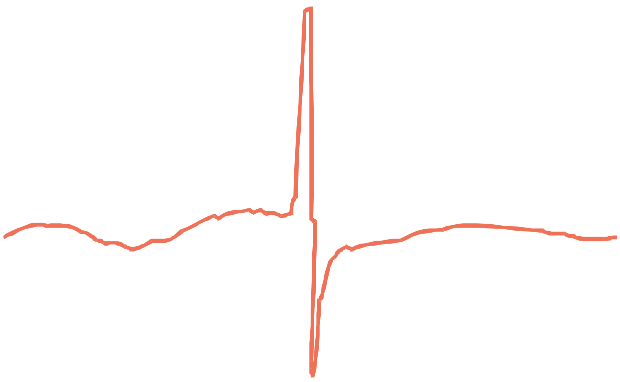
SOUND LAUNDRY ™
VERSION 2.5

OPERATOR'S MANUAL

Preliminary Version



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1. Introduction to *Sound Laundry* ™

1.1 Welcome

Thank you for purchasing **Sound Laundry**™, the world's fastest, easiest to use and ultimately sounding audio restoration software for the PC. It is specialized in high-efficiency audio cleaning without affecting original sound. Unlike other professional systems, **Sound Laundry**™ does not require any additional special hardware for real-time performance of its functions. Any current multimedia PC running MS Windows equipped with a Windows compatible sound card can be used to restore your old vinyl, shellac (78rpm), or tape recordings. You will find no preview button in **Sound Laundry**™, as all processing functions are performed in real-time during playback of your audio material. This makes parameter setup and optimizing child's play with **Sound Laundry**™. State-of-the-art algorithms using the latest achievements in the field of DSP technology and psychoacoustics ensure the superb studio quality of all our products. You are invited to compare the cleaning results out of **Sound Laundry**™ with any other audio restoration software. We believe that your choice of **Sound Laundry**™ is a good one and we wish you lots of fun and joy with the restoration of your audio treasures.

Algorithmix®, July 2001

1.2 System Requirements

The **Sound Laundry**™ is very efficient software; the basic set of the restoration PlugIns (stand-alone version) runs in real-time already on Pentium™ 166 without any problems. All current Windows™ versions (95/98/2000/MI/NT4.0) are supported. Any Windows™ compatible sound card can be used for recording and playing back sound files. We recommend, however, using high-quality sound cards, especially if transferring analog recordings to digital domain. The best results are to expect if using high-quality external AD/DA converters connected to the sound cards supporting digital input and output. To test the quality of your sound card we recommend using our **Algo*Test***™.

For transferring 78rpm or vinyl records directly from turntable a special preamplifier is necessary. It is normally a part of the Hi-Fi amplifiers. In the meantime, however, the German sound card company TerraTec offers a very convenient solution: a turntable preamplifier which is powered via game port and can be connected directly to any PC sound card.

1.3 Installation

1.3.1 Stand-Alone Version

We deliver the **Sound Laundry**™ and some demo files for testing your equipment on a CD configured for automatic start-up on current Windows™ systems. As all manufacturers of CD-recording software highly recommend switching the auto-start feature off (or they do this automatically when installing their software), the **Sound Laundry**™ set-up may not start on your system automatically after inserting the installation CD. In this case please run *setup.exe* from the root directory of the **Sound Laundry**™ CD-ROM.

1. Introduction to *Sound Laundry* ™

The setup program offers you an installation path, asks whether it should set up icons for starting **Sound Laundry**™ directly from your desktop, and finally asks for your name and the installation key. Make sure you enter the key exactly as supplied, as a wrong key will allow you to use **Sound Laundry**™ in demo mode only. The demo mode (provided only with CD version) expires after two weeks or after 30 trials.

For information purposes a special demo version of the **Sound Laundry**™ has also been prepared that is limited to three minutes processing time and has no file save function (Internet version). During installation of this version you will not be asked for a *software key*.

If you decide to remove the installed desktop icons later, you can safely delete them by clicking once with the left mouse button and pressing DEL. This will not affect the **Sound Laundry**™ program itself. To uninstall **Sound Laundry**™ completely, choose *uninstall* from the start menu or use the *add/remove* software feature of Windows. This can be found in the *system manager* group. If you simply delete the directory where you installed **Sound Laundry**™, some entries will remain in the Windows registry. That may prevent you from installing a later version of **Sound Laundry**™. The uninstall program, however, will completely remove all programs, shortcuts, and registry entries previously installed by the **Sound Laundry**™ setup.

Only one short demo wave file is copied automatically to the hard disk during the installation. If you want to use the further wave files supplied for testing your system, we recommend you copy them from the CD to your hard disk. This can easily be done with drag-and-drop using the Explorer. Please note that the demo files take considerable disk space, as they are all recorded with 44.1KHz / 16-bit stereo, which uses 173 KB per second. If you do not want to copy the files to your hard disk, you may also try to play them directly from the CD, although this may cause dropouts and playback may stutter.

1.3.2 Optional PlugIns for Stand-Alone Version

For installing the Sound Laundry™ only one software key is required, but you may later buy additional PlugIns which are not included in the basic version. In this case just click on the self extracting file xxxxxx.exe which includes the new PlugIn software. You will be asked for entering a specific *software key* related to this particular PlugIn. After successful installation the new PlugIn appears in the *PlugIn Station* list.

1. Introduction to *Sound Laundry* ™

1.4 Quick Start

After successful installation of the **Sound Laundry**™ the application can be started by double-clicking the appropriate desktop icon or from the **Sound Laundry**™ entry *AlgoPlug.exe* in the Windows start menu.

*Opening window of the **PlugIn-Station**.*

At the startup time some system checks are performed and the **Sound Laundry**™ is automatically configured for the highest performance on your system. The setup window should also show the registration information with your name.

If instead of your name the message *UNREGISTERED VERSION* appears, the user/key combination you entered during installation was wrong. In this case, you will have to reinstall **Sound Laundry**™ and supply the correct key. Just at the beginning of the new installation you will be asked to agree to delete the previous (in this case demo) version of the **Sound Laundry**™. Say “yes” and do not forget to mark “full version” (not “demo”) in the check-box.

After closing the setup dialog by clicking the rotating **Algorithmix**® logo (or just waiting a while), the **PlugIn Station** main window will pop up. Press the lower button with diskette and loudspeaker. The **Playback Station** opens automatically and starts the playback of a demo file. By pressing [LOAD] you can select a .WAV file of your choice.

The **Playback Station** offers a variety of advanced features for playback of .WAV files; please refer to the appropriate chapter in the handbook for a detailed description of all available options.

The playback can be started with the **[>]** button stopped by clicking the **[•]** button. To restart from the current cursor position the playback button **[>]** has to be pressed again. The [|•] button sets the cursor at the beginning of the wave file.

1. Introduction to *Sound Laundry* ™

To apply a processing function during playback of your wave file, select a PlugIn from the first drop-down list in the *PlugIn Center* section and restart the playback of selected wave file. The corresponding PlugIn window pops up, and the influence of the processing module on your music as well as the changes of its parameter settings are immediately audible. To listen to the changes that a processing module applies to your music, enable the *difference* checkbox in the **PlugIn Stations** output section and you can hear the clicks taken out by the **DeScratcher** PlugIn or the noise removed by the **DeNoiser** PlugIn.

1.5 Registration

The customers who purchased the **Sound Laundry** ™ or separate additional PlugIns via Internet are automatically registered.

Those customers which bought the Sound Laundry ™ on CD via our distributors are invited to register online via **Algorithmix**® web page. We can keep you up to date with newsletter, new PlugIns, and software updates.

Please note that we can offer you technical support and special conditions for updates only if you are registered.

1.6 Technical Support

We hope you will enjoy rescuing your sound treasures with the **Sound Laundry**™. We have been continuously improving this product and developing new PlugIns. We invite you to visit our web page often

http://www.algorithmix.com

to check out our new products and to share your experiences with us.

Before you send us an email concerning any troubles, please try to read related chapters in the operating manual.

You can order our new products online with a credit card 24 hours a day, 7 days a week, 52 weeks a year. For your safety, the credit card data is transferred only via a secure server. After ordering online you will receive your software within minutes to the e-mail address you entered.

If you'd like to order new PlugIns, share your experiences, or send us suggestions concerning new developments, choose one of the e-mail addresses:

- Information	info@algorithmix.com
- Support	support@algorithmix.com
- Sales	sales@algorithmix.com
- Job Opportunities	job@algorithmix.com

Algorithmix®, July 2001

2. PlugIn Station ™

2.1 Overview

The **PlugIn Station**™ is the host for **Sound Laundry**™ PlugIns. It is responsible for handling the input and output, audio files, as well as for loading and chaining audio processing PlugIns.

The **PlugIn Station**™ includes a sophisticated .WAV and MP3 player, the **Algo*Player***™, with advanced functions like differing playback speed, reverse playback, scrubbing, loop mode, and real-time playback of files compressed with **Algo*Press***™. For achieving the highest audio quality the **Sound Laundry**™ accepts also standard 32-bit IEEE float input audio files and can also create output files in this format. For the enhanced audio quality, when using 16-bit output audio files and if monitoring audio over 16-bit sound card, *dither* has been added to the output. It can be optionally switched off in the in the *I/O Options* window.

*Screenshot of the **PlugIn Station***

The *PlugIn Center I & II* allows loading up to 10 PlugIns in the audio processing chain of the Sound Laundry™. They can be individually bypassed with check-box “use”, closed and opened with the check-box “open” without taking them out from the processing chain, and removed with the “X” check-box.

The *CPU Load* display allows permanent monitoring of the CPU usage for all currently activated PlugIns and help to prevent system overloads. The barograph *Input/Output Level Meter* helps you to check the input level and avoid potential overloads after processing. The numeric fields on the right hold the peak playback level in every input and output channel. They can be reset by clicking inside the field.

Due to **Sound Laundry**™ extra internal headroom, the level meter allows displaying levels above 0 dB. With standard 16-bit wave files, however, the final output level must never exceed 0 dB to avoid annoying distortions. Use the **SignalControl** PlugIn to reduce output levels if they peak above 0 dB.

2. PlugIn Station ™

One of the main features of the **Sound Laundry**™ is the handling of *temporary pres*ets. You can store up to six *temporary presets* containing all parameters and positions of loaded PlugIns. The *noise profile* is also stored giving you opportunity to easily compare different setups in your project. The active preset slots are marked *red*, loaded, but unused preset slots *green*, and empty preset slots *white*. You may store or reload the active presets to hard disk for later use. If your preset does not change the PlugIn configuration, but only the parameters, you can usually change the temporary presets in real time without audible distortion.

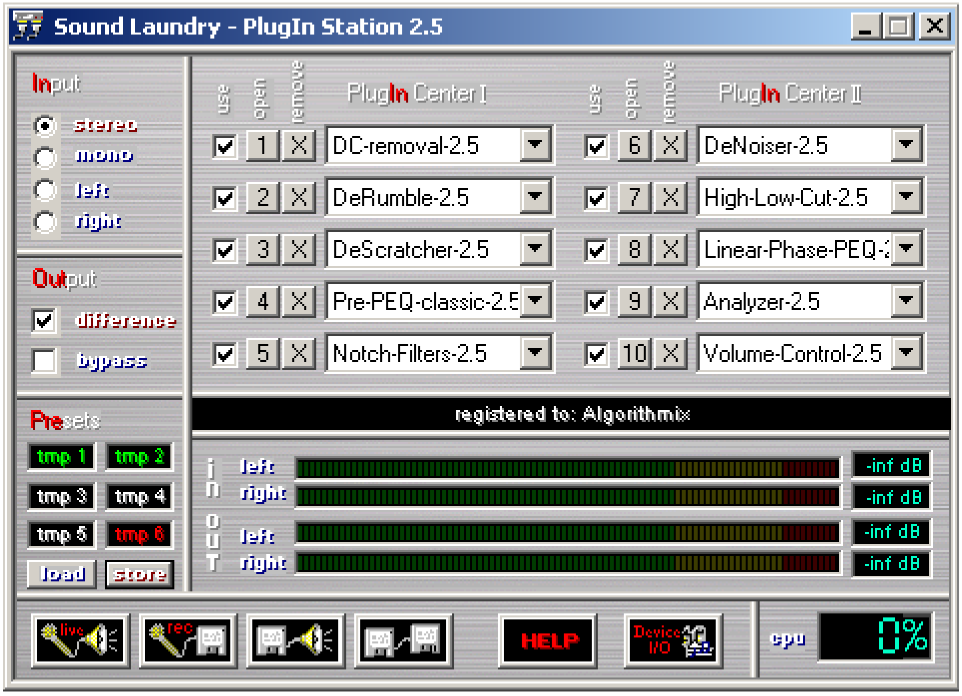
The four mode buttons at the bottom of the **PlugIn Station**™ invoke four different real-time processing modes:

1. **Live Processing** – input of the sound card live processed and sent to its output
2. **Recording** – input of the sound card live processed and sent to wave file
3. **Playback** – wave file processed and sent to the output of the sound card
4. **Off-line Processing** – input wave file processed and sent to an output file (faster than real-time)

The **PlugIn Station**™ can also playback MP3 files, just for compatibility reasons. However, we do not recommend using restoration processing on MP3 coded files. They are so strongly pre-processed that sometimes quite strange results are to expect.

2. PlugIn Station ™

2.3 PlugIn Station™ Quick Reference



- **Input** – selects input signal: *stereo, mono, left* or *right*
- **Output** – activate *difference* function and *bypass*
- **Presets** – use the left / right mouse button to add / recall up to 6 temporary presets, click [store] / [load] to store or recall global presets
- **PlugIn Center I and II** – allows to load a processing chain of up to 10 PlugIns
- **Status Bar** – displays online help and registration information
- **Level Meters** – left / right input and output level monitoring, click numeric field to reset the desired peak hold display
- **Control Section** – live / record / playback / offline mode
- **Help** – opens the appropriate help file
- **Device I/O** – opens the *I/O Options* window to set up the input / output and recording devices
- **cpu** – monitors the CPU load

2. PlugIn Station ™

2.7 Sound Laundry™ Hotkey Table

To speed up tasks if intensively working with **Sound Laundry**™ a hotkey list has been implemented. The definitions are works for PlugIn Station, AlgoPlayer™ and for the PlugIns (if applicable).

d	→	difference	b	→	bypass
1 to 0	→	select PlugIn slot 1 to 10	`	→	bypass selected PlugIn
-	→	toggle selected PlugIn	=	→	remove selected PlugIn
F1 - F6	→	activate Preset 1 to 6	shift F1 –F6	→	store current configuration in Presets 1 to 6
x	→	toggle all PlugIn windows			
space	→	start / stop playback	backspace	→	rewind to begin of block / file
o	→	load new file	l	→	toggle loop mode
e	→	export	p	→	process
u	→	undo	c	→	cut block
s	→	split file at cursor	i	→	import file at cursor
\	→	clear block	q	→	quit play dialog
insert/+	→	zoom +	del / num -	→	zoom –
begin	→	zoom max in	end	→	zoom max out
pg up	→	switch wave mode forward	pg down	→	switch wave mode back
ctrl-c	→	copy setting in PEQ / Notch	ctrl-v	→	paste setting in PEQ / Notch
ctrl-t	→	toggle last / current settings			

2. PlugIn Station

2.5 AlgoPlayer™ Quick Reference

The **Algo*Player***™ opens after pressing the *Playback* mode button in the **PlugIn Station**™ window.

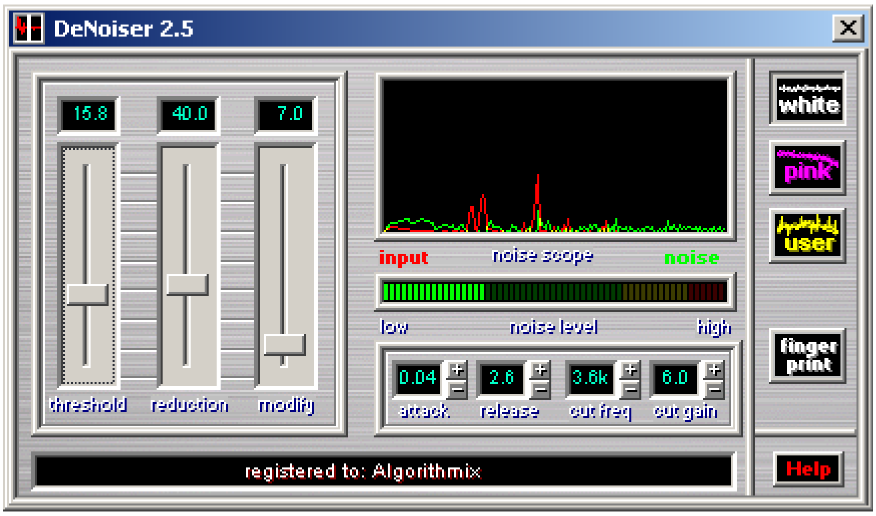
*Screenshot of the **AlgoPlayer**™*

- **waveform display** – use the left mouse button to position cursor in playback or scrub mode, use the right mouse button to select an area (double click right to maximize the selected area)
- **scrollbar** – click on the left/right edge or in the middle to zoom or to move the waveform, use double click to maximize
- **locator** – shows actual playback position, click to enter required numeric value
- **cue slider** – sets playback speed in cue mode, click left *and* right to lock its position
- **zoom in/out** – changes the zoom mode for the waveform
- **display mode** – selects the moving waveform, the moving cursor or both
- **block** – displays selected block, click numeric field to enter required value, click clear to reset
- **status bar** – displays file information and online help
- **playback control** – load file, start / stop playback, rewind file
- **processing control** – export selected area, process selected area, undo processing (one level)
- **edit control** – cut selected area, split / import file at the current cursor position
- **loop** – loops selected area or the whole file if selected area is clear
- **playback mode** – playback / cue / scrub, fast / very fast (set cue / scrub speed with cue slider)

5. DeNoiser PlugIn

5.1 Overview

One of the quite common problems in the daily business of the audio engineer are noisy signals. Therefore there is a need for solutions how to extract desired signal from the accompanying noise components. The standard method of extracting signal from noise is to design an appropriate filter which removes the noise components and at the same time lets the desired signal go through unchanged. This sounds plausible, but in typical cases in practice does not work. If the signal and noise spectra overlap there is no traditional filter technology which can perfectly extract the desired signal. In such cases we need to compromise; the noise reduction can be only done at the expense of distorting the signal we want to preserve.



Screenshot of the ***DeNoiser*** *PlugIn*

One of the classical methods to prevent noise before it arises are complementary system like the famous tape noise reduction system from DOLBY™ Lab. The recorded signal is processed in a special way before recording to the tape. Then after playback an inverted operation is applied to the coded signal. The idea behind is to reduce the noise, but keep the signal unchanged.

But what is to do if we get noisy recordings which have not been processed that way. Does any chance exist to reduce or remove the noise without substantially changing the original signal?

The **Algorithmix** ® **DeNoiser** PlugIn belongs to the so-called *single-ended noise reduction* systems. This means it does not need any special coding procedure before recording, like the DOLBY™ Lab system mentioned above. It is a tool that efficiently removes any kind of broad-band noise from pre-recorded audio tracks. The **Algorithmix**® **DeNoiser** PlugIn is a weapon against tape hiss, noise from telephone-call cuts, background noise from live recordings, and residual noise from old records after processing with the **DeScratcher** PlugIn. The **DeNoiser** PlugIn is very useful in forensic applications, too.

If you have to remove clicks and impuls-like disturbances you absolutely need to treat your signal first with the **DeScratcher** PlugIn and then subsequently use the **DeNoiser** PlugIn to clean your signal from the remaining broad-band noise. Applying the **DeNoiser** directly to heavy clicks will create significant, singing-birds type artifacts.

5. DeNoiser PlugIn

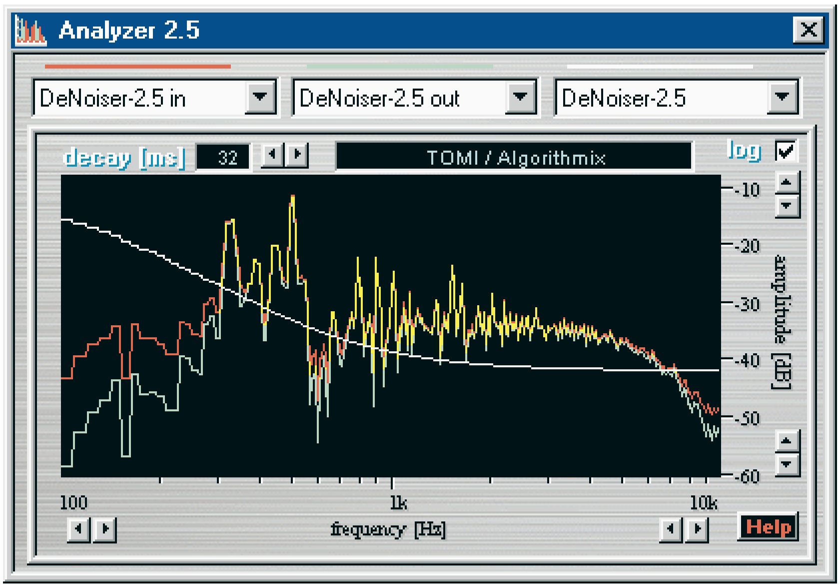
5.2 DeNoising Process

In a single-ended noise reduction system, the user decides which noise characteristic have to be applied for the *de-noising* process. In the **DeNoiser** PlugIn, there are two predefined noise profiles: *white* and *pink*. In addition, to receive the highest performance from the system, the user has an opportunity to record his own *noise profile*. When recording an application specific *noise profile*, we highly recommend recording it from a portion containing the background noise only**.**

Since the entire process runs in real time, you can simply switch between the three noise profiles while listening to the output signal. This helps you discover which *noise profile* is best suited to the audio material being processed. We discovered that a shelving-like modification of the upper end of the *noise profile* helps to improve the results a lot. This can be done by adjusting the parameters *cut freq* and *cut gain*.

The *noise reduction* process in the **DeNoise**r PlugIn is controlled basically by just two parameters: *threshold* and *reduction* allowing an easy search for optimal results depending on the given input signal. The remaining parameters are for the fine tuning.

The frequency spectrum of the input signal as well as the spectrum of the noise taken out during the *de-noising* process can be followed in the *noise scope.* You can make an estimation of noise amount being removed based on the *noise level* indicator (see Fig.1). Although for accurate examination of the frequency spectra, we recommend using the **Analyzer** PlugIn. The **Analyzer** PlugIn can display the input and output signals of the **De-Noiser** PlugIn as well as the *noise profile* applied to processed audio material.



***Analyzer** window with the correct setup for the **DeNoiser** PlugIn.*

5. DeNoiser PlugIn

By setting the **Analyzer** PlugIn to *De-noiser in* (red), *De-noiser out* (green), and the white line *to De-Noiser noise profile,* you can intuitively follow the effect of the de-noising process on the processed material. The *noise profile* (white) marks the threshold border, above which no noise reduction is applied. The *threshold* parameter moves this *noise profile* up and down and can be used to set up the profile just above the background noise level. For a given *threshold,* the second parameter, called *reduction,* controls the amount of the spectral components removed below the chosen noise profile.

A good starting value for *threshold* is to set the *noise profile* just above the background noise level (approx. 10 dB). A subsequent increase of the *reduction* parameter should significantly reduce the background noise. If noticeable artifacts in the form of so-called singing or robot-like sounds appear (time aliasing phenomenon), decreasing the *reduction* parameter and increasing the *threshold* level (up to about 30 dB above the background noise) usually helps. Further reducing of artifacts can be achieved by careful setting of the third slider *modify*.

The overall performance of the **De-Noiser** PlugIn is heavily dependent on the proper adjustment of the parameter in the *Quality Window* which are related to the frequency-domain processing. Especially for the final mix, the rules recommended for the *Quality Window* settings should be carefully followed.

The *noise scope* window displays the frequency spectrum of the input signal as well as the noise taken out during the *de-noising* process. This allows objective control of the *De-Noiser* algorithm activities.

However, especially when using a user defined *Noise Profiles*, the **Analyzer** PlugIn is strongly recommended. It allows visual control of the *de-noising* process (placement of the *noise profile*) and precise measurement of the frequency spectrum from 10 Hz to 20 kHz with a dynamic range of more than 130 dB. The display of the **Analyzer** PlugIn can simultaneously show three characteristics: *De-Noiser* input, *De-Noiser* output, and the *noise profile.*

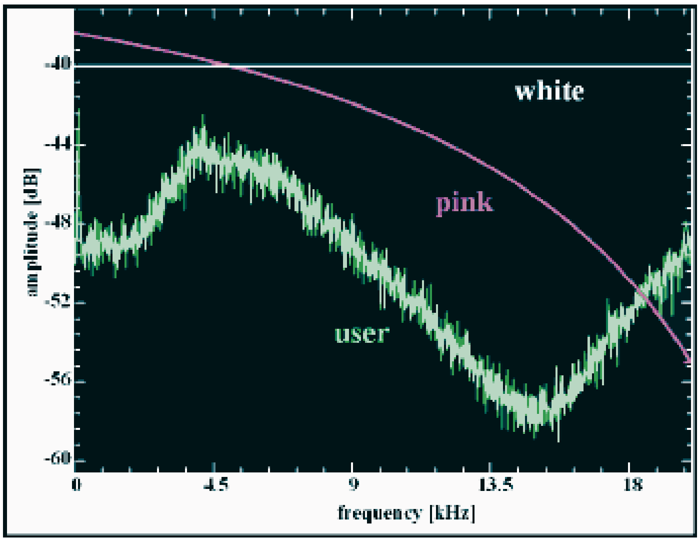
If the **Analyzer** PlugIn is not available, the *noise level* meter is quite useful for rough estimation of the noise amount removed from the input signal. If the background noise varies greatly, there is no general rule for the practical use of the *noise level* meter.

The more noise to be removed, the more the operator has to be aware of possible artifacts arising from the noise removal system. These artifacts are greatly influenced by the *Quality Window* settings and the *noise profile* used. Although the *noise level* meter can help in objective comparison among different setups, the final judgment should always be made by carefully listening to the output signal with a high-quality speaker system or headphones.

The *noise profile* should ideally represent the frequency distribution of the noise to be removed from the noisy input signal. It is a kind of reference *spectral horizon* used by the *de-noising* algorithm. The position of the *noise profile* relative to the input signal can be controlled with the *threshold* parameter. The **Analyzer** PlugIn provides an intuitive visual control of both the input signal spectrum and the *noise profile*.

Applying user-defined *noise profiles* taken properly from the audio material to be processed, the quality of the entire *de-noising* process can be considerably enhanced. It is important that the *noise profile* should be taken from a part of audio material containing nothing but the noise signal to be removed. This can be done from the *Noise Profile Manager* dialog, which opens after clicking on the *Noise Print*. The **Algo*Player*** ™ allows a precise marking of the waveform part best suitable for taking the *noise print.* It can be also stored for the later use.

5. DeNoiser PlugIn



Frequency spectrum of different noise profiles.

Playback of the area marked in **Algo*Player*** starts after pressing *record* in the *Noise Profile Manager* window. If the “noise-only” part of a recording is very short (below 2 seconds), it is recommended you set up a loop for repetitive playback before recording a *noise profile* (see **PlugIn Station** help for details). The loop can be played back several times to obtain an averaged *noise profile*. As already pointed out, the loop should contain nothing but noise (or whatever has to be removed from the input signal). If a noise profile includes spectral components of the signal to be recovered, they will also be removed or at least lowered in the *de-noising* process. Therefore, much care and sensitivity is recommended when preparing a user-specific *noise profile*.

Noise Profile Manager

In many cases excellent results can be achieved using just the standard noise prints: white and pink properly modified by the *cut freq* and *cut gain* parameters. We recommended storing the good working noise prints for the future use.

Dependent of the intensity of the de-noising process some artifact in form of “singing birds” or whistle-like tones can occur. In addition to mentioned before slider *modify*, they can be effectively minimized by proper setting of *attack* and *release* parameters. Normally attack time should be set up to the values below 0.1 seconds (e.g., 0.03 sec.), and the release time above 1 second (e.g., 2 sec.). Longer attack times than recommended can blur the signal details. Shorter release times than recommended cause cutting of the ambience, but at the cost of keeping more noise. In this case an improvement can be done by proper setting the *cut freq* (usually 2 – 4 kHz) and *cut gain* (positive values).

5. DeNoiser PlugIn

The fine adjustment of the parameters: *modify, attack, release, freq cut*, and *gain cut* is a iterative process requiring some experience and good feeling for the trade-off between original signal, remaining noise, and artifacts.

For forensic applications artifacts are usually not critical. The main goal there is to get as much valuable information as possible. For the speech recordings the most important parameter is the ineligibility. So, noise reduction used there is much more intensive comparing to the recording restoration applications. Therefore *attack* can generally be low (0 – 20 ms) to allow all signals above the noise print level to pass the *denoising* part. *Release* can normally be kept quite large (seconds), but lower values gives more abrupt noise reduction. This can be advantageous at extremely noisy material. Use spectral modification with *cut freq* setting *cut gain* below 0 (negative values) to preserve harmonic contents of the processed audio material. An additional combination of two PEQs, one pre-FD and one post-FD, might help to extract important information. The pre-FD PEQ can emphasize particular spectral details before de-noising process and the post-FD PEQ flatten the spectrum to the original one. This effect can easily be followed with *difference* function enabled.

5.3 DeNoiser Quick Reference

The **DeNoiser** PlugIn provides following controls

- **threshold** – defines the reference level for the de-noising algorithm (moves the noise profile)
- **reduction** – determines the amount of noise to be removed (0 already cause noise reduction)
- **modify** – reduces artifacts if working with critical audio material
- **attack** – set up the response time of the de-noising algorithm (0 – 1 sec)
- **release** – set up the release time of the de-noising algorithm (0 – 10 sec.)
- **cut freq** – cut-off frequency of the *noise profile* shelving modifier
- **cut gain** – boost/cut gain of the *noise profile* shelving modifier
- **white/pink/user** – selects noise profile used as a reference for de-noising process
- **noise print** – opens *Noise Profile Manager* for recording, loading, and saving noise profiles
- **help** – opens the help file

and following displays

- **noise scope** – displays frequency spectrum of the input signal and the noise taken out
- **noise level** – shows the amount of noise being removed

5. DeNoiser PlugIn

5.4 Applications Tips

The **De-Noiser** PlugIn is an easy-to-use audio cleaning tool based on efficient signal processing algorithms. In most cases, you will automatically achieve good results. To maximize success, especially when working with heavily noise-polluted audio material, there are some practical rules:

- For the best results, record your own noise profile for every piece you de-noise. Choose a portion of the recording that does not contain any material you want to recover, only noise you intend to remove. If the audio piece to be processed contains more only-noise parts, try to record a few noise profiles and test them to determine the one that works best.
- If the part containing only the noise signal is very short (under 2 seconds), we recommend setting up a loop for repetitive playback before recording a *noise profile* (see **PlugIn Station** help).
- If you cannot find in your entire recording any noise-only piece, use pre-defined profiles and modify them with *cut freq* and *cut gain* parameters, or try to record a user noise profile from a low-level part of the signal (hopefully with pauses in the useful signal). Finally, look for the careful adjustment of the parameters *threshold* and *reduction*
- Very strongly changing noise level and noise characteristic may be problematic. In such cases professional mastering engineers try to cut the original piece in parts and treat them individually with different de-noising parameters. Later the de-noised pieces are joined together.
- Do not exaggerate the parameters *threshold* and *reduction* to avoid artifacts occurring in the form of singing or robot-like sounds (*time aliasing*). Begin with a moderate adjustment by setting the *noise profile* to just above the background noise level (approx. 10 dB) and gradually increasing the *reduction* parameter. Finally try to recursively find the best relation between these two parameters. Do not forget to correctly set up the time and frequency resolution as mentioned in the **DSP Quality** window.
- If you are working on heavily disturbed material, find a good acoustical compromise between the level of the remaining noise and artifacts introduced to the output signal. Be indulgent if you have hopelessly noisy material. Nobody can restore the original signal without having enough original information.
- It helps sometimes to apply the *de-noising* process two or more times consecutively with a moderate parameter setting rather than one pass with an extreme setup.
- A recommended tool that significantly helps to adjust the *De-Noiser* parameter is the **Analyzer** PlugIn. It allows for following the de-noising process, especially the influence of the *noise profile* and parameter settings.
- For the best results use your own ears in connection with the *difference* feature of the **PlugIn Station**. Switch between the original input signal and the input/output difference, i.e., the portion of signal removed by the *de-noising* algorithm. This differential signal normally should not contain any parts of the original signal you want to preserve.

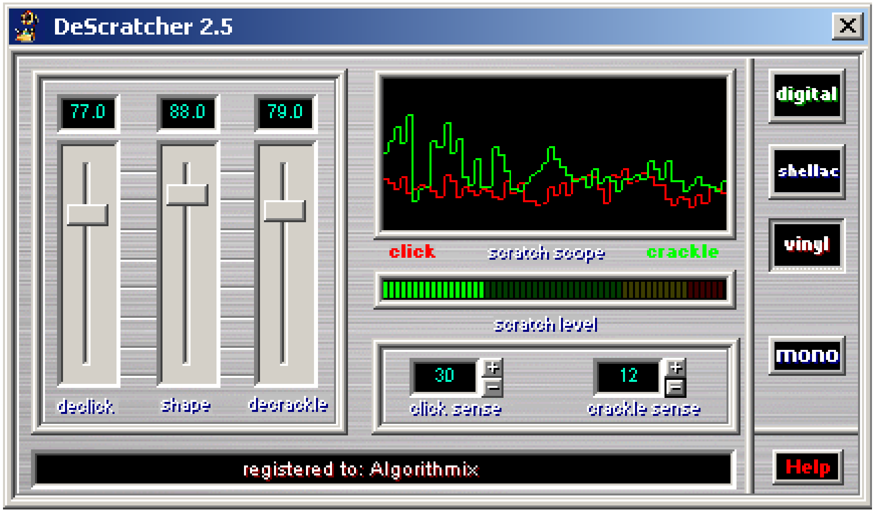
Important Remark:

The **De-Noiser** PlugIn is a very fast and very effective tool. You will be amazed at how dramatic the audio quality of noisy recordings can be improved. But please, do not expect miracles if you process material containing so much noise that the original signal is no longer distinguishable. The information theory says that once the information is sunk in noise (bad signal-to-noise ratio) and there is not enough information about the properties of the original signal, or even worse, the original signal is non-linearly distorted, the de-noising process can deliver only limited-quality results.

6. DeScratcher PlugIn

6.1 Overview

The **De-Scratcher** PlugIn effectively removes *clicks* and *crackles* from old vinyl and shellac records, or audio recordings tainted by switching noise, digital cross-talk, or thyristor buzz. Unlike other systems, **Algorithmix**® **De-Scratcher** PlugIn works virtually without artifacts, providing the correct setting for all parameters.



Screenshot of the ***DeScratcher*** *PlugIn*

Since the CPU requirement of the **De-Scratcher** PlugIn is very low (about 20% on an old 133 MHz Pentium for a 16 bit, stereo, 44.1kHz .WAV file), you can optimize all the parameters while listening to the processing in real-time. With today`s PCs all the remaining Sound Laundry PlugIns can be chained to the **De-Scratcher** for complete *real-time filtering* of your audio material. You can even use this combination for cleaning live old records, by connecting your turntable (assuming, it is equipped with an appropriate pre-amplifier) to the input of your sound card and its output to your stereo system.

The *de-scratching* algorithm consists of two main parts: the *de-clicking* filter and the *de-crackling* filter. While the *de-clicking* filter is used to remove heavy clicks from old shellac and vinyl records or switching noise coming from audio equipment, the *de-crackling* filter removes any remaining small clicks and crackles.

The *scratch scope* and the *scratch level meter* help you to find the optimal setting for the *declick* and *decrackle* parameters. Advance parameter like *shape*, *click sense*, and *crackle sense* are useful to minimize the occurrence of artifacts if working with difficult audio material. We recommend you to perform final precise real-time parameter optimization with a critical listening to the signal removed with the *de-scratching* algorithm by using the *difference* feature of the **PlugIn Station.**

The **DeScratcher** provides well defined *application profiles* which preset the internal parameters and external advanced parameters helping in typical restoration situations: *digital* spikes, *shellac* and *vinyl*.

In addition to its main task, removal of clicks and crackles, the **DeScratcher** PlugIn successfully smoothes any kind of distortion causes by signal overload (clipping).

6. DeScratcher PlugIn

6.2 DeClicking Process

The *de-clicking filter* of the **DeScratcher** PlugIn removes clicks from old records as well as any impulse-like noise arising from analog or digital audio equipment. The higher the *declick* parameter is set, the more clicks are removed. At a setting of zero, virtually all clicks pass through the filter. For the *de-clicking* of typically polluted vinyl records transferred to the digital domain, a *declick* value of approximately 30 to 50 works well in the most cases. Higher values can cause artifacts and should be adjusted to get a good compromise between clicks removal performance and sound quality of the remaining original signal.

The *Shape* parameter is closely associated with the width of the clicks and their character. Normal starting value is around 70. You should try to keep it as small as possible to prevent touching the original audio transients as much as possible. If working on very heavy damaged audio material it would be necessary to increase the value looking for a good compromise between click removal and the quality of the remaining original signal. In general, smaller values of *shape* parameter reduce the risk of artifacts, but some clicks may pass through. *Click Sense* parameter determines the minimal width of transients which should be interpreted as clicks to be removed. Try to lower this value only if you hear artifacts at default value of 30. Normally lower values lead the algorithm to remove very strong clicks only.

The number of clicks removed for a certain set-up of the *declick* parameter is displayed in the *scratch scope* by the red line. This provides for monitoring the click's detection level. The final setting of the *declick* parameter should be adjusted by carefully listening to different parts of the material intended to be processed.

Typical click in the signal taken from a vinyl record (left) and restored signal (right).

The internal parameters of the de-clicking algorithm, as well as *shape* and *click sense* parameters are well predefined for the typical audio restoration situations. Use *digital* button if you need to remove narrow and very sharp clicks like these coming from digital switching or cross-talk. For restoring shellac records press just *shellac*. Vinyl records which are normally characterized by wider clicks are best treated after pressing *vinyl* button.

For additional help, we highly recommend using the *difference* feature of the **PlugIn Station**, allowing for intuitive and optimal parameter setting. You can switch between the original input signal and the *input/output difference*, i.e., the part of signal taken out by the *de-scratching* algorithm. Normally, this differential signal should not contain any audible parts of the original signal you want to preserve.

6. DeScratcher PlugIn

6.3 DeCrackling Process

The *de-crackling* filter of the **De-Scratcher PlugIn** removes crackles and small clicks left after the *de-clicking* process, or other crackle-like disturbances included in audio signals. The higher the *decrackle* parameter is set, the more crackles are removed from the original signal. Normally values between 50 and 70 works well in most situations. The extreme setting, close to 100, results in a smoothing of the input signal. In general, this parameter can safely be set to 80 without any audible artifacts. In some cases, even an extreme setting up to 100 can still improve the quality of processed audio material.

The amount of crackles removed from the input signal and parameter setting influence is monitored by the green line in the *Scratch Scope*. For audio material containing a high crackle background noise, the results may be better with the green line in the *scratch scope*, hitting the top of the display.

The parameter *crackle sense* determines the maximal width of fast low-level transients which are classified as possible crackles. The default value is 30 and should be decreased only if you can lower artifacts. It can be the case if you work with heavily demaged audio material.

As in the *de-clicking* process, the final setting of the *decrackle* and *crackle sense* parameters should be performed by critically listening to different parts of the audio file to be de-crackled. We highly recommend using the *difference* feature of the PlugIn-Station, allowing optimal parameter set-up in a very intuitive way. You can switch over between the original input signal and the *input/output difference*, i.e., the part of signal removed by the *de-scratching* algorithm. This difference signal normally should not contain parts of the original signal you want to preserve.

Enabling the *mono* switch forces the **De-Scratcher** PlugIn algorithms to merge both channels of a stereo file back to mono, but only <u>after</u> clicks and crackles have been removed. For genuine mono input files, or if selecting only the left or right channel in the *Input* section of the **PlugIn Station**, this switch has no function.

To achieve the highest possible performance when restoring and mastering old mono vinyl or shellac records to CD, we recommend using a stereo pick-up to record the signal in stereo mode.

Applying the stereo recording mode to mono records and merging both channels to mono first after *de-clicking* and *de-crackling* each channel individually improves signal-to-noise ratio by at least 3dB (which is actually a factor of two) compared to restoration procedures applied to pure mono files.

6.4 Scratch Scope and Scratch Meter

The *scratch scope* allows monitoring and reproducable set-up for the *declick* and *decrackle* parameters controlling the activity of the **De-Scratcher PlugIn** algorithms. The vertical axis of the display is scaled in *clicks* (red) or *crackles* (green) removed from the input signal per time unit. The *scratch scope* shows the last 12 seconds of the *de-scratcher* activities. The current *click* and *crackle* level is displayed on the right side and conveyed to the left side of the *scratch scope* display.

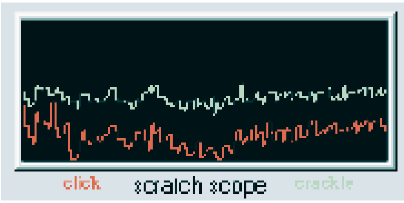
In general, you will find that the *click level* (red) rises from the bottom only if significant clicks are present in the input signal. For records in good condition with the correct setting of the *declick* parameter, the red line should normally stay in the bottom half of the *scratch scope*.

The green line, indicating the crackle level, is usually more sensitive to the setting of the *decrackle* parameter. Even for audio material without any crackles, there will be a significant *de-crackle level* displayed in the *scratch scope* if the setting of the *decrackle* parameter is above 80. In this case, the output signal is a smoothed version of the input signal and there may be some loss in high frequencies.

6. DeScratcher PlugIn

For very bad audio material with a high *crackle level*, it may be necessary to use a parameter set-up corresponding to the green crackle level reaching the top of the *scratch scope* display.

If you have problems correctly interpreting the levels displayed in the *scratch scope*, always ask your ears. For further advice about correct using of the **DeScratcher** PlugIn look at *Application Tips*.



Scratch Scope

The *scratch level meter* offers an at-a-glance option for evaluating the overall click and crackle level. The more clicks and crackles detected, the higher the scratch level displayed.

This meter can help examine the influence of the *de-clicking* and the *de-crackling* function on the audio material being processed. For higher settings of the *declick* and *decrackle* parameter, or wrong setup of *shape*, *click sense*, and *crackle sense*, the *scratch level meter* may display removal activities, even when working on audio material without any noticeable clicks and/or crackles. This is due to the exaggerated, material non-adequate algorithm sensitivity set up. These high sensitivity levels blur the distinction between scratches and attack regions of the original audio signal. Nevertheless, high settings are sometimes useful for heavily damaged audio material. However, it is up to you to find the proper compromise between the level of disturbances and the relative audio quality of the signal after processing.

6.5 DeScratcher Quick Reference

The interface of the **DeScratcher** PlugIn provides following controls:

- **declick** – set up the amount of clicks being removed
- **shape** – adjust the de-clicker to reject narrower (digital) or wider (vinyl) clicks
- **decracle** – set up the amount of crackles being removed
- **click sense** – determines the min. width for clicks to be recognized
- **crackle sense** – determines the max. width of crackles to be recognized
- **digital** – profile recommended for very narrow clicks and digital spikes
- **shellac** – profile recommended for restoration of 78rpm shellac records
- **vinyl** – universal profile for vinyl records and the most of restoration tasks
- **mono** – switch recommended for mono records played back with stereo pick-ups
- **help** – opens the appropriate help file

and following displays:

- **scratch scope** – shows the amount of click and crackle removal activities during the last 12 seconds
- **scratch level** – momentary amount of clicks and crackles

6. DeScratcher PlugIn

6.6 Applications Tips

The **De-Scratcher PlugIn** is an easy-to-use audio restoration tool based on extremely efficient signal processing algorithms. In most cases you will automatically achieve good results. To get the maximum, especially when working with heavily damaged audio material, there are some practical rules:

- Use a stereo record player if you restore old mono vinyl or shellac records and process both channels individually before merging again to mono. This improves the *signal-to-noise ratio* by at least 3dB (actually a factor of two) when compared to the one-channel *de-scratching* procedure.
- Transfer recordings directly to .WAV file without using any processing devices like limiter or compressor prior to the *de-scratching* procedure.
- Use the **DC-Removal** PlugIn to remove the *DC-offset* before going into the **De-Scratcher** PlugIn. This may improve the audio quality for the low-level signals in the whole processing chain.
- If the audio material to be restored contains very strong clicks, you may allow some *clipping* while transferring to the digital domain. We recommend you make a few versions recorded with different input gain and compare the results after the *de-scratching* process.
- To get good results in short time work systematically: first set up the *declicking* part (setting up the *decrackle* parameter at 0) and then subsequent *decrackling* part of the **DeScratcher** PlugIn. Start with proper profile (*digital, shellac, vinyl*), *decklick* equel to 60, and *shape* equal to 70. Dependent on how heavy are clicks you need to remove look for a proper *declick* setup (good compromise between original material to be prevented and possible artifacts). In the second step investigate if the *shape* parameter adjustment gets fewer artifacts. You may need to recursively readjust *decklick* if *shape* needs to be significantly changed. In third step you can try to lower *click sense* if it decreases the intensity of artifacts. In fourth step you should activate the *decrackle* parameter starting with 70 and find the best setting. If working with critical material you can try to decrease *crackle sense*, assuming you hear any advantage in sound.
- Do not exaggerate with the *declick* and *decrackle* parameter to avoid artifacts. When working on heavily disturbed material, use a good acoustical compromise between the level of remaining disturbances and artifacts introduced to the output signal. Be indulgent if you have hopelessly damaged material; nobody can restore original data from nothing.
- To complete the restoration process for old records, first use the **De-Noiser** PlugIn to remove broadband residual noise and eventually PlugIns to modify the frequency range: **Low/HighCut**, or **PEQclassic**.
- For the best results, use your own ears in combination with the *difference* feature of the **PlugIn Station**. Switch between the original input signal and the *input/output difference*, i.e., the part of the signal taken out by the *de-scratching* algorithms. This difference signal normally should not contain any parts of the original signal you want to preserve.

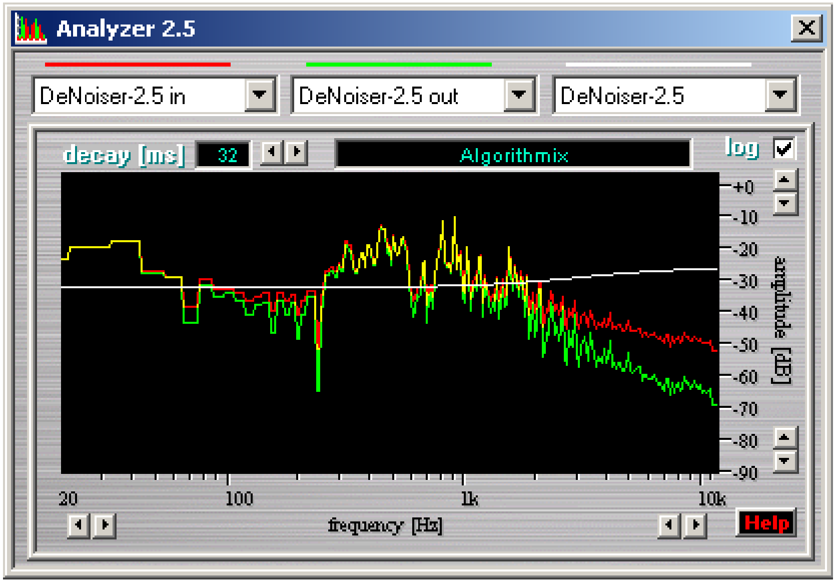
Important Remark:

The **De-Scratcher PlugIn** is a very fast and very effective tool. You will be amazed by how dramatically the audio quality of old records can be improved. But please do not expect miracles if you process material with long gaps or jumps. The information theory says that once the information is lost and there is not enough redundancy in the remaining material, the restoration process for the original material is impossible. In such hopeless cases, experienced mastering engineers try to transfer similar recording parts into gaps using very precise audio editors.

7. Analyzer PlugIn

7.1 Overview

The **Analyzer** PlugIn is intended as an additional graphical tool for supporting the **Algorithmix**® frequency-domain PlugIns (**DeNoiser, Low/HighCut, PEQ linearPhase**). It can display the frequency spectrum for two independent channels (red and green) and a frequency characteristic (filters or 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.



*Screenshot of the **Analyzer** PlugIn*

Each input channel of the **Analyzer** PlugIn can be connected to any input or output belonging to any frequency-domain PlugIn (**DeNoiser, Low/HighCut, PEQ linearPhase**). Because the PlugIns are chained, note that some outputs are identical to some inputs and can be chosen alternatively (e.g., **De-Noiser** output can carry the same signal like **Low/HighCut** input).

In addition to the two independently selectable signals in the display of the **Analyzer** PlugIn, the filter characteristic of the **High/LowCut** PlugIn, or the *noise profile* used currently in the **DeNoiser** PlugIn, can be displayed (in white). This is very helpful for successful parameter setup 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.

7. Analyzer PlugIn

7.2 Analyzer Reference

To visualize 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 **High/LowCut** PlugIn, or to the *noiseprofile* currently used in the **DeNoiser** PlugIn. These curves can be displayed simultaneously with any two spectrum lines pre-chosen from the drop down lists.

When loading the **DeNoiser** PlugIn prior to the **Analyzer** PlugIn, the **Analyzer** will be automatically set up to display the input (red) and output (green) spectrum of the **DeNoiser** 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. 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.

7.3 Applications Tips

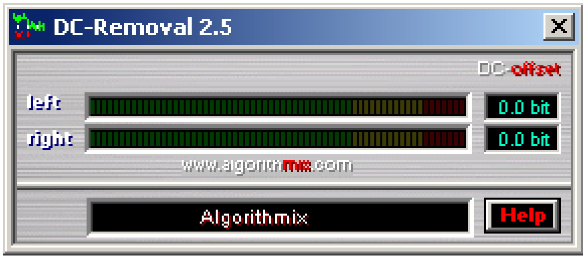
If you realize that the display of the Analyzer flickers by Windows2000™ switch off the function "pointer shadow" in **My Computer → Control Panel → Mouse → Pointers**.

In the most cases audio spectrum gives us the most adequate information if observed on the logarithmic scale (check-box *log* on). It is bacuase of the method how our ears respond to the different frequencies. If looking for harmonics related to a given frequency, e.g. 50/60 Hz hum, it is convenient to use linear scale (check-box *log* off). It is a very useful feature fir the **Notch Filters** PlugIn in order to find optimal adjustment for the frequencies, amplitudes, and bandwidths of the notch filters.

8. DC-Removal PlugIn

8.1 Overview

Many of today's multimedia boards are equipped with low-cost *analog-to-digital converters* that tend to produce a so-called *DC-offset*. This means that the analog zero signal is not converted to digital zero, but to a certain positive or negative value, the *DC-offset*.



*Screenshot of the **DC-Removal** PlugIn*

The DC-offset narrows the maximum dynamic range, and causes problems during editing and dynamic processing, especially for low-level signals. Short cross-fading of signals with different DC-offset values can produce an audible thumb. Dynamic processors may react improperly to the DC-offset tainted signals.

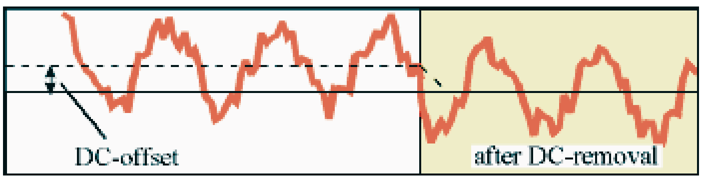
8.2 DC-Offset

The left part of Fig. 2 demonstrates signal with a positive *DC-offset*. All other non-zero samples are also shifted by the amount of the *DC-offset*. To avoid signal clipping, the input volume has to be decreased compared to an ideal converter, resulting in a narrowed dynamic range of the recorded signal. It is not uncommon for this to measure five bits of *DC-offset*. Frequently the *DC-offset* of the analog-to-digital converter is not constant and drifts with time and/or temperature.

A low-level audio signal including *DC-offset* or even a high-level signal attenuated digitally can produce very annoying sounds on the playback side due to asymmetry.

The **DC-Removal** PlugIn is a remedy for those recordings tainted with *DC-offset*. It removes *DC-offset* from sample strings, bringing your signal perfectly to the middle of the system's dynamic range as shown in Fig. 2 on the right.

The *DC-Removal Filter* is designed as a digital high-resolution (80 bits) *high-pass filter* with a very low cut-off frequency (a few Hz). It removes DC components from poor recordings even if the value of the *DC-offset* permanently drifts.



A signal tainted with DC-offset (left) and after DC-filtering (right).

8. DC-Removal PlugIn

8.3 DC-Removal Reference

The presence of the *DC-offset* can be examined separately for the left and right channels in the display. Momentary values are displayed as bar graphs. The exact peak values of the *DC-offset* are monitored in the numerical fields on the right side of the PlugIn window in bits. These values are permanently updated and hold on the highest level occurring to the current time point. By clicking the left or right numeric field, you can reset the numerical peak values anytime. The **DC-Removal** PlugIn then begins to cumulate the new *DC-offset* peak value again.

The *Help* button lets you jump to the chapter describing the **DC-Removal** PlugIn in this manual.

8.4 Applications Tips

By using the **DC-Removal** PlugIn, you can easily test the *DC-offset* and its drift in the *analog-to-digital converter* on your soundcard. The best way is to record a steady signal. A *sinewave* signal is recommended but not absolutely necessary. The recording level of the test signal has to be pretty high, just below any clipping occurs. The amount of the converter *DC-offset* can be read after playing back a few seconds of the test signal through the activated **DC-Removal** PlugIn. Repeating this test in an hour shows you the amount of the *DC-offset drift,* i.e., how strong the *DC-offset* changes over time. The high values of the *DC-offset drift* can be attributed to different recording quality over this time period. Especially large values can be expected when comparing *DC-offset* just after equipment switch-on and an hour or so later. An interesting check is to observe the *DC-offset* value during a longer time period to see if the *DC-offset* saturates, i.e., if it remains unchanged after warming up.

It is recommended that you always use the **DC-Removal** PlugIn as the very first PlugIn in the processing chain to prepare a *DC-offset* free signal for any other PlugIns. The improvement of the overall sonic quality, especially for low-level signals, can be considerable.

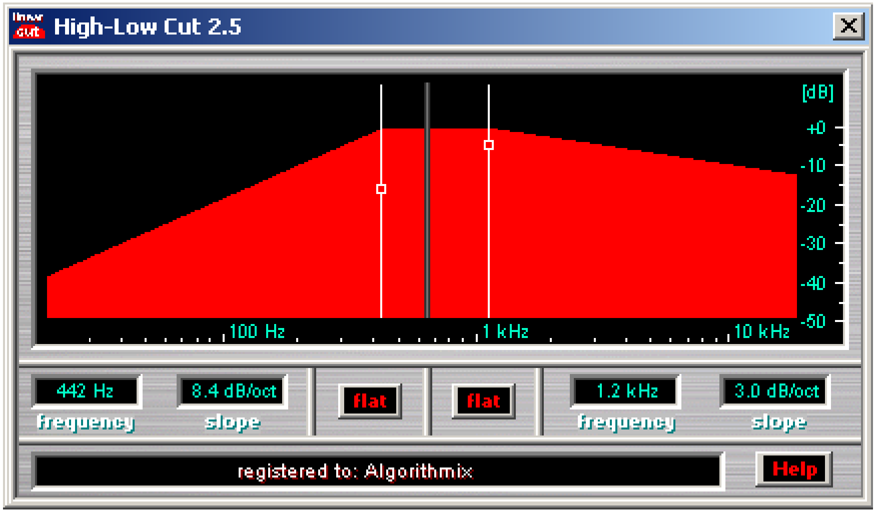
Due to high-precision design, the **DC-Removal** PlugIn can be chained into the **PlugIn-Station** by default without fear of any signal degradation, even in the case of perfectly recorded sample strings.

9. High/LowCut PlugIn

9.1 Overview

A *low-cut filter (high-pass filter)* sharply attenuates frequencies below a certain frequency (*cut-off frequency*). It is used to remove low-pitched noises such as studio rumble, microphone handling noises, vibration of microphone stands, microphone breath pops, and hum from guitar amplifiers. Another application is for cutting-off the table rumble of record players.

A *high-cut filter (low-pass filter*) sharply attenuates frequencies above a certain frequency (*cut-off frequency*). It is used to reduce hiss-type noises such as tape noise from low-quality cassette recorders.



Screenshot of the ***High-Low Cut*** *PlugIn*

Algorithmix® brings high-end professional technology to the multimedia user. The **High/LowCut** PlugIn represents a unique solution in the world of mastering audio filters. It contains two <u>*linear-phase filters*</u> (a *highcut* and a *lowcut*) with a <u>*continuously adjustable slope*</u> from 0 to 24 dB per octave.

9.2 Linear-Phase High/LowCut Filtering

In a typical mastering *low-cut* or *high-cut filter*, the attenuation rate (*slope*) can be adjusted to 6 dB per octave (*1st order filter*) or 12 dB per Octave (*2nd order filter*) and its *cut-off frequency* usually to some fixed values. In addition, a typical audio filter is a *non-linear phase filter*. That means different frequencies are delayed differently when passing the filter. The phase shift may spread the attack edge of impulse-like signals (e.g., drums) and blur the sound.

It is practically impossible to implement a *linear-phase filter* in analog technology. Digital signal processing technology made it a reality. Until now this type of filter has been used only in exotic high-end professional audio equipment. One reason for that is the high complexity of this filter; another is the actual studio praxis driven by the analog predecessors.

9. High/LowCut PlugIn

For the first time in digital audio technology, the **High/LowCut** PlugIn uses *low-cut* and *high-cut* filters with continuously adjustable *slope* ranging from 0 to 24 dB per octave. Even the most advanced filters used in the digital audio domain have only fixed slopes of 6, 12, or 18 dB per octave. Slopes lower than 6 dB per octave and slopes like 9, 13, or 16 dB per octave are impossible to implement with common filter architectures; slopes higher then 12 dB per octave (but still in 6 dB steps) are normally achievable only by cascading more filters. The **High/LowCut** PlugIn is a novel audio processing tool, allowing a filter with any slope you like between 0 and 24 d per octave. A slope of 24 dB per octave corresponds to a classical filter of the *4th order*!
The *cut-off frequency* of the *low-cut* filter can be continuously adjusted in the range from 20 to 600 Hz, the *cut-off frequency* for the *high-cut filter* in the range from 700 Hz to 20 kHz.

Changing the parameter of the filters during playback causes no audible artifacts. Therefore you can safely start your wave-file player and look for the best sounding result while adjusting the filter parameters live.

9.3 High/LowCut Reference

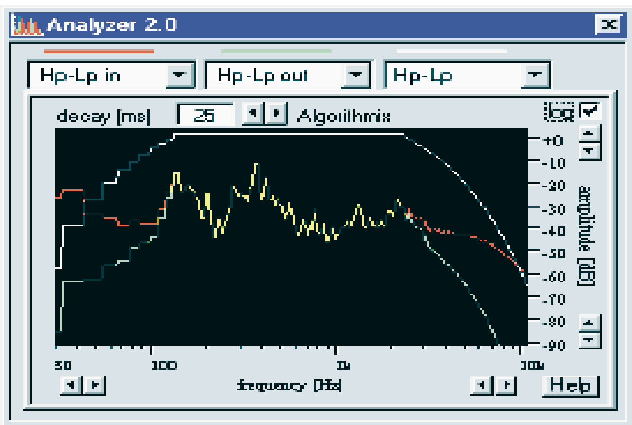
The parameter set-up interface is displayed in an intuitive graphical form. You can easily change the *cut-off frequency* and *slope* of the *hig-hcut* and *low-cut* filters by clicking and moving the white square on the proper marker while holding down the left mouse button.

For the technically oriented user, both the *cut-off frequency* and the *slope* for the current setting also appear in numerical form below the graphic window. If you move the sliders with the mouse, you can get only a limited resolution, e.g., your parameter increase every three unit (103, 106 ...). However, if you use the numerical window you can enter the parameter exactly (101.15). Directly click on the numeric field you want to change, enter the intended value, and quit with return.

The range of the *low-cut* filter *cut-off frequency* extends from 20 Hz to 600 Hz, the range of the *high-cut* filter *cut-off frequency* from 700 Hz to 20 kHz. The *slope* is continuously adjustable for both filters in the range between 0 and 24 dB per octave.

The respective *Flat* button switches the *low-cut* and *high-cut* filters to a linear characteristic that does not affect the input signal. This function is recommended as a starting position before setting up a new filter characteristic.

For a detailed technical evaluation of the filter characteristics and their influence on the input signal, we recommend using the **Analyzer** PlugIn.



***Analyzer** with setting for the **High-Low Cut** PlugIn.*

9. High/LowCut PlugIn

By setting the Analyzer to *High/LowCut in* (red line), *High/LowCut out* (green line), *High/LowCut* (white line), and the upper limit of the amplitude scale to 0 dB, you can easily follow the effect of the high/low-cut filtering on the processed audio material.

Dependent on sound material to be processed, the performance of the **PEQ linearPhase** PlugIn can be slightly modified using different settings for the parameters *Time Resolution* and *Frequency Resolution.* These are common for all frequency-domain PlugIns and therefore located in the *DSP Quality* window associated with the **PlugIn Station**™ or placed in the **AlgoX**, DirectX Load Frame (if you use DirectX mode). The *low-cut* filter works best with a high value of *time resolution*. If your PC is too slow and can not accept higher settings of the *time resolution* parameter in real-time, you can use the offline mode in the final processing step.

9.4 Applications Tips

The **High/LowCut** PlugIn is a unique tool that does not change the phase relationship in the signal, unlike other steep high/low cut filters. Therefore it is ideal for final mastering if no additional sound coloration of the final mix can be tolerated.

Some typical applications for the *low-cut* and *high-cut* filters have already been mentioned in the Overview. There are some other sound processing situations that make both filters very useful.

Recordings from old gramophone records can be made pleasant again by cutting turntable rumble, low frequency resonances in the pick-up (*low-cut* function), and hiss (*high-cut* function). If you need a *low-cut* filter with a very low *cut-off frequency* and the *slope* continuously going below – 100 dB, set up to a high frequency resolution in the *DSP Quality Window* or try our **DeRumble** PlugIn..

Sometimes audio material recorded with low-quality equipment sounds either too shrill or too dull. This is often due to the wrong tone balance between high and low frequencies from the psychoacoustical point of view. In this case it helps to cut the proper end of the frequency bandwidth.

Background music normally heard in stores or public places is not hi-fi quality. It is cut off at the low and high end of the frequency spectrum to make it less pushy and more stimulating. The *low-cut* and *high-cut* filters can also be used for special effects like the simulation of telephone call quality.

When using these filters as a technical mastering tool, remember that sometimes noise you want to remove has spectral components in the range of the useful program content. In this case you have to be very careful adjusting the *cut-off frequency* and *slope*. A typical example is the compromise between hiss and higher frequencies of the audio signal. If it is not necessary, do not use very steep attenuation; gradual sloping is usually more “musical”.

We do not recommend chaining the **High/LowCut** PlugIn before the **De-Scratcher** PlugIn or the **De-Noiser** PlugIn. To achieve the best performance, it is better if those PlugIns get the unprocessed sound material. However, after *de-scratching* and/or *de-noising*, an additional signal make-up (frequency range correction) with the **High/LowCut** PlugIn can be very useful.

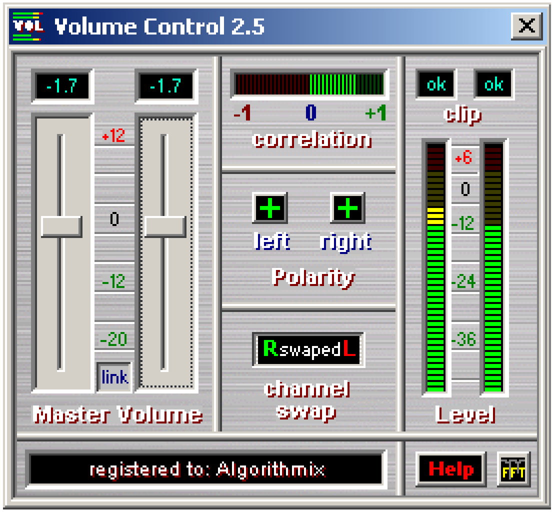
To hear the part of the input signal that is actually filtered out, use the *difference* function in the **PlugIn Station**.

If complex sound shaping is required use **Algorithmix** equalizer PlugIns: **PEQ classic.** If you need to remove discrete frequencies precisely, try the **NotchFilters** PlugIn.

10. SignalControl PlugIn

10.1 Overview

The **SignalControl** PlugIn allows adjusting the output volume and the polarity of every stereo channel separately. A channel swap is possible if the left and right channels have been exchanged. A precise *Level Meter* with *Clip Indicator* is provided in order to guard a correct output level. To maintain the mono compatibility a *Correlation Meter* is also included. The **SignalControl** PlugIn is normally intended as the last module in the audio chain, but if necessary it can be also placed before frequency-domain PlugIns by pressing the **[FD]** button.



*Screenshot of **SignalControl** PlugIn*

10.2 Volume Control and Level Meter

The *Master Volume* sliders can be use separately or coupled. The separate use is recommended if the left/right balance has to be corrected. In this case different level for every channel has to be applied. If the left and right channel has correct stereo image (proper channel balance) couple the both channel with the [link] button placed between both sliders. This cause that one slider pulls the second slider to exactly the same level.

The *Master Volume* can be set up even over 0 dB. It is normally not recommended and therefore indicated by red numbers in the numerical fields over the sliders. Because in the digital audio 0 dB is the maximal possible level, the use of the 0 to +12 dB range makes sense only for week signals. If setting up the *Master Volume* watch carefully the *Level Meter* and the *Clip Indicator* showing the number of subsequent clips. Normally you have to avoid frequently clipping and high numbers of subsequent clips. If they occur very sporadically two or three subsequent clips are normally not audible. To avoid overloads check the setting of the *Master Volume* for the whole piece and find the proper setting for the worst case; sometimes there are very loud parts which are strongly clipped if the *Master Volume* is too high.

Note that output levels over 0 dB shown on the *Level Meter* are acceptable only internally. For the external playback all signals are clipped to 0 dB because of the compatibility to the sound cards.

10. SignalControl PlugIn

10.3 Signal Polarity and Channel Swap

Sometimes particular stereo recordings are afflicted with phase errors or reversed channels. Phase or polarity errors can be caused by wrong mixing console settings. Such recording sounds out-of-phase, i.e., the stereo image is not correctly distinguishable. In addition, such stereo signal is no more mono compatible, i.e. summing of the stereo signals to mono leads to gaps in level and frequency, especially low frequencies are significantly weaker. The phase of every channel can be inverted individually with the +/- polarity buttons. Check mono compatibility before and after changing the polarity using *Correlation Meter*. In stereo signal it is enough to invert any of the two channels if they are out-of-phase.

Channel reversing is quite frequent error. Normally it arises from cable reversing in the subsequent devices of a audio chain. Using the channel swap button you can correct it exchanging left and right.

10.4 Correlation Meter

The *Correlation Meter* it is a important tool in studio and broadcast stations. In studio it is used to check if the left/right stereo channels are of the same polarity and in-phase. It can happen that one of the channels by mistake has inverse polarity. In such instruments or vocals normally coming from the center between loudspeakers are weak and shaky, and the stereo image not existing. As the next consequence, the mono signal after summation the stereo channels is also weak and often without low frequencies. This so called mono compatibility is what broadcaster test before transmitting program.

Sometimes, despite of correct polarity some stereo recordings show a out-of-phase character. This can be caused by very intensive use of effects like flanger, phaser or chorus. Such recordings may sound interesting in stereo mode, but they are not useful after summation to mono mode.

If the left and right channels are identical, the *Correlation Meter* displays +1. For mono compatible signals the value should fluctuate between +0.5 and +1. If the value changes between +0.5 and –0.5 the signal is not mono compatible. If he signals are of inverse polarity the value fluctuate between –0.5 and –1.

10.5 SignalControl Quick Reference

- **Master Volume** – set up the volume for each channel from –24 dB to + 12 dB (exact numerical value can be enter via the numerical fields over the sliders, the numbers become red if the level is over 0 dB)
- **Link** – couple the both volume sliders
- **Channel swap** – swaps the left and right channel at the output
- **Polarity** – allows to change the phase of each input channel: **+** normal, **–** inverse (180°)
- **Correlation** – shows the phase difference between channels: +1 in-phase, -1 out-of-phase
- **Level** – shows the output level for each channel
- **Clip** – shows the number of subsequent clips (above 0 dBFs) in output signal
- **FD** – toggles the positions of the PlugIn in audio chain between *pre* nad *post* fraquency-domain PlugIn block (see **Recommended Configuration**)
- **Help** – opens the appropriate help file

10. SignalControl PlugIn

10.6 Applications Tips

Normally the **SignalControl** PlugIn is intended as the last module in a audio chain to check the phase and mono compatibility of the stereo signals, as well as to correct the final volume to prevent overload and digital clipping. Especially if you used PEQ it may happen that the output level exceeds 0 dB.

Of course, you can raise the level if the whole piece is too quite, but be aware that you also raise the noise. In this case it is recommended to place the **SignalControl** PlugIn at the beginning of the audio chain, before processing modules.

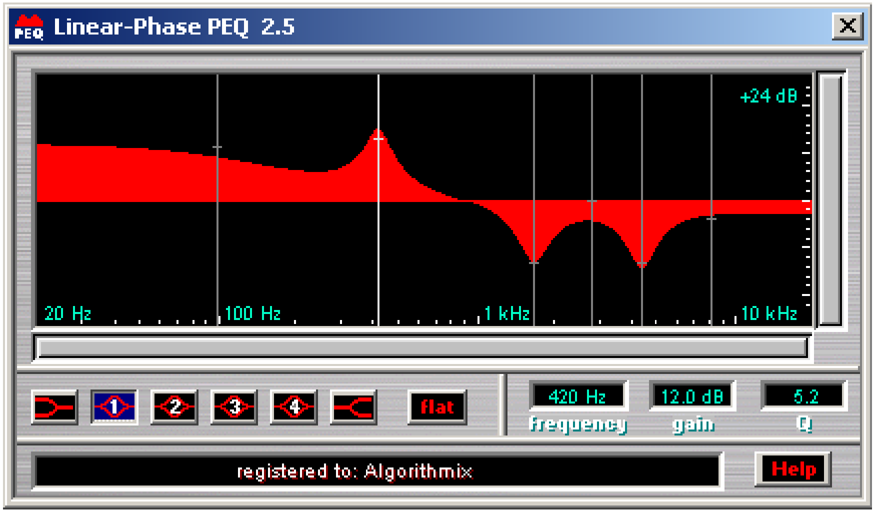
If the *correlator* value stays between 0 and –1, the stereo signal is out of phase. Try to change the polarization of the L or R channel. If it still does not help, it may be an indication that the signal was treated by a phasing effect like stereo chorus or flanger.

11. PEQ classic PlugIn (optional)

11.1 Overview

Equalizers, popularly called EQ, are one of the most important sound-shaping tools. While the level control changes the amplitude of the whole signal, the equalizer can selectively alter its parts. This means that signal components at certain frequencies can be increased or decreased in amplitude leaving the others unaffected. While working with an equalizer you will be cutting or boosting signal frequencies expressed in *Hz* by levels expressed in *dB*.

The **PEQ classic** PlugIn is a *fully-parametric* equalizer including six *minimal-phase* filters: *low shelving*, four *bell filters*, and *high shelving*. All filters can be set independently using graphical curve representation or numerical fields. For the shelving filters, the *corner frequency* and *boost/cut* amount can be set. For *bell* filters, *center frequency*, *boost/cut*, and *Q-factor*. For more boost or cut, the bell filters can be overlapped.



Screenshot of the ***PEQ classic****.*

For advanced *de-noising* applications a combination of two equalizers could be effective. For that reason, two versions of the **PEQ classic** are available: *pre-cleaning* and *post-cleaning*. This means the **PrePEQ classic** PlugIn should be used before **DeNoiser** and **PostPEQ classic** after the **DeNoiser** PlugIn (see last chapter for more details).

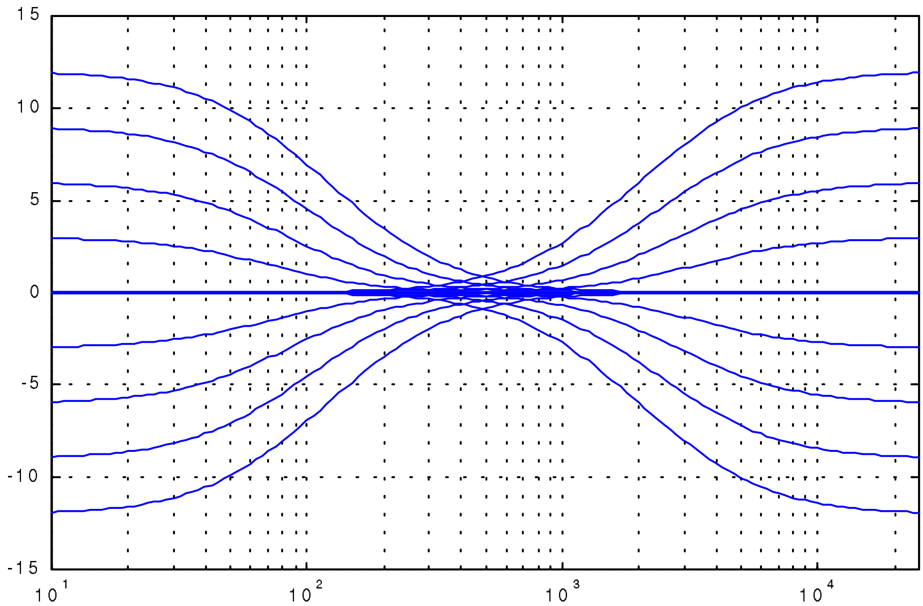
When using a few different equalizers simultaneously (**PrePEQ classic**, and **PostPEQ classic**), you can copy and paste the settings between them using *ctrl-c* (copy) and *ctrl-v* (paste). A toggle between the last and the actual setting is available using *ctrl-t*.

The spectrum of EQ applications is very wide: improving tone quality, compensating for microphone placement, reducing unwanted noise, compensating for losses in recording equipment, balancing the mix, and introducing special effects.

11. PEQ classic PlugIn (optional)

11.2 Classical Equalization

There are many different types of equalizers. It's important to know the differences before using them. One possible criterion for differentiating equalizers is their frequency characteristics. The simplest is the standard bass/treble control typically used in stereo systems. The bass control is like a volume control for low frequencies, usually from 200 Hz down. The treble control changes high frequencies, usually above 4 kHz. This kind of equalization smoothly influences a wide range of frequencies and is called *shelving* type (Figure below). The **PEQ classic** also contains these equalizer filters called *low-shelving* and *high-shelving*.



Frequency characteristic family of low-shelving and a high-shelving filters

More sophisticated equalizers include one or more stages for control of middle-frequency ranges. The filter characteristic used here is the *bell* type. It allows you to adjust (*boost or cut*) the signal components in the range around the *center frequency* only, leaving the remaining frequencies unaffected. If the center frequency can be selected, we call such an equalizer *quasi-parametric*. If, in addition, the width of the bell curve is adjustable, such an equalizer belongs to the family of *parametric* or *fully-parametric* EQs. The width of the peaks or dips characterize the range of frequencies affected by the equalizer stage and is usually expressed as the so called *Q-factor* (Fig. Y). *Q* is the relation of the center frequency to the bandwidth taken 3 dB below the bell maximum (boost case) or 3 dB above the bell minimum. The higher the Q, the narrower the frequency range. The **PEQ classic** includes four *fully-parametric bell* stages.

If the bell filters are equally spaced over the whole frequency range, e.g., every octave, and Q is fixed to a certain value dependent on the frequency spacing, we call such EQs *graphic equalizers*.

11. PEQ classic PlugIn (optional)

Some equalizers, even very famous ones, have a few troublesome problems coming from the filter structure used for the implementation. The frequency characteristics for boost and cut are not symmetric, e.g., the peak is wider then the respective depth in spite of the same *Q* for both situations. This means the peak does not cancel the respective depth.

Another typical problem of many equalizers is that the Q-factor is dependent on the boost or cut amount. This means the frequency range affected by this parametric EQ stage is narrower if the boost/cut is larger and wider if the boost/cut is smaller. Modern equalizers which largely preserve the bandwidth while adjusting boost or cut have been found to be much easier to use. The **PEQ classic** belongs to this group which is called *constant-Q* equalizers (Fig. Y).

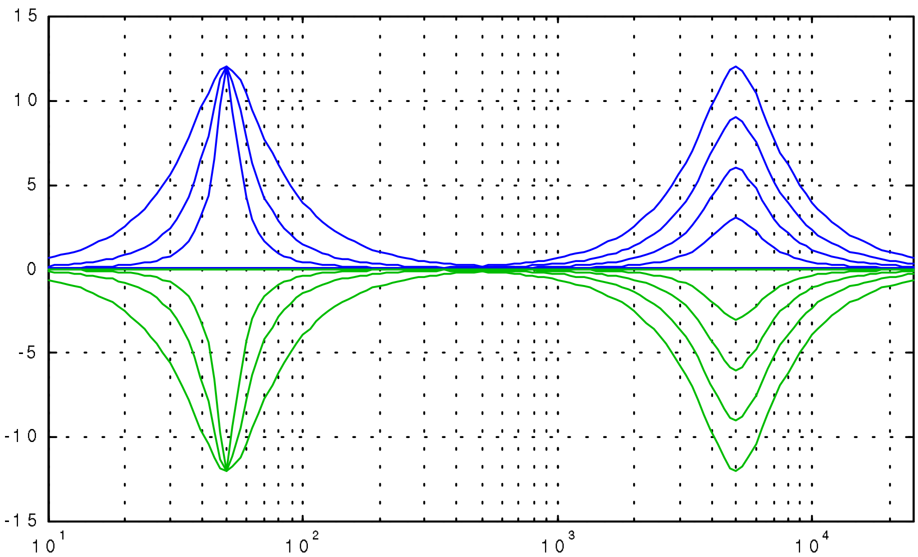


Fig. Y Different setups of a two-stage bell equalizer.

Almost all of the equalizers used in the world change not only the frequency characteristic (what we want), but also the phase characteristic of the signal (what we normally do not want). Many users do not even know about this. The reason is that the phase changes are inaudible for pure sinus tones and they cause only very subtle timbre changes to the real audio signals. Nevertheless, sometimes, especially if manipulating complex mixes, a wrong or exaggerated use of EQ can negatively change the artistic quality of a recording.

The concept of a liner-phase equalizer that affects only amplitudes of the frequency spectrum and leaves the phases unchanged is quite old. But it has been practically impossible to implement with analog technology. Digital signal processing supported by great computational power first made it possible to implement reproducible linear-phase filters. Very complicated algorithms, however, used in professional linear-phase audio equalizers make them very, very rare still. **Algorithmix**® is proud to offer you such a unique tool in the near future.

A linear-phase equalizer, however, has a property that can cause problems in some situations. It has a relatively long processing delay that is not relevant in playback situations, but can be annoying in live applications. Therefore a typical EQ like the **PEQ classic** still has its justification. It is based on the *minimal-phase* filters and therefore causes the shortest processing delay physically possible.

11. PEQ classic PlugIn (optional)

11.3 PEQ classic Reference

The **PEQ classic** parameters can be adjusted graphically through the filter display or numerically using the numerical fields and the keyboard directly.

First click the proper *Filter Button* to select the setup of the filter you want to adjust.

[>–] or **[–<]** – low shelving / high shelving
[<1>] to **[<4>]** – bell 1 to bell 4

Instead of using the Filter Buttons, you can left click on the center line of the filter you want to activate. Then the center line changes color from gray (inactive) to white (active).

A left double click on a *Filter Button* resets the gain of the associated filter.

The *Flat* button sets all filter gains to 0 dB but does not change the last *center frequency* and *Q-factor* settings.

All PEQ PlugIns (**PrePEQ classic**, and **PostPEQ classic**) allow you to copy and paste the settings between different windows using *ctrl-c* (copy) and *ctrl-v* (paste). A toggle between the last and the actual settings is available using *ctrl-t*.

The *Help* button lets you jump to the chapter describing the **PEQ classic** PlugIn in this manual.

Parameter Adjustment via Filter Display

To change the *corner frequency* for *shelving filters* or *center frequency* for *bell filters* as well as *boost* or *cut* amount (*gain*), use the left mouse button. Double clicking resets the *gain*. The *shift* key locks the frequency adjustment allowing you to concentrate on the gain adjustment only.

The width of the *bell* filters or its *Q* factor can be adjusted using the right mouse button. A double click resets the *Q* factor to the default value 1.0.

There are two scroll bars for very precise amplitude and frequency adjustments. Click on the edge of horizontal scroll bar to change the amplitude scale between +/- 2 dB and +/- 24 dB. Double-click on this scroll bar to switch between full scale (+/- 24 dB) and the last adjusted value. Click on the edge of the horizontal scroll bar to zoom the displayed frequency range from the full audio range (20 Hz to 20 kHz) to only one octave range. The narrowed range can then be swept over the whole audio frequency range, like a magnifying glass.

Parameter Adjustment via Numerical Fields

You can use the numerical fields for very precise adjustments of the *center (corner) frequency*, gain *(boost/cut),* and *Q*. Click on the proper numerical field, enter the numerical value of the parameter you want to change, and confirm with *return*.

11. PEQ classic PlugIn (optional)

11.4 Application Tips

The equalizer is a nice tool allowing quite spectacular sound manipulation with only few controls, but it is mostly not what you or your customer want. The real world is not only fun and therefore you need to follow some basic rules to get a great sound:

- Do not overuse your EQ. Often compare the original recording with its equalized version. Do not try to “fix” good sounding original instruments.

- Do not exaggerate with high Q values. In mastering application *Q* over 1.0 is quite seldom. Higher *Q* values are used only for “fixing” solo tracks; very high values make sense only for surgical sound manipulations, mostly to remove or to lower unwanted spectral components.

- If you only need to gradually emphasize the low-end or high-end frequencies for loses in your analog recording equipment use the low-shelving and high-shelving filters which allows a smooth good sounding corrections.

- Strong boosting of wide frequency ranges with a weak original signal increases the noise level, too. If possible use the subtractive technique: keep the problematic frequency region unaffected and decrease all others. If necessary correct the overall level with **SignalControl** PlugIn.

- Use the subtractive technique to emphasize harmonics, i.e., to remove unwanted frequencies rather then accenting those you like. The idea behind is to cut the fundamental a few dB. Practically you can start with boosted EQ sweeping the bell until you get the fundamental much louder. Once you hit it, change the characteristic from boost to cut and look for the proper adjustment.

- Be aware that increasing certain frequency region can cause signal clipping at the output of the sound card. Therefore for the signal amplitude correction use the **SignalControl** PlugIn, preferably after EQ. It is always better to go into the PEQ with the full possible amplitude and correct it first after processing, of course only if necessary. Using **Algorithmix** PlugIns you do not need to be afraid about any overflow inside and between the processing modules. They are implemented with up to 80-bit floating point processing resolution.

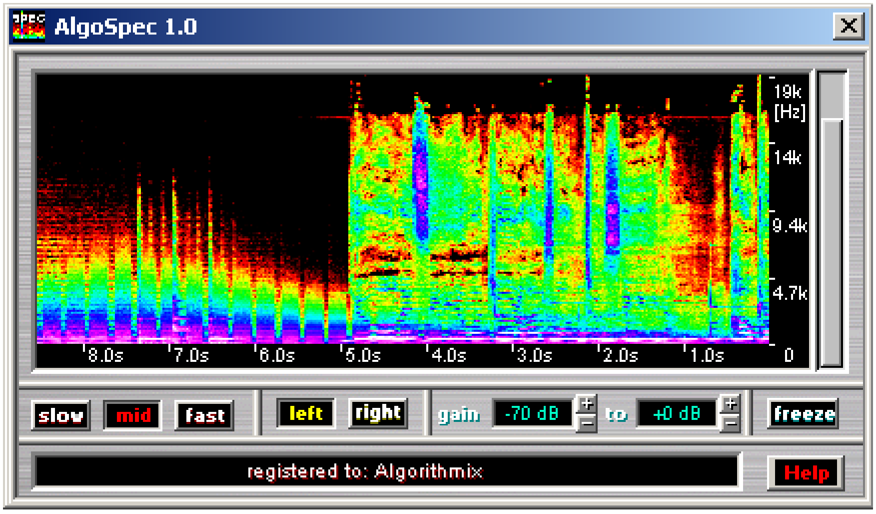
- For advanced filter application during the de-noising process two **PEQ classic** (*pre-FD* and *post-FD*) can be used in combination with the **DeNoiser** PlugIn. Such configuration allows emphasizing or suppressing certain frequencies prior to the de-noising and undoing the equalization (including the phase shifts) after de-noising process. To achieve this function it is necessary to set the gain of the *pre* and *post* PEQ in opposite keeping the same center frequencies. E.g., if a particular filter in the **PrePEQ classic** is set to 1 kHz, Q = 5, 7.3 dB, the same filter in the **PostPEQ classic** has to be set to 1 kHz, Q=5 and –7.3 dB. You can check this with the *difference* function; if the set up of the both PEQs is exactly opposite, input and output settings exactly cancel and there is no difference signal (assuming that any other PlugIn in-between is active).

- Before fixing instruments or vocals learn the ranges of fundamentals and harmonics associated with them. For warmth and fullness increase the lower end of fundamentals; if the tone is too bassy lower the fundamentals. Turn up the harmonics for presence and definition; turn them down if the tone is too harsh and sizzly.

12. AlgoSpec™ PlugIn (optional)

12.1 Overview

The **AlgoSpec**™ PlugIn shows the moving spectrogram of the audio signal processed by the modules activated in **PlugIn-Station**. As it is implemented in the frequency domain the signal can be displayed before and after every frequency domain PlugIn (**DeNoiser**, and **Low/HighCut**). Of course if there are time-domain PlugIns loaded in the audio chain before frequency-domain group, it is possible to observe spectrum after the time-domain PlugIns.



*Screenshot of the **AlgoSpec**™ PlugIn*

The **AlgoSpec**™ PlugIn shows the analyzed audio signal in the three dimensional view. The vertical axis is associated with the frequency of the signal spectrum, the horizontal axis shows the time, and the colors represent the intensity for every spectral component. The intensity scale corresponds to the rainbow colors: black, red, orange, yellow, green, blue, violet, and white. The black color represents zero intensity, the white color maximal intensity. The lowest and the highest level assigned to the black and white colors can be defined in the two numerical gain fields.

The diagram moves from the right to the left after starting playback. There are three different speeds possible. In the **[slow]** position, one spectrum slice per audio block (as specified in the *Device I/O* settings of the PlugIn-Station) is plotted. In the **[mid]** position one spectrum slice per processing block (as specified in the *Quality Window*) is added, in the **[fast]** position one spectrum per each oversampled block. This means, that if the *oversampling* parameter in the *Quality Window* is set to one, the **[mid]** and **[fast]** result in the same diagram speed. The moving diagram can every time be freeze with the **[freeze]** button.

To observe particular details of the moving spectrum the frequency axis can be zoomed as narrow as 2 kHz. In addition, this 2-kHz window can be shifted over the whole 20 Hz to 20 kHz range. To change the range click on the edge of the scroll bar and decrease or increase the range as required. To chose a particular window of interest click on the middle of the scrollbar and shit it up or down observing the frequency scale on the right of the display.

12. AlgoSpec™ PlugIn (optional)

12.2 AlgoSpec™ Quick Reference

slow / mid / fast – update speed depended on audio block size and *Quality Window* settings
left / right – the audio channel to be displayed
gain – gain range assigned to the color scale: black and red for low gain, white for high gain
freeze – freezes the momentary display
frequency scrollbar – a frequency window of interest can be adjusted: zoom up to 2 kHz placed anywhere between 20 Hz and 20 kHz

12.3 Applications Tips

The **AlgoSpec™** PlugIn allows investigating suspicious signal anomalies which are difficult to hear. Discrete high-frequency tones coming from switching signals or interferences with digital equipment can be successfully localized. Even if they have very low level they appear as a horizontal line in the moving spectrum. You can then use the **NotchFilters** PlugIn to remove these discrete tones. You can very precisely adjust a notch filter watching the moving spectrum and changing the center frequency of the notch.

Wow and Flutter can be also detected quite easy. They appear as the modulation of the low-frequency components. Similar behavior can be observed if looking on the spectrogram of a voice with vibrato. In that case the fundamental frequency and quite often the associated formants looks like sinusoidal horizontal lines.

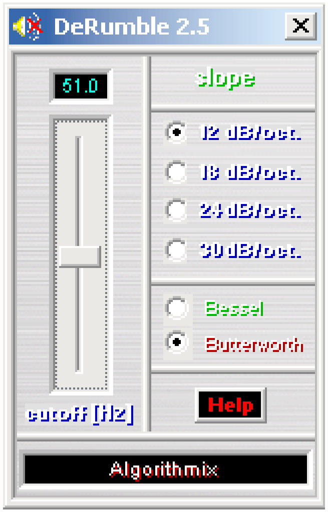
Clicks are visible as bright vertical lines.

In forensic applications a abrupt spectrum change is a significance that two pieces of audio material was “glued” together. Monotone horizontal lines can indicate particular devices like particular type of fun or refrigerator.

13. DeRumble PlugIn (optional)

13.1 Overview

The **DeRumble** PlugIn is intended to attenuate or eliminate low-frequency and subsonic signals. It includes cascaded *high-pass* filters with a stop-band attenuation for very low frequencies greater then it is possible with the *low-cut* filter being a part of the **High/LowCut** PlugIn.. It is very useful for removing the rumble of the turntable, as well as wind and plop noises coming from the microphone.



*Screenshot of the **DeRumble** PlugIn*

The cut-off frequency can be continuously adjusted between 4 and 100 Hz. Four different setups for slope are available. The higher the slope, the stronger the attenuation in the stop-band, but also stronger the phase shift. Because phase shift can influence the timbre of the instruments being close to the chosen frequency range it is not recommended to use steep filters if it is not really necessary.

To allow different combinations between filter slope and sound coloration two different filer characteristics are provided: *Besel* and *Butterworth*. Besel characteristic has a quite smooth roll-off, but a constant delay for every frequency. It is a very important if we want to maintain the natural timbre and transient characteristic of the original signal. The Butterworth characteristic has a steeper roll-off, but is still free of ripples in pass-band (like elliptic or Tschebyshev filters) and with relatively negligible phase shift. It is suitable when higher attenuations are more important then possible sound coloration.

[HP frequency characteristic]

HighPass Filter with set up to 10Hz, 12dB/oct. and to 100Hz, 24dB/oct.
(blue - Bessel, green - Butterwoth)

13. DeRumble PlugIn (optional)

13.2 DeRumble Quick Reference

- **cutoff** – set up the 3 dB cut-off frequency of the high-pass filter in the range 4 to 100 Hz.
- **slope** – determines the slope of the filter in four steps: 12, 18, 24, and 30 dB/octave
- **Bessel/Butterworth** – switch the filter characteristic between *Bessel* (with the constant group delay) and *Butterworth* (with the lowest ripples in pass-band)

13.3 Applications Tips

Note that due to phase shifts, the difference function makes audible not only low-frequency attenuated signals, but also parts of the signals influenced by phase shift causes by the filter.

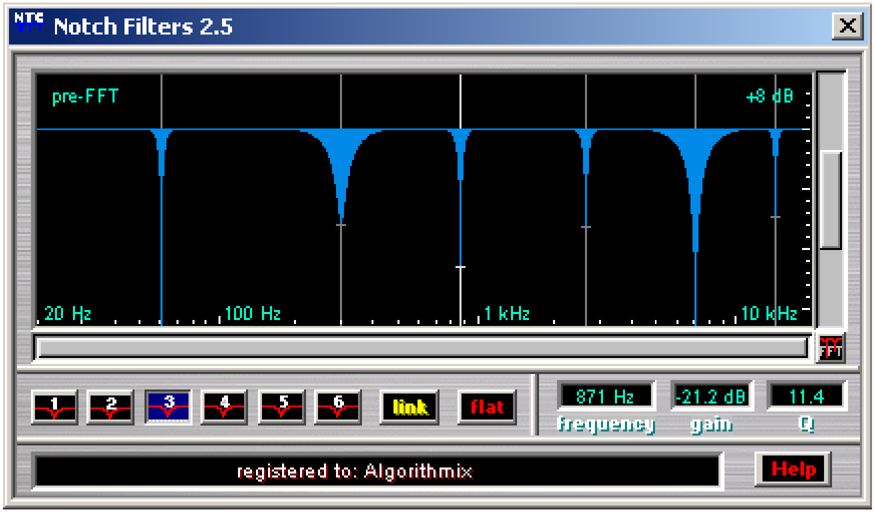
If you use the filter to cut off rumble of a turntable, a setting of 20 to 30 Hz with 12 or 18 dB/octave slope should be sufficient.

14. Notch Filters PlugIn (optional)

14.1 Overview

A *notch* filter, called also *band-stop* filter, sharply attenuates a narrow band of frequencies. In general, it is similar to a parametric *bell* filter in cut position, but usually much narrower. Thus it allows taking out or attenuating particular frequencies without substantially affecting the remaining signals.

Notch filters are mostly use to eliminate mains hum and its harmonics, as well as other disturbances concentrated at discrete frequencies. They help in cleaning film-location sound, live interviews in noisy environment and critical recordings in the field of forensic applications.



Screenshot of the ***Notch Filters*** *PlugIn*

The **NotchFilters** PlugIn includes six independent *band-stop* filters. Their center frequency, cut gain, and bandwidth can be individually adjusted. The relative bandwidth is defined in term of so called Q factor which express the ratio of the particular center frequency to the bandwidth taken at -3 dB below zero. Using Q and not the absolute bandwidth in Hz let all filters having the same Q and gain look exactly of the same wide independent on the position on the logarithmic frequency scale. It corresponds better to the way how our ears respond to the bandwidth.

Note that the bandwidth for notch filters is defined at –3dB below zero and not –3dB above the negative peak like in parametric equalizers. If comparing a cut bell of a PEQ and a notch having the same amplitude and the same nominal Q factor the notch is much more narrower then the PEQ bell.

For special applications like removing hum and its harmonics it is possible to couple all the six notch filters on the frequency axis, so, that the distance between two adjacent filters expressed in Hertz is identical. It is very useful function saving time needed for precise adjustment.

14. Notch Filters PlugIn (optional)

14.2 Notch Filters Reference

The **NotchFilters** PlugIn parameters can be adjusted graphically through the filter display or numerically using the numerical fields and the keyboard directly.

First click the proper *Filter Button* to select the setup of the filter you want to adjust.

[¯\1/¯] to **[¯\6/¯]**– notch 1 to notch 6

Instead of using the Filter Buttons, you can left click on the center line of the filter you want to activate. Then the center line changes color from gray (inactive) to white (active).

A left double click on a particular *Filter Button* resets the gain of the associated filter.

The *Flat* button sets all filter gains to 0 dB but does not change the last *center frequency* and *Q* settings.

The position of the **NotchFilters** PlugIn can be changed between pre- to post-FFT with the [FFT] button.

The *Help* button lets you jump to the chapter describing the **NotchFilters** PlugIn in this manual.

Parameter Adjustment via Filter Display

To change the *center frequency* of the *notch filters* in the range 20 Hz to 20 kHz as well as *cut* amount (*gain*) in the range 0 to –60 dB, use the left mouse button. Double clicking resets the *gain*. The *shift* key locks the frequency adjustment allowing you to concentrate on the gain adjustment only.

The width of the *notch* filters or its *Q* factor can be adjusted between 3 and 60 using the right mouse button. A double click resets the *Q* factor to the default value 3.0.

There are two *scroll bars* for very precise gain and frequency adjustments. Click on the edge of vertical *scroll bar* to change the amplitude scale between -8 dB and - 60 dB. Double-click on this *scroll bar* to switch between full scale (- 60 dB) and the last adjusted value. Click on the edge of the horizontal *scroll bar* to zoom the displayed frequency range from the full audio range (20 Hz to 20 kHz) to only one octave range. The narrowed range can then be swept over the whole audio frequency range, like a magnifying glass.

Parameter Adjustment via Numerical Fields

You can use the numerical fields for very precise adjustments of the *center (corner) frequency*, cut gain, and *Q*. Click on the proper numerical field, enter the numerical value of the parameter you want to change, and confirm with *return*.

If you want to use the whole set of the notch filters for removing a particular frequency and harmonics associated with it, set the first notch to the fundamental, the second notch to the second harmonic and click [LINK] button. The remaining four notches will be automatically placed at third to sixth harmonic. E.g., if the first notch is on the mains frequency 60 Hz and the second on the 120 Hz, the remaining notches will be placed on 180, 240, 300, 360 Hz after clicking on [LINK]. For the best precision use numerical fields.

14. Notch Filters PlugIn

14.3 Applications Tips

Note that due to physical considerations deep notches and high *Q* values (narrow filters), especially at low frequencies require significant time to reach the supplied values. They also produce large phase shifts at the center frequency. Therefore do not exaggerate with extreme parameter settings if it is not necessary.

To search for harmonics use the **Analyzer** PlugIn set up to linear scale (check-box *log* off).

If you have to cope with harmonics number grater then six you can use two subsequent runs with different filter adjustments.