and crackles having a level up to 30dB above the music level. This makes it difficult to obey the rules outlined in the previous point. If you want to remove clicks and crackles with **Algorithmix® De-Scratcher PlugIn** you may proceed exceptionally and safely to overdrive your recording, as long as the music level still stays sufficiently (approx. 6 dB) below the maximum input level, 0dB. We recommend making a few versions using different input levels to be sure the useful signal is loud, but not distorted. Unlike other de-clicking/de-crackling systems available on the market, the **Algorithmix® De-Scratcher** properly deals with overdriven clicks.

- If you transfer analog audio material intended for *de-scratching,* prior to digitalization, never use any non-linear processing devices like a *compressor*, *limiter*, or *maximizer*. Any correction of the dynamic range or equalization of frequency characteristics should be performed first after the *de-scratching* and *de-noising* procedures.

- Start audio signal processing with an empty **PlugIn Station** (e.g. no PlugIns selected) using the Real-Time Processing mode for playback and carefully listen to the original audio material. If you intend to master Hi-Fi compatible CD´s from your recordings, do not use poor multimedia loudspeakers for monitoring. At least use your Hi-Fi set, or, even better, an external D/A converter (e.g., from your DAT recorder) and high-quality speakers or headphones.

- If you want to use more PlugIns at the same time, proceed gradually. Load the first PlugIn and review its influence on the original sound. If you need to remove clicks and crackles from old analog records, use the **De-Scratcher PlugIn** (sometimes after **DC-Removal PlugIn**). Follow the guidelines in the corresponding online help files. After you finished, load the second PlugIn if necessary (e.g., **De-Noiser PlugIn**) and proceed as before.

- If you are satisfied with the processing results, stop the player and select Off-Line Processing to generate a final .WAV file for further use (e.g., CD recording).
- When working in *real-time processing* mode, the CPU Load must not exceed 80% to prevent *possible drop-outs*. Special care has to be taken in the selection of the input and output devices when using the Live Processing mode to avoid buffer underrun and resulting drop-outs.

For further information about other **Algorithmix®** PlugIns and new products visit us on the Internet at:

http://www.algorithmix.com

or send e-mail to:

support@algorithmix.com

- if you need any information about installation and performance of this product.

info@algorithmix.com

- if you have general suggestions and questions concerning our product line.

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