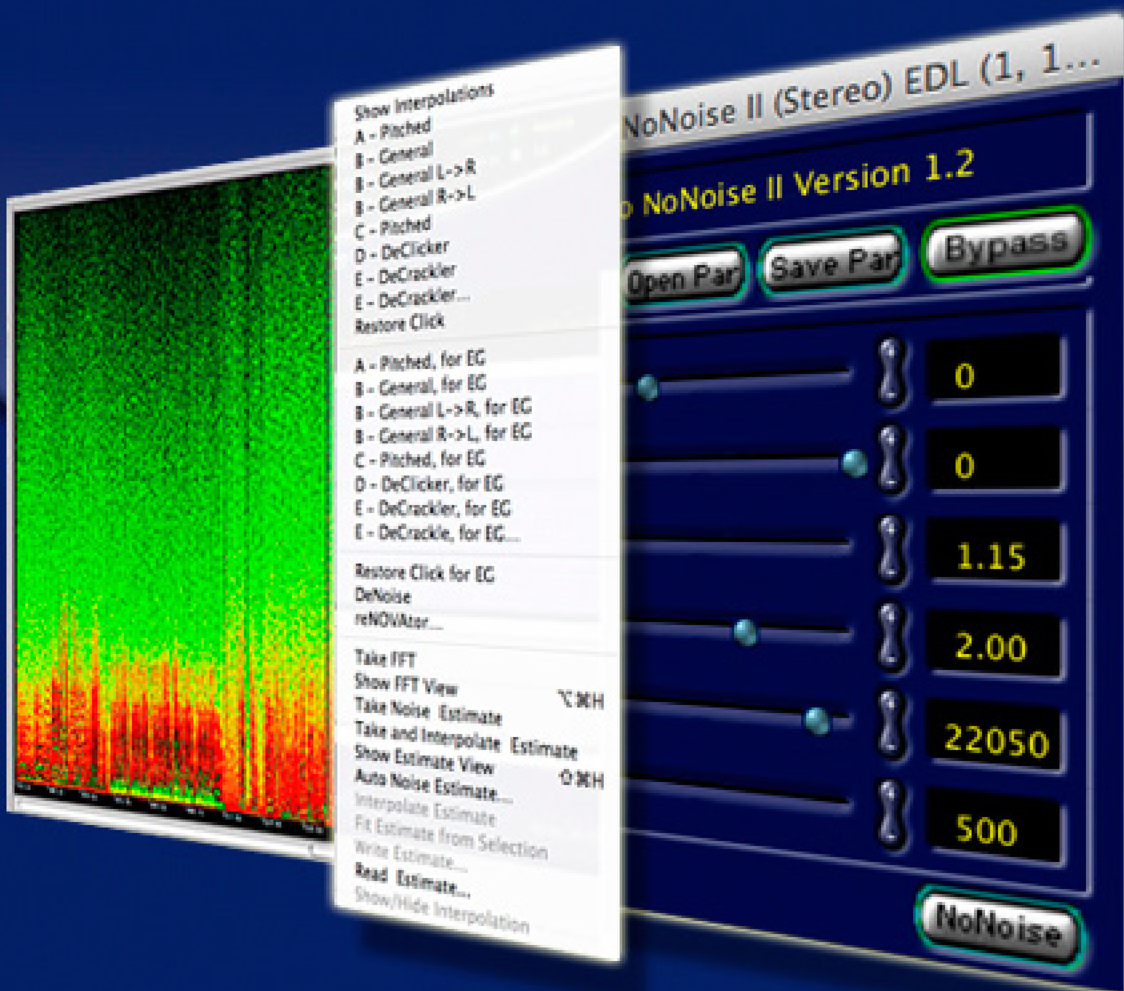
NoNOISE II SR

User Manual



High Resolution Audio Restoration & Enhancement Tool

for soundBlade

Sonic Studio

reNOVAtor Legal Notice

http://www.algorithmix.com

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NoNOISE II SR User Manual Table of Contents

Chapter 1 .... NoNOISE II SR

NoNOISE II is the world's premier tool for restoring vintage and problematic audio recordings. NoNOISE II SR includes the entire NoNOISE II toolkit, including Manual Declick, complete Broadband DeNoise, Production DeCrackle FFT (Fast Fourier Transform) generation and full Sonic EQ. Plus, NoNOISE II SR adds the full implementation of Algorithmix's reNOVAtor.

The Manual DeClick toolset precisely removes impulse noise, analog or digital overloads, harmonic and intermodulation distortion, and provides a unique solution to obscenity masking. The Broadband DeNoise option is the highest fidelity single–ended noise reducer available today and can rescue both old and new recordings alike.

Production DeCrackle is the background or faster than real time version of theType E interpolator. The DeCrackler, both manual and background versions, provide unparalleled distortion reduction for a segment or an entire sound file.

Sonic EQ is a minimal phase EQ toolkit, providing seventeen different topologies, and is suitable for both repair, enhancement and correction, while Algorithmix's reNOVAtor is the high performance spectral repair tool. Unlike work–alike products, reNOVAtor is able to dig into recordings with very low signal–to–noise ratios and pull out usable program. Plus, the ability to cut and paste time and spectral patches and its harmonic noise reducer functions all add to its amazing usefulness.

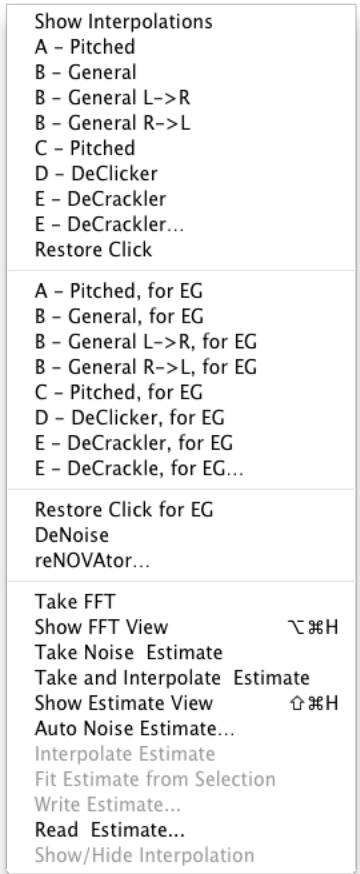


Figure 1.1: The NoNOISE II menu

NoNOISE II Options Installation —

NoNOISE II requires the installation of both an iLok license and Sonic Studio Options Files for NoNOISE II FX, NoNOISE II FR or NoNOISE II SR to be enabled for use. Running the appropriate Options Installer, located in your soundBlade app folder's Sonic Options folder, installs the necessary software.

This does not install the actual option software, it merely enables specific options you have licenses for on your iLok.

NoNOISE II Options Notes:.

1) Options are installed on the Boot Volume at:

/Library/Application Support/Sonic Studio/Sonic Options 2.0/

2) Use the Remove NoNOISE II Options utility for removing options.

NOTE: Only install NoNOISE II options that you have licenses on your iLok for. Installing options that are not on your iLok will make soundBlade launch *very* slowly.

1.1 Manual DeClick

The Manual DeClick option provides tools for isolating and removing individual transient impairments in a sound file. Manual DeClick assists in removing unwanted noises such as clicks, pops and thumps. It offers five different interpolation algorithms that are capable of correcting even difficult audio anomalies.

The algorithms analyze audio on either side of the anomaly and, based on this information, synthesizes replacement samples. Manual DeClicking substitutes the repaired samples for the original program material.

Note that Manual DeClick processes generate two new files for each sound file that is repaired. These ".cd" and ".rl" files contain the samples removed during repair and a list of their locations, respectively. If you move or delete either of these files, you will not be able to restore the original samples, undoing the repair.

With the exception of the Type E interplator, Manual DeClick is generally used on very short duration regions of 14 msec. or less. Though the algorithms are capable of credible repairs over longer durations, only experience using each algorithms will allow you to judge sensible parameters.

1.1.1 Interpolation Algorithms

There are several interpolators that are available in soundBlade. Each is suited to a particular type of audio problem and context.

1.1.1.1 The B Type General Interpolator

The Type B interpolator is the general purpose algorithm. The majority of declicking situations can be handled by simply choosing this option.

The default Type B interpolator examines the audio on either side of the selection to determine the context for resynthesizing audio to fill the gap. For the B Type, there are two additional variations of the command that bias the context in a particular "direction," ignoring the material before or after the impairment.

If, for example, a click occurs just after a percussive event, the default Type B interpolator would include part of the percussive event in its resynthesis, producing a unconvincing repair. The B - General R-> L option would ignore the audio to the left, using only the samples after the selection is build the repair. Conversely, the B - General L-> R option uses only samples before the selection to resynthesize a repair.

Here are three examples:

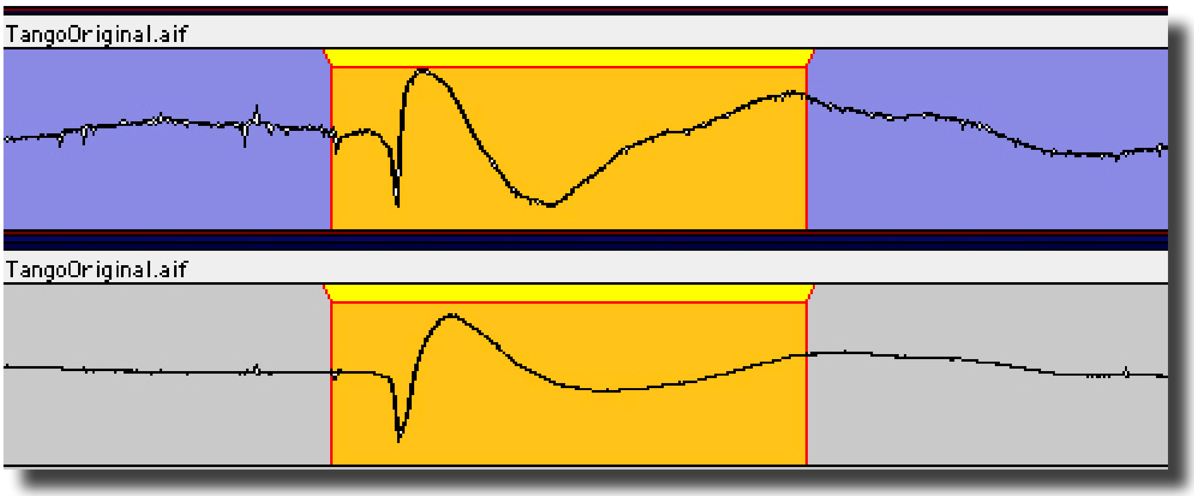


Figure 1.2: The original material

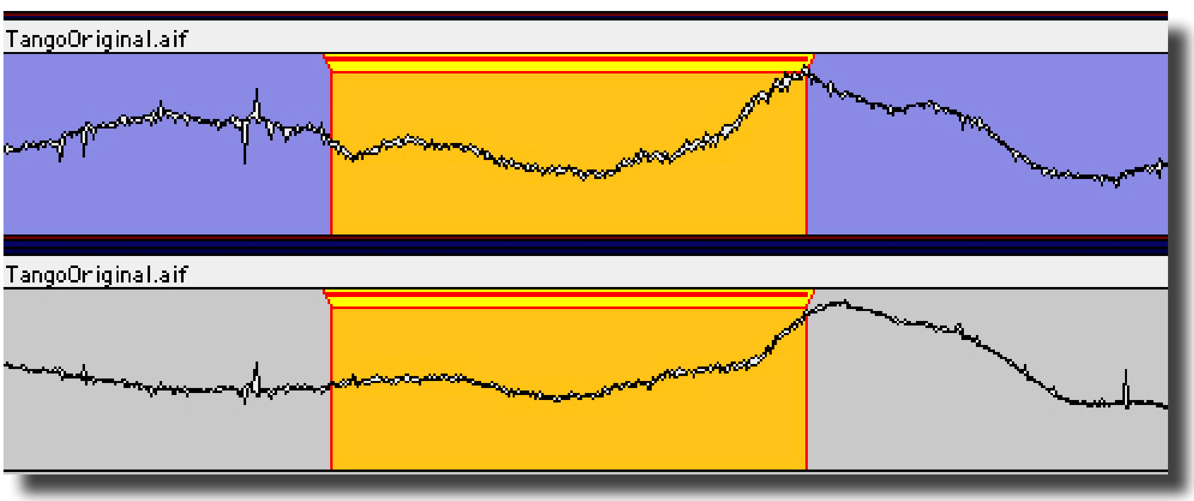


Figure 1.3: A "repair," using the default B Type algorithm

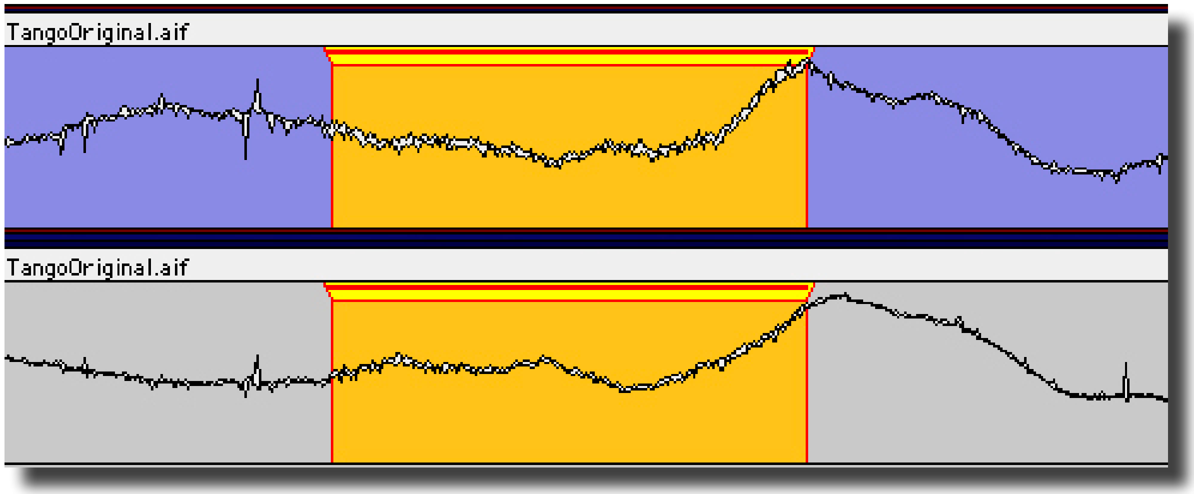


Figure 1.4: A better "repair," using the B - General L-> R option

Though the above examples are extreme, a 60 msec. selection to clearly show the result, they should illustrate the concept of using the left–oriented or right–oriented B Type option when needed.

1.1.1.2 The A & C Type Pitched Interpolators

The Type A and Type C interpolators are designed for pitched or periodic material, such as solo instruments or any time the fine structure of a waveform is visually repetitive. The difference between these two options is that the C Type has validation built into it for certain cases in which the interpolator may produce less than perfect results. The A Type lacks these "protections," so its results may occasionally be unusable. In addition, both the A and C Type may fail to find an "acceptable" periodicity and will alert you to try a different algorithm. If this occurs, try the D Type or, select a slightly different region of audio and try again.

The A and C Type interpolators take contextual information from six periods to the left and right of the selection. After interpolation, the Restore Bar may extend for some distance outside the selected region. This is because these interpolators are repairing based on wavelength in addition to simple selection duration.

1.1.1.3 The D Type DeClicker

The D Type and E Type DeClickers, discussed below, are very high-order algorithms used to correct problems that elude other repairs. Both interpolators use 64 bit precision to produce very high quality interpolations.

The Type D DeClicker is tailored for use on the human voice, though it will provide excellent results on most any semi–periodic material. It is only capable of replacing about 80 milliseconds worth of samples before it bogs down. Even so, a repair half that duration will still take quite a while, even on a fast computer.

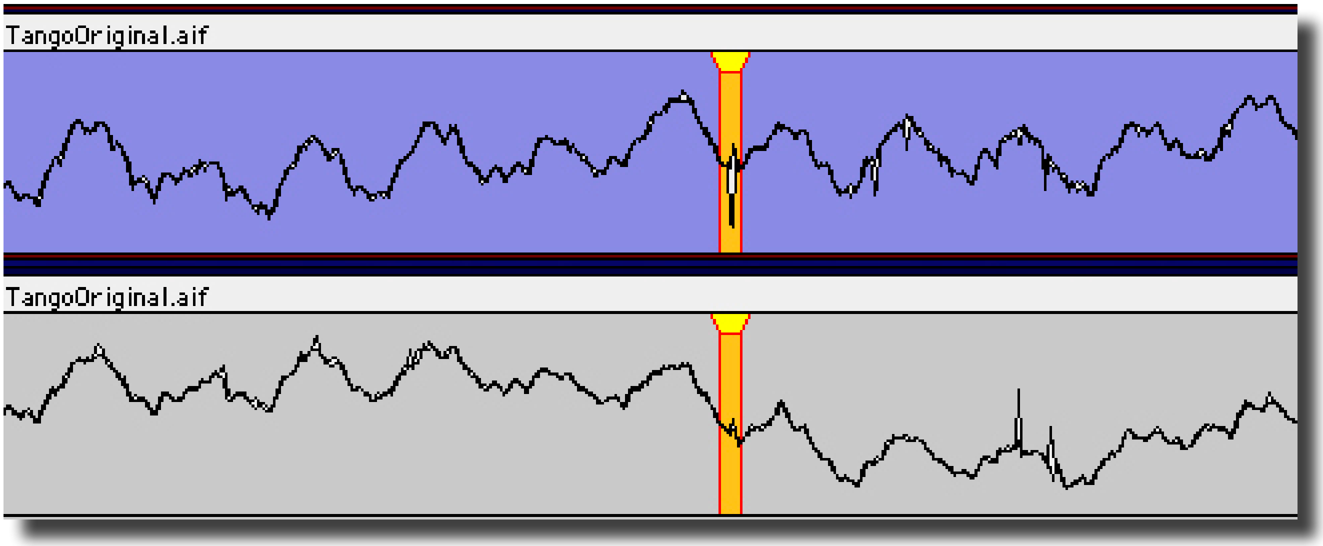


Figure 1.5: A fairly periodic section with click in Panel 1, sympathetic click in Panel 2

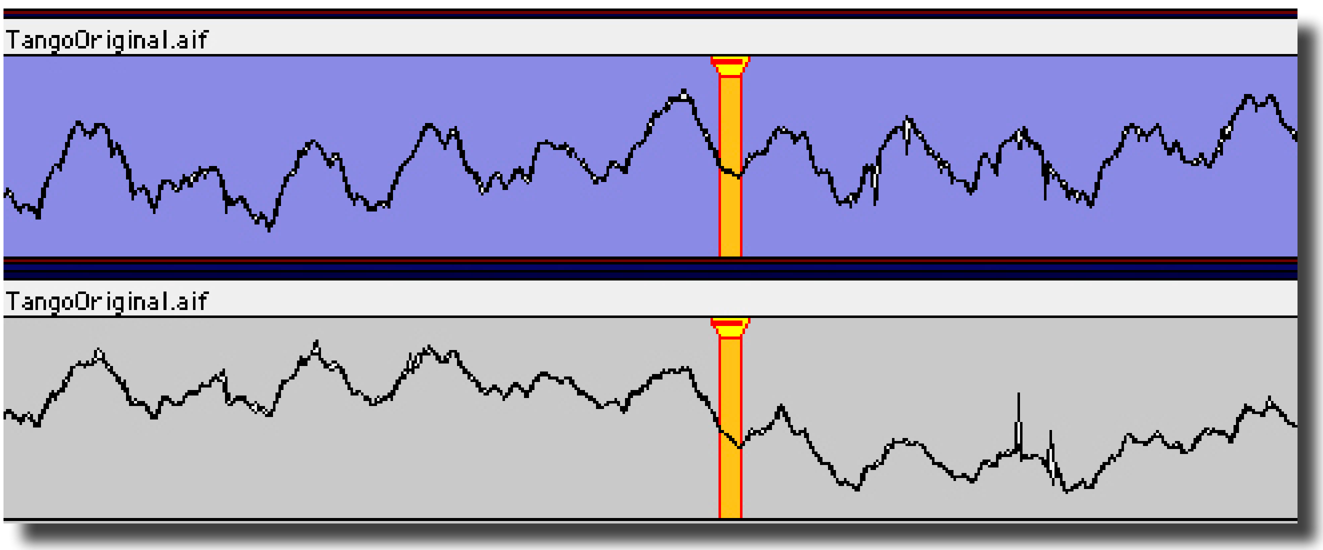


Figure 1.6: The repair using DType

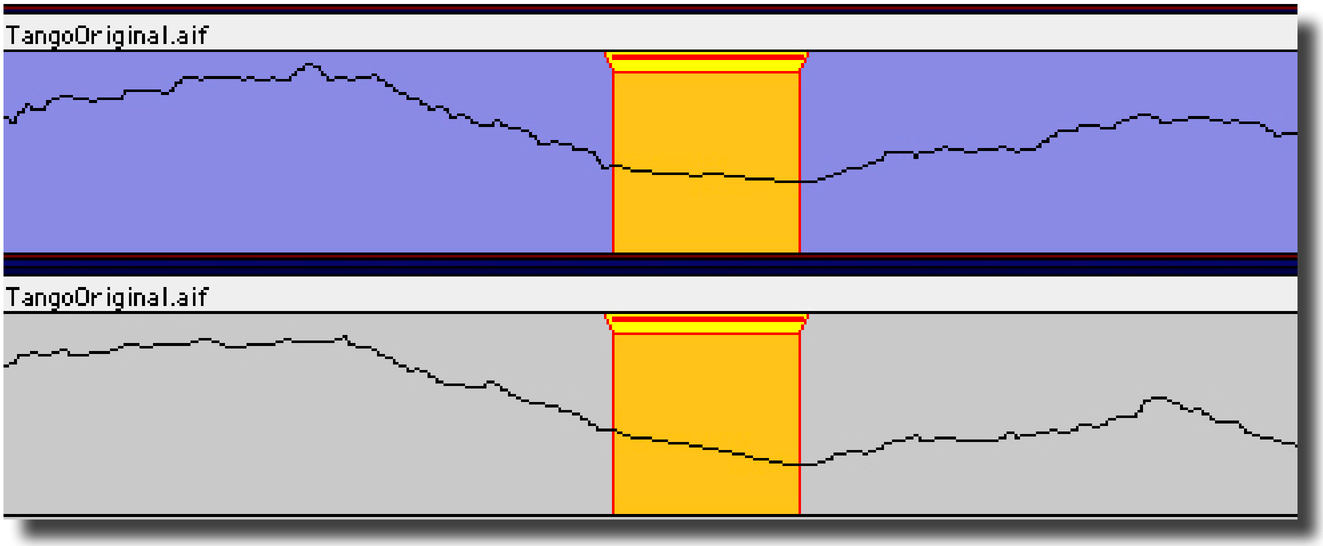


Figure 1.7: A zoomed in view of the above repair

1.1.1.4 The E Type DeCrackler

The Type E Interpolator, though fundamentally similar to the D Type, is implemented so it can be applied to passages of unlimited duration. As with the D Type, expect to wait a while for your result but, it is worth it. The E Type DeCrackler is capable of reducing distortion, including offensive, harsh sounding material. Its micro–repairs leave the audio sounding better without resorting to low pass filtering.

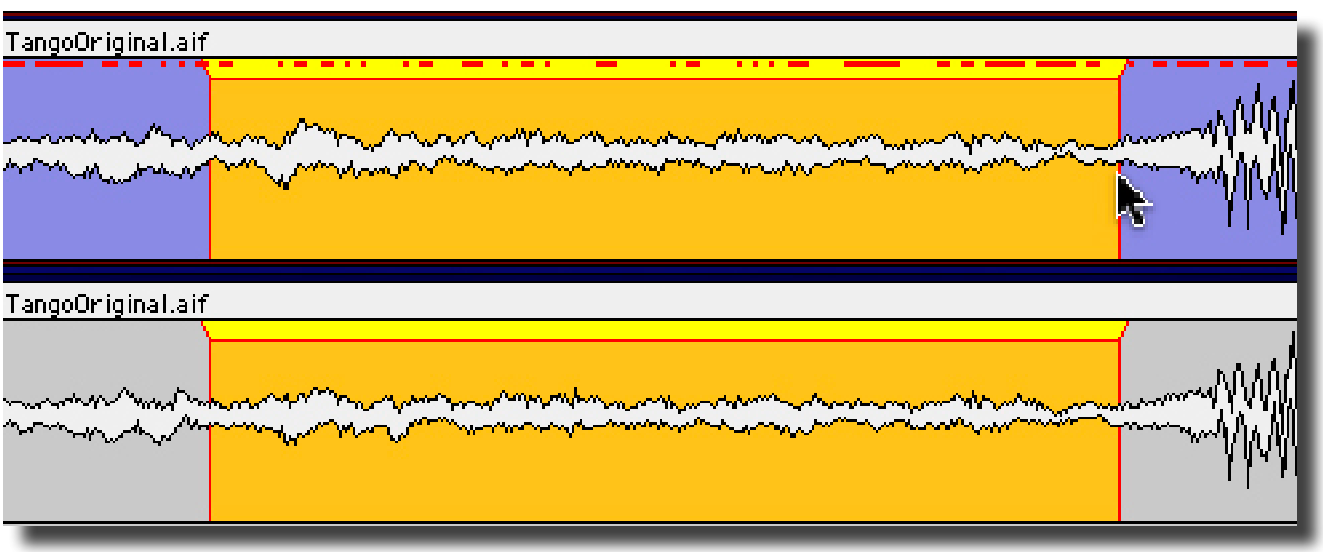


Figure 1.8: E Type's micro–repairs

If you own the DeCrackle native option for NoNOISE II, then an additional E Type command appears under the default entry in the NoNOISE II menu. The second E - DeCracker… command, with an ellipses in the command name, is an adjustable Type E interpolator, with a "Clip Fraction" or threshold that ranges from 0.95 to 0.99. The 0.95 setting is highest sensitivity, with the heaviest amount of repair, while the 0.99 setting is least sensitive, with the lightest amount of repair.

Note that this command also forces or presets the threshold of the default E Type command for subsequent repairs. The current threshold value is shown, in parentheses, next to the command's name as part of the menu entry.

1.1.2 Using Manual DeClick

Manual DeClick performs stereo repairs. Either channel of a stereo pair can be operated on, the repair will be performed on both.

1.1.2.1 Removing Clicks

1. Using the Waveform display and playback, identify the location of an impairment.
2. Zoom in until you can clearly see the impairment.
3. Click–drag in the Panel to create a time region selection that fully contains the damaged samples.
4. From the NoNOISE II menu, select one of the Manual DeClick types.

NoNOISE replaces the compromised audio with repaired samples. Don't worry about selecting on zero crossing boundaries, the software's intelligence will provide a seamless transition.

1.1.2.2 Undoing — Restoring Repaired Clicks

1. You can undo a Manual DeClick repair at any time, restoring the original samples back into the sound file. To do this, zoom in on the waveform and locate the red "Restore Bar."
2. Click–drag in the Panel to select a region that contains the Restore Bar.
3. From the NoNOISE II menu, choose Restore Click.

NoNOISE replaces the interpolated audio with the original audio samples containing the anomaly.

1.1.2.3 Hiding Interpolation Bars

The Show Interpolations command in the NoNOISE II menu is a toggle. When enabled, it displays the interpolation bars that mark the site of a repair. When disabled, the bars are not visible, which reduces visual clitter.

1.1.3 Obscenity Reduction

In addition to restoration duties, Manual DeClicking can be used to insure the public acceptability of obscene material or to conceal any audio that may not "pass muster" with downstream listeners. Simply select the obscenity as though it were an impairment, and choose your Type. The B Type, when given a one second region to "repair," does a great job of removing the objectionable material and inserting something that will often be preferable to editing in replacement audio. An additional consideration is that, since no material is added or removed, the timing or tempo is not affected.

1.2 Broadband DeNoise

1.2.1 Introduction

Broadband noise, whether white, pink or brown in spectrum, is one of the most common forms of audio degradation. Noise can be introduced from any of a number of sources, including the modulation and asperity noise inherent in analog tape recording and Johnson or thermal noise from microphones, preamps, and other analog signal processing equipment. Broadband DeNoise is a single ended broadband noise reducer that, unlike less refined examples, can suppress or eliminate broadband noise with little or no audible artifacts, even at extreme settings.

To suppress such noises in your program, it is necessary to analyze the noise spectrum and adapt the denoising algorithm to the characteristics of the material. Broadband DeNoise operates by means of analysis and resynthesis, though it can be thought of as 2048 bandpass filters, each followed by a below–threshold expander. This is somewhat analogous to the classic analog Dolby and dbx multi–band, double–ended noise reduction systems of yore.

An FFT or Fast Fourier Transform analysis is performed on a user–specified sample of noise from the material to be processed. The average amplitude of noise in each of 2048 individual frequency bands or "bins" is determined. The result of this analysis is a "Noise Estimate," which is used to set the average threshold of the noise reduction. The Estimate sets aggregate threshold for the expansion.

During processing, the source material is also subjected to a 2048 point FFT analysis. The average amplitude of signal in each bin is compared against the threshold value determined by the noise Estimate. Based on this comparison, the algorithm determines whether a given band at that particular instant contains audio signal or only noise.

If a frequency bin is found to be at or above threshold, its gain remains at unity. If it is determined that the signal amplitude in that band falls below threshold, then it is considered "noise" and the amplitude of that band is reduced by an amount determined by the Attenuation parameter, discussed later. The results of this comparison and amplitude compensation for all bands is a modified version of the original FFT frequency analysis. A reverse FFT is then performed using the new, adjusted version, reconstituting the audio signal with aggregate noise attenuated by the specified amount. Because the Broadband DeNoiser operates with high frequency resolution and at extended precision, the removal of noise is precise and artifact–free.

1.2.2 The Noise Estimate

The first step in denoising is to derive a Noise Estimate from the material to be processed. The Noise Estimate, or simply Estimate, is an individual "fingerprint" of the noise and determines local threshold values for each frequency bin. The NoNOISE II menu provides a group of commands for creating, processing, editing and storing the Estimate. The Estimate determines the result of the entire denoising process, so it is important to ensure that the estimate taken is valid and represents the true noise floor of the source sound file.

1.2.2.1 Noise Estimates

The procedure for taking a usable Estimate has several steps. First, open the source sound file into a Project. Then, identify a short section of audio where there is only noise or predominantly noise. About 0.3 to 0.5 seconds is sufficient. Click–drag to create a time region selection of the noise. Section 1.2.2.2 below discusses where to take a Estimate in detail.

From the NoNOISE II menu, select the Take and Interpolate Estimate command to first create the Estimate, then automatically interpolate the Estimate, applying “Bin Controls,” which are individual threshold controls for separate regions of the frequency spectrum.

The separate Take Noise Estimate and Interpolate Noise Estimate commands are for experimenting when creating an optimal Estimate and can usually be skipped in favor of the combined NoNOISE II > Take and Interpolate Estimate command. See section 1.2.2.3 below for a discussion of these individual commands and their function.

The Auto Noise Estimate... command combines the Take and Interpolate Estimate and Write Estimate commands for those jobs where you are in a hurry or cost takes precedence over quality. It is not recommended to use this shortcut if you desire the best result as it does not include the interim step of adjusting Bin Controls.

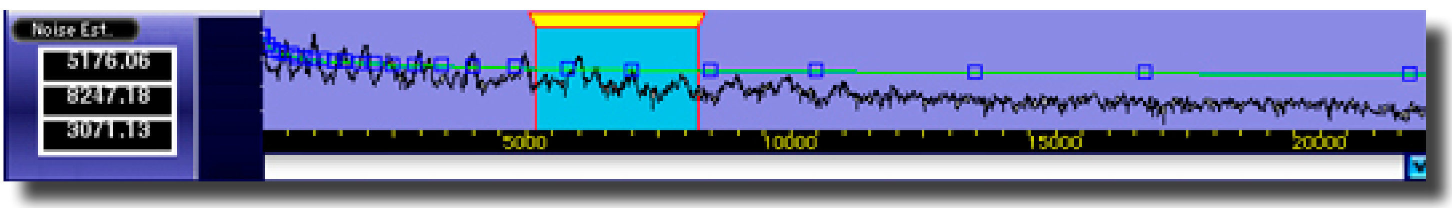


Figure 1.9: An interpolated Estimate with Bin Controls

At this point, you now have a green “Interpolation,” with blue Bin Controls attached. Select the Write Estimate... command to save the Estimate as a file that will be used by the Broadband DeNoise processor.

1.2.2.2 Where to take an Estimate

Once the source sound file is opened into a Panel of a Project, the first step is to identify a suitable location from which to take the Estimate. Since the denoising algorithm depends on a constant amplitude and spectrum in the noise floor, try to listen and locate a time region with uniform noise. As to duration, optimum results are obtained when the Estimate is taken from a section of pure noise between about 0.3 and 0.5 seconds in length, with a worst case minimum of 100 milliseconds.

If a region of pure or “clean” noise, noise uncontaminated with program, is unavailable as is often the case with tightly edited material, then choose a region with minimal program. The resulting Estimate will require manual adjustment as discussed below. If an Estimate must be taken in presence of signal, it is advisable to avoid sections of spectrally complex or non–harmonic material as it makes manual adjustment more time consuming.

It is usually necessary to derive a separate Estimate for each cut or take. If these are contained in a consolidated sound file and the spectrum of the noise varies for each cut, then the consolidated file should be denoised in sections, so that the optimal combination of estimate and parameters can be applied to each cut.

Unless there is strong reason to believe that each cut in a compilation or consolidated file was:

1. recorded in the same session with the same equipment
2. at precisely the same levels onto the same media
3. stored in the same way
4. transferred to the same intermediate media in precisely the same way
5. and converted to digital samples in the same signal chain

then it is advisable to take separate estimates for each cut or take. When denoising stereo material, it is also recommended to derive a separate Estimate for each channel. Likewise, if the character or level of the noise floor can be heard to change at all during the recording, then the best results are be obtained by dividing that piece into sections to be denoised individually. After denoising, the individual sections can edited together and sequenced to create a seamless whole.

1.2.2.3 Post–processing an Estimate

After taking an estimate, the Panel changes to show the Estimate, a smoothed FFT. Before the Estimate can be used to provide threshold data for denoising, it must be additionally smoothed and, perhaps, manually adjusted to account for the presence of desired signal within the noise.

After deriving the Estimate, select the Interpolate Estimate command. soundBlade performs a best fit approximation from the composite FFT data, displaying it as a smooth green spline connecting a series of blue boxes, the Bin Controls mentioned above, superimposed on the composite FFT. You can toggle the Interpolation display by clicking on the checkbox located in the lower right corner of the Estimate display or with the Show/Hide Interpolation command.

If the Estimate is derived from a clean sample of noise, then Interpolate Estimate command produces an accurate measurement of the representative noise floor. In cases where a clean, program–free sample of the noise floor is not available, a smooth Interpolation can be derived from just a portion of the FFT display, provided that the noise in question is purely analog tape hiss, by using the Fit Estimate From Selection command.

Analog tape hiss has a characteristic noise curve or shape. Fit Estimate From Selection takes advantage of this property to produce a smooth Interpolation from any individual region of the Estimate based on that region's average amplitude. Make a time region selection of a portion of the Estimate that appears to be outside the range of the contaminating audio program. Choose Fit Estimate From Selection from the NoNOISE ^{II} menu. soundBlade displays an Interpolation with a characteristic tape hiss curve using the average amplitude of the Estimate's selected region.

1.2.2.4 Editing an Estimate

The Fit Estimate From Selection technique works well for recordings containing typical tape hiss, but many recordings have other noise sources or peculiarities. In such cases, it is often necessary to manually adjust the Interpolation, differentiating between the desired program content and the noise floor.

As with a waveform display in a Panel, the same Zoom In/Out and Zoom To Selection commands work for an Estimate. This means you can zoom to a view that provides a comfortable level of detail.

To edit an Interpolation, hold down the option key and drag a Bin Control up or down. If, for example, a number of piano harmonics are contaminating the Estimate, the Bin Controls can be moved down so that the Interpolation follows the average amplitude rather than the local amplitude. Moving a Bin Control down will command a lower threshold and cause less sensitivity in that Bin, resulting in less noise reduction while moving a Bin Control up sets a higher threshold, causing more noise reduction.

You can move multiple Bin Controls at once. To do so, click–drag in the Panel to create a time region selection that contains the Bin Controls. The numeric display at the left side of the Panel shows the selection's start frequency, end frequency and span in Hertz. To edit the boxes within the selection, hold both the shift and option keys and drag one of the included Bin Controls. All Bin Controls within the selection will move together.

1.2.2.5 Saving an Estimate

Once you are satisfied with the Interpolation, you need to save it. To do so, select the Write Estimate... command from the NoNOISE II menu. soundBlade displays a Save File dialog. Choose a location for the Estimate file, rename it if you wish, then click OK. soundBlade saves the file, including the Interpolation and Bin Control state, to disk.

1.2.3 Running the Broadband DeNoiser

1.2.3.1 Overview

Once you have the Estimate file written, you can start the real–time process. From the NoNOISE II menu, select the BBDN… command, which brings up the BBDN window.

The window, with its six controls discussed below, also has buttons to open an Estimate file, to open and save the controls and their parameters, to bypass the process, and to toggle between the processed signal and the suppressed or removed portion of the signal.

Figure 1.10: The Broadband DeNoise window

Note that BroadBand DeNoise is also available for use as a plug–in. This plug in can be used as a Desk Event in an EDL, as a plug in in the Mixer window or in the Meter window. This plug in is accessed from the contextual menu for plug ins.

1.2.3.2 Button Functions

The Open Est button spawns a standard Mac browser that allows you to specify and open an Estimate file. This will set the threshold and sensitivity of each Bin.

The Save Param buttons allow you save all the current settings in the window, including the name and location of the currently selected Estimate file. The Open Param button allows you to open an existing Parameters file. These files are ASCII text and can be opened in TextEdit for inspection.

The last two buttons are the Bypass, which patches the process in or out of the signal flow, and the NoNOISE/Noise button, which allows you to hear either the processed signal or the portion of the signal that is being suppressed or reduced. When in the "Noise" mode, you can adjust the controls and hear the direct result of your actions.

The operation of the Broadband DeNoiser is a subjective process. In general though, if too much program is present in the suppressed signal, then you are probably using setting that are too aggressive or your Estimate requires adjustment. Also, it's fairly easy to misinterpret noise as high frequency content so, critical listening with wideband reproducers are required for proper operation.

1.2.4 Broadband DeNoise Parameters

The following parameters are available as horizontal sliders:

Parameter	Range	Default Value	Recommended	Extreme
Threshold	-60 to 60	0	n/a	n/a
Attenuation	-60 to 0	0	-10 to -24	
Sharpness	0.5 to 2.5	1.25	0.80 to 1.25	> 1.50
Bandwidth	0.5 to 3	1.50	1.8 to 2.2	< 1.0
High Cutoff	0 to 22050	22050	22050	n/a
Low Cutoff	0 to 22050	500	50 to 2000	25

1.2.4.1 Threshold

The Estimate defines the aggregate threshold applied to each of the over 2,000 individual Bins used by the denoise process. The Threshold parameter allows the entire curve, as a whole, to be moved up or down. Together with the Attenuation parameter, the Threshold parameter provides the basic control over how aggressively the process is applied.

As Threshold is raised, more of the signal at all frequencies is processed. At extremely high settings, a distinctive watery aliasing may be heard in the resulting audio. If the Threshold is set too low, little or no noise reduction is obtained. The Threshold can be thought of as the fine line between noise and music, globally raising or lowering the entire Estimate curve relative to its original position. Local adjustment of the threshold according to frequency bands is effected by adjusting the Interpolation curve and saving a new Estimate file.

The default Threshold is arbitrary. Threshold and Attenuation settings should generally be adjusted together for best results. The Noise/NoNOISE button can help to determine an acceptable compromise setting.

1.2.4.2 Attenuation

This value, in decibels, sets the maximum attenuation to be applied to any Bin. A setting of 0 produces no noise reduction. The higher or more negative this value is set, the greater the reduction in noise, but with increasing danger of producing audible artifacts in the audio signal. Again, Threshold and Attenuation settings should be adjusted together for best results. Critical listening and the Noise/NoNOISE button can help to determine an acceptable compromise setting. If the maximum attenuation setting is too extreme, ambience and/or high frequency content may be lost.

1.2.4.3 Sharpness

As mentioned earlier in the introduction, the denoising process works much like a multiband downward expander. As signal level in a particular band drops, the process reduces the gain in that band even further, using an internal attenuation curve. The Sharpness parameter sets the slope of this curve. Higher values cause quicker attenuation as instantaneous energy falls off from the Threshold value, resulting in a response similar to that of a noise gate.

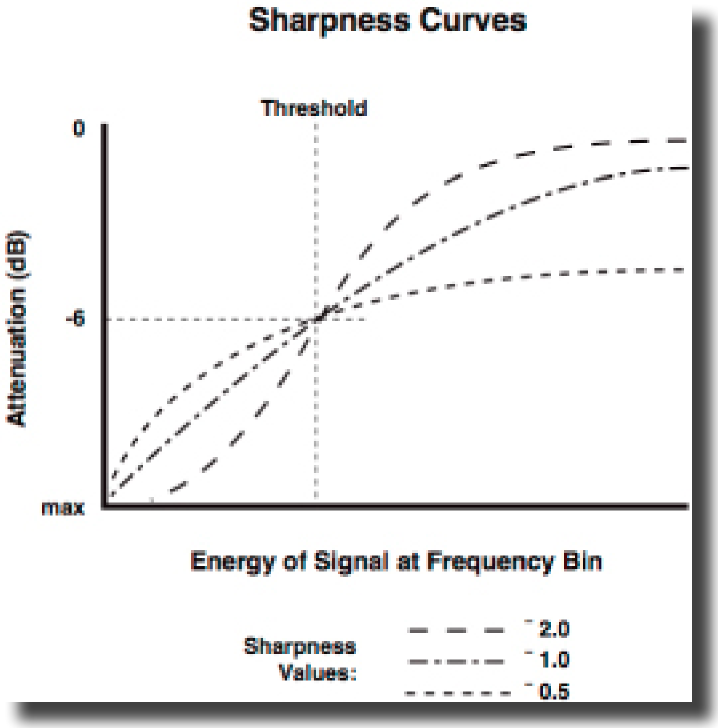


Figure 1.11: Sharpness Curves

If Sharpness is set too low, there may be no reasonable amount of noise reduction possible despite what the other parameters are set to. Generally this parameter should be set as high as possible without audible ill effects on the program. A value of 1.0 is recommended for common tape hiss problems, while a value of about 1.2 has been found useful for standard 78 RPM phonograph recordings. If the sharpness is too high, you may hear the noise floor become less uniform, producing a rapid modulation.

1.2.4.4 Bandwidth

Denoising with each Bin adjusted separately produces an unnatural sounding result. In Broadband DeNoise, individual Bins share information for more pleasing subjective result. Bandwidth governs this process. Higher values produce more intra–Bin sharing of gain changes. Subjectively, a higher value will, in general, create a more natural sounding result but, with a risk of noise floor modulation. A low value for Bandwidth eliminates the possibility of noise pumping, but may sound more muffled. Low values command little sharing while higher values command more sharing. As with other Broadband DeNoise settings, experiment for the best compromise setting. This depends entirely on the program material and should be high enough to retain high frequency response but low enough to avoid pumping or distortion caused by the noise floor being modulated by the high order harmonics of a signal.

1.2.4.5 High Frequency Cutoff

The denoiser lets signal in frequency bins above the high frequency cutoff point pass through the process untouched. That is, they are not processed. This limits processing to frequencies below the high cutoff point. This can be used in situations where noise is not objectionable above a certain frequency, but in most cases, this parameter is left set at 22,050 Hz.

In special cases, better results may be obtained by processing your program in two passes, with a low frequency pass and a high frequency pass run separately, using different parameter values. Run your program through Broadband DeNoise twice, using the low and high cutoff frequencies to define the two bandpass ranges to be noise reduced.

1.2.4.6 Low Frequency Cutoff

This parameter, the complement to the High Frequency Cutoff parameter discussed in section 1.2.4.5 above, allows signal below the specified frequency to pass through unchanged. This can be useful if noise is not objectionable below a certain frequency and you wish to leave it unprocessed. Low Frequency Cutoff is usually left set around 50-100 Hz though setting it at 2.5 kHz or above will ensure that the critical midband will remain untouched. If this parameter is lower than 25 Hz, there may be artifacts because program wavelengths will exceed the analysis "window."

1.3 reNOVAtor

The ReNOVAtor™ option for soundBlade allows localization, identification and very precise removal of unwanted audio events without affecting the audio material you want to keep. The removed sound is replaced by a signal re-synthesized from the surrounding material. reNOVAtor does not make deep gaps in your sound track when eradicating a disturbing sound event. Rather, it's an exactly tailored hole in the spectral representation of the processed signal that can be removed and replaced. The interpolation may even be restricted to certain gain ranges within the selected area, which is very useful if only a certain part of the signal needs to be treated (e.g. one specific harmonic). The reNOVAtor window is fully resizable for increased accuracy and optimal compatibility with all screen resolutions.

Working with reNOVAtor is easy and intuitive. reNOVAtor loads the requested part of audio material you've chosen and analyzes it. The result is displayed as a 3D spectrogram with time on the horizontal axis, frequency on the vertical axis and amplitude of the spectral components color-coded. The color assignment follows the order of the rainbow: red and yellow for low energy; green and blue for middle energy; and finally purple and white for high energy. After getting some experience, this 3D spectrogram representation allows a good feeling for localization and identification of sudden unwanted acoustical events. The spectral area of interest can be precisely marked with a resizable rectangular window. A Play button allows you to hear selected parts of the processed signal.

1.3.1 Main Features

- up to 192 kHz sampling rate

- extraordinary results compared to any other cleaning method, due to selective treatment of spectral representation of the signal and not its waveform.
- enormous time savings when repairing critical live recordings
- easy-to-learn identification and localization of unwanted audio events
- efficient removal of unwanted audio events and their replacement by signals
- re-synthesized from the surrounding audio material
- resizable and zoomable spectrogram window for sound repairing with surgical precision
- multiple selections of harmonics and automatic identification of tones and clicks
- audition of any selected area before and after processing
- multiple undo functions
- gain selective signal treatment
- different types of interpolations
- replacing one spectral region by another (copy & paste)
- no audible changes in desired signal and ambience after removing typical discrete audio disturbances

1.3.2 Typical Applications

- Removing unwanted noises like sneezing, chair squeaks, coughing, car horns, fallen coins and keys, ringing of a mobile phone, etc.
- Correcting instrumental tracks by removing scratches from stringed instruments, wrong notes, rustle of sheet music, keyboard pedal noise, vocalist's breathing, lip smacks and microphone pops
- Restoring old recordings by removing scratches and dropouts
- Cleaning up environmental noise on location recordings for film and television

1.3.3 Getting Started

Start reNOVAtor by click–dragging a time region selection in a Project's Panel. Then, select NoNOISE II > reNOVAtor… to start the processor. After you have selected a wav file to be processed, the reNOVAtor window pops up and shows the spectrogram of the left channel (left).

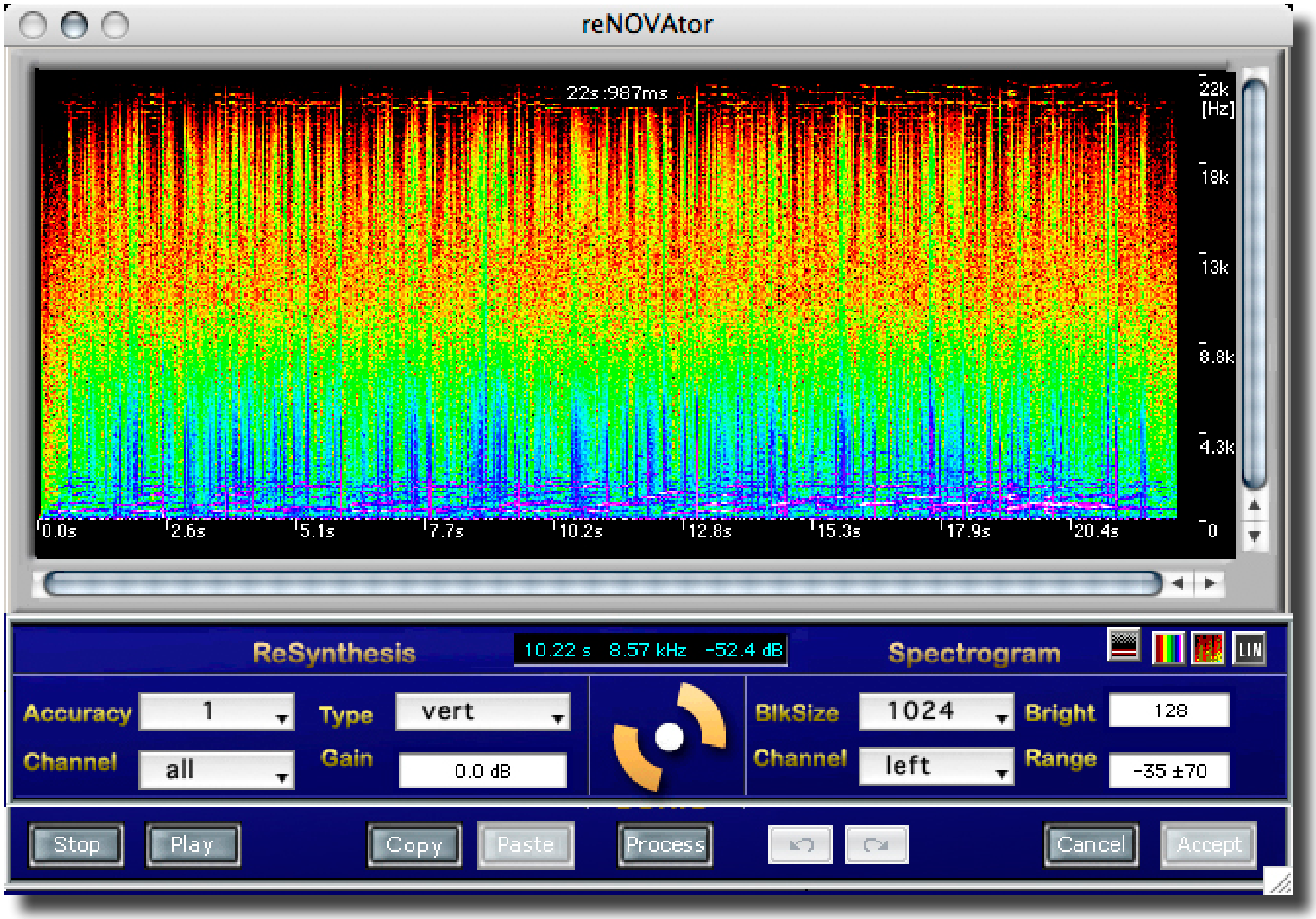


Figure 1.12: The main reNOVAtor window

Typically, files up to 10 minutes can be loaded all at once. The maximum length of audio files is dependent on your host's computational power and RAM complement, the required spectrogram resolution and sample rate for the source sound file. If you experience any performance problems with long files, cancel out of reNOVAtor and shorten the time region selection.

Though reNOVAtor processing is undo–able at any time, we recommend you always make a backup of your original audio material to have the opportunity to start again if, for any reason, the NoNOISE metadata, the ".cd" and ".rl" files, are lost.

Before you begin, remember that "tool tips" are available at any time to remind you of the function of a particular control. To see a tip for any control, simply locate your cursor over a control and hover there for a moment. A contextually appropriate tool tip will then pop up with a brief description of that control or feature.

To adjust the display color range to the material, click–drag on the Range field and move the mouse. Mouse movement along the y-axis, up and down, adjusts the middle position of the gain while movement along the x-axis, left and right, adjusts the overall gain range. As with all numeric fields in reNOVAtor, a double click in the field resets the gain parameter to the default value. In this case, -35 ± 35 dB.

To the right of the Spectrogram label, there are four buttons. The second button toggles between three color mapping schemes:

- "physical," with red representing lowest energy, through yellow, green and blue to white, representing the highest energy
- "standardized," with blue representing the lowest energy, through green and red to white, representing the highest energy
- monochrome, with black representing the lowest energy, through to white, representing the highest energy
- inverse monochrome, with white representing the lowest energy, through to black, representing the highest energy

To zoom in on an area of interest, first select the portion of the spectrogram by holding the left mouse button and drawing a selection or marquee. Then right-click to open the popup menu, as shown below. Finally choose one of the zooming operations (see Quick Reference for details).

The selected area can be moved or resized as indicated by the mouse cursor. Its length in seconds and milliseconds is displayed inside. The shading grade of the area can be controlled by the Bright parameter in the Spectrogram Parameter Group from darkening (as shown above) to brightening. A double left-click on the Bright resets it to the neutral position (192).

Another way to select the visible spectrogram portion is by resizing and moving the scrollbars beside the display. By left-clicking and moving the edges of the horizontal or vertical scrollbar, the visible spectrogram region can be zoomed in or out and moved. When the SHIFT-key is also held down, zooming is performed symmetrically. The length of the displayed signal is indicated in the upper part of the spectrogram.

The displayed audio channel can be toggled by left-clicking on the channel field (left, right, l+r). Block size applied to the spectrogram and interpolation is accessed by left-clicking on the Blocksize field, holding the button and selecting the value from the pop-up list (selectable values: 64, 128, 256, 512, 1024, 2048, 4096, 8192, 16384 and 32768). In general, interpolation of short disturbances (like clicks) requires smaller block sizes, while a frequency selective interpolation (like removal of discrete tone) requires larger block sizes.

The whole loaded audio material or any portion can be replayed at any time by positioning the white play cursor at the desired position and clicking the Play button. To place the play cursor, left-click at the bottom (black) end of the spectrogram where the mouse cursor changes to the play cursor , or --while holding ALT-key-- just click on any position in the spectrogram. A left-click on the Stop button stops the audio playback. After stopping, the play cursor returns to its initial position.

To remove an unwanted disturbance, select an area around it and set up the desired parameters in the ReSynthesis Parameter Group. These are Accuracy, Channel, Type and Gain (see in Quick Reference for all setup possibilities). To remove clicks as shown in the example above, you will usually use the hor interpolation type to replace the selected area containing respective click with a new signal re-synthesized from the material surrounding the click along the time lime.

The interpolation types left or right can be used, if one side of the click is not suitable for proper interpolation (e.g., it includes strong percussive beats). However, the results in case of one-side interpolation are less accurate.

To perform the interpolation in a selected area, hit the Process button or select Process from the drop-down menu. After processing has been done, you can immediately listen to the result by resetting the play cursor and hitting Play button. The lower screen shot from the previous page shows the audio piece from the upper screen shot after click removal.

If you are not satisfied with the result, you may undo the interpolation with the button or by pressing ctrl-z. The maximal number of undo steps is limited only by the available computer memory. If you reach the memory limit you can clear the undo buffer by hitting the button. Once an undo step is performed, it can still be re-done by hitting the button or ctrl-y on the keyboard. However, after the reNOVAtor has been closed with Accept or Cancel, all pending re-do steps are no longer accessible.

To finish your reNOVAtor session, hit the Accept button to accept the changes and write them back to the track or hit cancel to exit the process and discard all changes.

For precise spectrum analysis the numerical display in the middle below the spectrogram is provided. It shows spectrum properties at the current cursor position: time, frequency and amplitude.

For precise spectrum analysis the numerical display in the middle below the spectrogram is provided. It shows spectrum properties at the current cursor position: time, frequency and amplitude.

Note that as long as you stay within a reNOVAtor session all processing steps are stored in the temporary memory and can be re-done. However, after closing the session with the Accept or Cancel button, all intermediate steps get lost.

All parameters, their settings, and remaining buttons are precisely described later in the Quick Reference.

1.3.4 Quick Reference

1.3.4.1 Spectrogram Setup

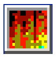
To adjust the spectrogram color range of the material, click–hold in the Range field and move the mouse. Movement along the y-axis, up and down, adjusts the middle position of the gain while the movement along the x-axis, left and right, adjusts the gain range.

The  button toggles between four color mapping schemes:

1. physical with red representing lowest energy through yellow, green and blue to white representing the highest energy
2. "standardized" with blue representing the lowest energy through green and red to white representing the highest energy

3. grey scale with black representing the lowest energy through to white representing the highest energy

4. inverse grey scale with white representing the lowest energy through to black representing the highest energy

The  button activates smoothing of the spectrogram. The 

button switches between linear and logarithmic frequency axis.

1.3.4.2 Spectrogram Resizing

In general, the entire reNOVAtor window is resizable. The spectrogram itself can be zoomed in or out. It can also be moved independently in both directions, horizontally and vertically, by resizing and moving the associated scroll bars. With the shift key held down, zooming is symmetrical. The length of the program portion actually displayed is indicated in the upper part of the spectrogram.

When clicking in the scroll bar area, but outside the bar, the zoom area moves 1/3 display toward the direction clicked. This can be useful for tracking the play cursor during playback.

1.3.4.3 Marking an Area

An area of interest can be click–dragging on the spectrogram and drawing a marquee or selection around it. The selected area can be moved or resized as indicated by the mouse cursor. Its length in seconds and milliseconds is displayed inside. The contrasting shading of the area can be controlled by the Bright parameter in the Spectrogram Parameters.

To move a marquee to another location, move the cursor to the center of the selection. The cursor changes to the Move cursor, a four quadrant arrow shape.

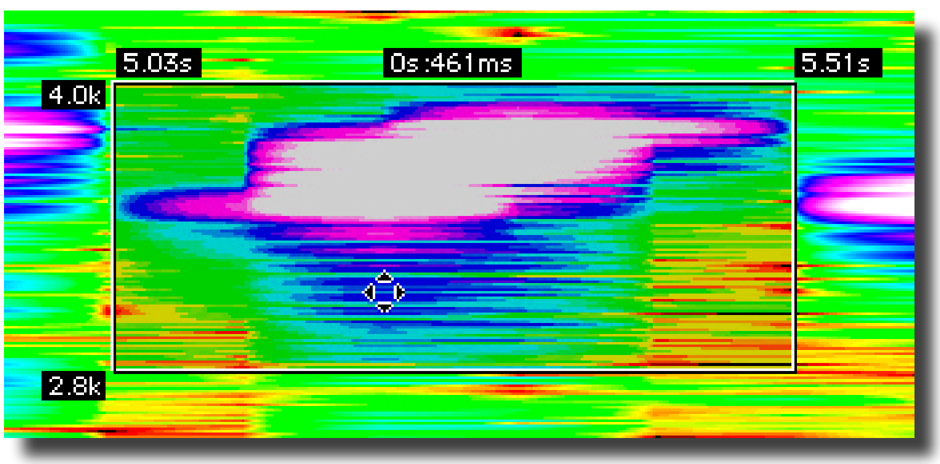


Figure 1.13: The Move cursor

Once you see the Move cursor, you can click–drag the marquee around the spectrogram. By holding down the shift key before dragging, you can constrain movement to only the horizontal or time axis.

1.3.4.4 Zooming a Selected Area

Zooming in a selected area can be performed in one of two ways. First, from the pop-up menu that opens after control–clicking in the spectrogram or by using shortcuts, see section 3.3.10 below. The second way is to place your curson in the black, horizontal time scale or the vertical, frequency scale. The cursor changes to a magnifying glass, allowing you to zoom in on that axis. Holding down the shift key allows you to zoom out.

Note that resizing the spectrogram with scroll bars respectively changes the size of the selected area in the spectrogram, but its absolute length, in seconds and milliseconds, remains unchanged.

A option–clicking inside the spectrum display enters the zoom/move mode. Moving the mouse up and down zooms out and in. Move left or right to scroll. With this feature, it is possible to click into an area of interest and move the mouse down in order to zoom the spectrogram around the mouse cursor.

Additional zooming possibilities are provided by key equivalent. See section 3.3.10 below for more information

1.3.4.5 Playback

The whole loaded audio material or part of it can be played back at any time by clicking in the spectrogram to position the white play cursor at the desired position. Clicking the Play button or tapping the space bar start playback. After playback has stopped, the play cursor returns to its initial start position.

Additional playback possibilities are provided by key equivalent. See section 3.3.10 below for more information.

1.3.4.6 Copy & Paste

A selected area can be copied to another spectrogram position. After clicking the Copy button or selecting Copy Area from the contextual menu, it can be moved around with the mouse to the desired destination. As long as the copy mode is active, the field Type shows copy.

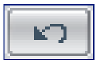
To paste the selected area into the desired location, click the Process button or, select Process from the contextual menu. The marquee changes from white to blue.

To exit the Copy mode, simply click outside the marquee. The ReSynthesis Type will switch back to its previous selection and the marquee changes back to white.

The copy and paste function can be restricted to the original position in frequency if the shift key is held down while relocating the marquee. This is very useful, when the area to be copied contains material, such as harmonics, that needs to be placed in exactly the same frequency region.

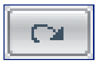
1.3.5 User Interface Details

1.3.5.1 Buttons



- Undoes the interpolation (or ctrl-z).

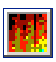
The maximum number of undo steps is limited only by the available computer memory.



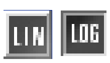
- Redoes the last undo step (or ctrl-y); be aware that after the reNOVAtor has been closed with Ok or Cancel all pending re-do steps are no longer accessible.

- Introduces the Automatic Selection of Harmonics mode, see section 1.3.8.2 below. It allows selecting multiple harmonics belonging to a marked fundamental frequency. The fine tuning of the fundamental frequency and enabling or disabling each individual harmonic selection can be done in the Harmonics Selection window that pop-ups after clicking the harmonic selection icon.

- Toggles between four color mapping schemas: (1) physical with red representing the lowest energy, over yellow, green, blue to white representing the highest energy; (2) “standardized” with blue representing the lowest energy, over green, red to white representing the highest energy and (3) monochrome from black representing the lowest energy, over gray to white representing the highest energy and, (4) an inverse monochrome mode.



- Activates smoothing in the spectrogram; it should normally be on; to switch it off makes sense if we need to precisely see the borders of very sharp spikes.



- Toggles between linear and logarithmic frequency axis.

1.3.5.2 Cursor Callout

if the mouse cursor is located within the spectrogram, the cursor callout indicates the properties of the spectral component at the current cursor position: time, frequency and amplitude.

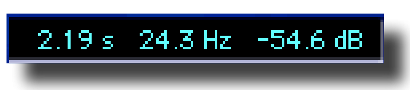


Figure 1.14: The cursor callout

1.3.5.3 Menus

A contextual menu opens after control-clicking on the spectrogram.

Figure 1.15: The spectrogram's contextual menu

- Zoom + – This choice zooms in the selected area to the maximum dimensions, limited only by the display size and spectrogram resolution.
- Zoom – – This choice zooms out from selected areas.
- Zoom Hor – This choice zooms in on the selected area horizontally only.
- Zoom Ver – This choice zooms in on the selected area vertically only.
- Copy Area – This choice copies the contents of the selected area, allowing you to move it to another location in the spectrogram for later processing or pasting. This is equivalent to the Copy button.

  Once copied, the marquee changes from black to blue, and the entire selected areas can now be dragged around to relocate it in the spectrogram. To limit movement of the copied area horizontally, to accurately paste harmonic content for instance, hold down the *shift* key before dragging.
- Process – This choice carries out the interpolation. This is equivalent to the Process button or hitting the P key on the keyboard.

1.3.5.4 Main Buttons

– Starts playback from the current cursor position

– Stops playback and returns the play cursor to initial position

– Carries out the interpolation in the selected area

– Discards all signal modifications and exits the option

– Accepts all signal modifications and modifies the source sound file with a processed region

1.3.6 ReSynthesis Parameters

Accuracy, Channel and Type can be selected from their respective menu that appears after clicking in the field. Gain can be adjusted by click–dragging the mouse up or down in the field.

Accuracy – Accuracy [1, 2, 4, 8] sets up the time resolution of the interpolation process and the spectrogram while maintaining the frequency resolution determined by the Blocksize field. The higher the number, the greater the accuracy in time. This doesn't mean that the highest number is always best, since it depends on your audio material.

Normally, for vertical interpolation, 1 or 2 works optimally. For horizontal interpolation, higher values might be more advantageous. Changing the Accuracy parameter results in recalculation of the whole spectrogram, which can take considerable time and memory. Therefore, make changes only when a smaller piece of audio is loaded, no more then a few minutes.

Channel – Channel [all, current, left, right] selects the channel on which the interpolation will be performed.

Type - Type [hor, vert, left, right, top, bottom, 2-dim, Gain, Copy] selects the kind of interpolation to be applied to your selection. All interpolation types, with the exception of Gain, perform a two step operation. First, they remove the signal from the marked area and subsequently fill the spectral hole with a replacement re-synthesized from the surrounding audio material. The name of the interpolation type explicitly describes the "context,' what part of the surrounding audio material is used for the re-synthesis.

The Horizontal (hor) interpolation type is most suitable for removing transient disturbances like clicks while Vertical (vert) is recommended for removing longer tones or harmonics. The interpolation types Left or Right are a special form of Horizontal, and should be used if one side of the impairment is not suitable as interpolation context. An example would be a strong percussive beat just after an impairment. A left interpolation would use only earlier material as the re–synthesis context, avoiding a flam effect or partial duplication of the later percussive event from inclusion in the resynthesis.

Similarly, the top and bottom are special forms of vertical interpolation and should be used if one "side" adjacent to the impairment includes strong tones or harmonics. However, be aware that the results of one-sided interpolations are not as accurate as in symmetrical Horizontal or Vertical choices.

The 2-dim or Two Dimensional interpolation is a special combination of both horizontal and vertical interpolation and is recommended for removing small "square" shaped disturbances.

The Gain interpolation type, see below, does not remove the original material from the selected area like all the other interpolation types. It simply reduces or amplifies existing spectral components in the selection, according to the Gain parameter in the ReSynthesis Parameters.

After selecting either Copy Area in the conxetual menu by control-clicking on the spectrogram, Copy is enabled in the Type field and persists during the copy procedure. It can not be selected directly from the Type field.

Gain –The Gain field provides two functions. For the Gain interpolation type, it controls the gain applied to the signal components in the selected area and, for all other interpolation types except copy, the value acts as an intelligent threshold. The range is -60 +20 with a default value of 0 dB. As with all numeric fields, a double click on the field resets the value.

The Gain interpolation type does not remove the original material from the selected area like all the other interpolation types. It simply reduces or amplifies original signal components in the selection, according to the Gain setting. In general, the higher the Gain value, the more of the original signal is preserved. This means that a deep interpolation, of any type, can be carried out only with a low Gain setting, such as -40 dB, while Gain or actually threshold above 0 dB preserves some parts inside the selected area which might be useful material. With Gain being too high, however, even the disturbances may partially remain. In such cases, lower the Gain until the expected result is achieved.

1.3.7 Spectrogram Parameters

As with the ReSynthesis Parameters, these parameters are either menus or numeric fields. As with all numeric fields, a double click on the Brightness or Range fields resets the value to the default.

Blocksize – Block Size [32, 64, 128, 256, 512, 1024, 2048, 4096, 8192, 16384, 32768], even though it resides in the Spectrogram parameters section, selects the signal analysis resolution for *both* the spectrogram and interpolation algorithm. In general, interpolation of short disturbances, like clicks, require smaller block sizes, while a frequency selective interpolation, like removal of discrete tones or harmonics, requires larger block sizes.

Block size allows you to locate specific impairments prior to effecting a repair. Because larger block sizes tend to visually isolate and repair areas at low frequencies, it allows you to avoid touching areas that should be left intact.

Channel –The Channel setting [left, right, l+r] shows the spectrogram of left, right, or a sum of both. The sum [l+r] allows the identification of all disturbances regardless of the channel in which they occurred.

Bright – Brightness [0 – 384] controls the contrasting brightness of the selected area from bright (192 – 384) to dark (191 to 0). It can be adjusted by click–dragging up or down on the field. A double click in the field resets the brightness to neutral.

Range – Range adjusts the spectrogram colors to the dynamic range of the audio material, with a middle value of -100 dB to + 40 dB, and a range of ± 10 to ± 70. A double click in the field resets the Range to the default values of -35 dB ± 35 dB.

Range can be adjusted by click–dragging vertically on the field, with up equating to higher sensitivity and down lower sensitivity for controlling the middle value. Click–dragging horizontally on the field controls the mapping range.

Be aware that the Range values are related to spectral components of the processed signal and not to the overall signal level. Therefore, prior to processing, they may look unusual.

1.3.8 Interpolation Alternatives

1.3.8.1 Copy and Paste

While interpolation is the preferred method of removing isolated artifacts from complex audio material, there are situations where this method suffers from adequate material around the artifact that could be used as context for the interpolation. An example of this would be where the artifact itself is surrounded by strong transients. In such cases, the copy and paste method can produce far superior results. With copy and paste, you can simply find another area that looks like it could "fit into the space" you want to interpolate and insert it. reNOVAtor copies an spectrum from one region to another.

To perform the area copy operation, click–drag to select a source area and click the Copy button. The selection can now be moved around with the mouse to the desired destination. As long as the Copy Mode is active, the Type parameter will show Copy. To paste the selected area into the destination location, either click the Process button or select Process from the contextual menu. To leave the copy mode, simply click somewhere outside the selection. The Type parameter switches back to its previous value.

Copy and paste can, in desperate situations, be used to pitch correct material. First, locate and mark the fundamental. Then, copy and paste that region and test the result. If it acceptable, then move up the "chain" of harmonics, pasting them in at the desired locations. Once you have achieved the desired result, localized in time, then you may want to perform a more global horizontal or 4-dim repair to smooth the result.

1.3.8.2 Automatic Selection of Harmonics

To accelerate removal of heavy hum or buzz as well as removal or shift of complete instrumental tones including thier fundamental and harmonics, reNOVAtor provides an automatic selection of harmonics associated with a selected fundamental. Of course, it's possible to remove an instrument tone just by drawing one big selection around the whole region covering both fundamental and harmonics. For longer tones, however, that approach will produce artifacts and reduce the integrity of your ambience. Using selection of harmonics, you keep much more of the spectrum surrounding your harmonics unprocessed. In case of continues and extremely heavy buzz, this is the only method to preserve at least a part of wanted signal. It is extremely useful in forensic applications, especially if the desired signal is so strongly masked by buzz that the signal-to-noise ratio drops even below zero. Also in a case when you need to distinguish two or more overlapping or mixed instruments playing different tones, the only chance to remove or delete individual tones is by using the Automatic Selection of Harmonics mode.

To activate Automatic Selection of Harmonics mode, mark the fundamental tone drawing a horizontal rectangular around it and press the the Automatic Selection of Harmonics button. The Harmonics Selection window pops up, allowing switching on or off each individual harmonic.

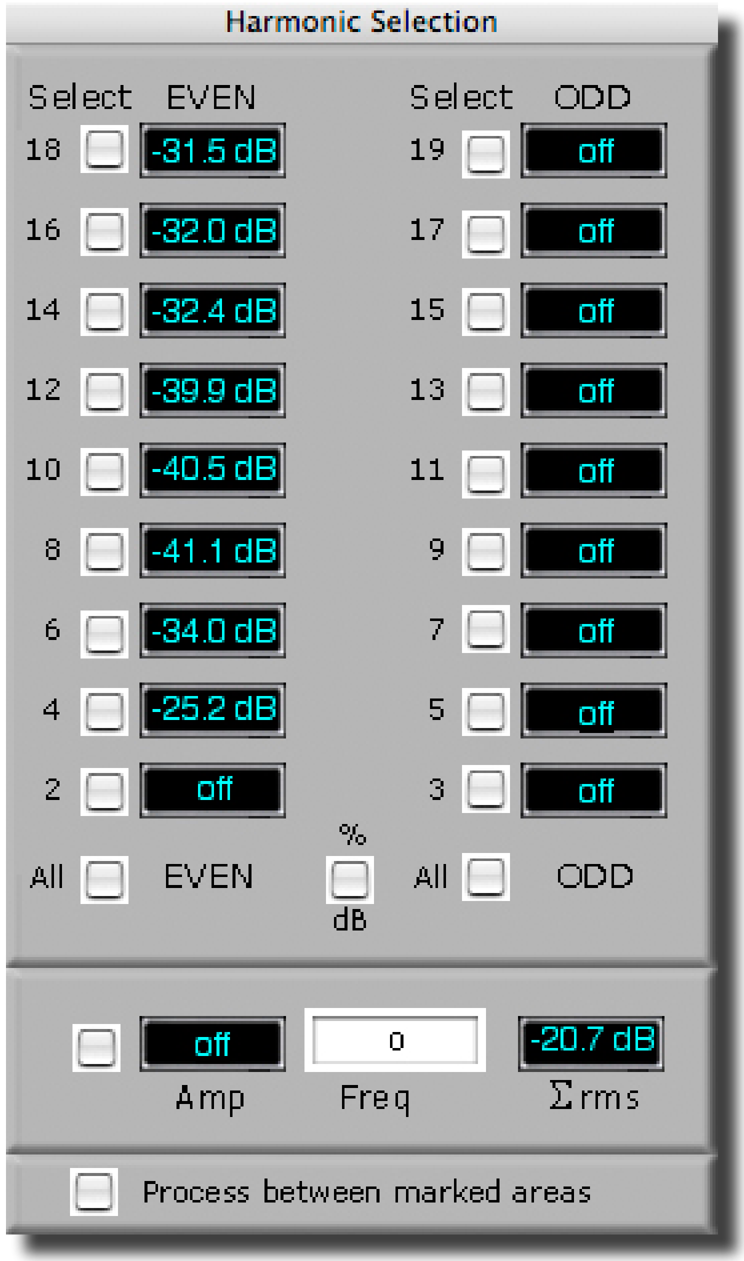


Figure 1.16: The Harmonics Selection window

This selection function automatically draws rectangles around each harmonic enabled in the interface. There are up to 19 individual harmonics and one fundamental which can be considered in the interpolation process. The Harmonics Selection window monitors levels of each individual harmonic belonging to the marked fundamental. The levels of harmonics, related to their energy, can be displayed in dB or in % relative to the level of fundamental. Observing these levels you can get a good feeling about weight of each harmonic. This helps also to estimate the correct position of the rectangles marking the harmonics. If only part of a harmonic is bordered, the displayed level is smaller.

The fundamental frequency originally selected on the spectrogram is numerically displayed in the Freq or Frequency field of the Harmonics Selection window. It can be fine tuned by click–dragging in this field up or down. This is a very important adjustment, because even small changes of the fundamental frequency determine the proper position of the resulting marking rectangles for harmonics. Alternatively, the frequency can also be fine-tuned by holding down the shift key while resizing the rectangle defining the fundamental. Each individual harmonic, and the fundamental, can be enabled or disabled with their respective button. Also, all odd or even harmonics can be simultaneously switched on or off by using the buttons All EVEN or All ODD.

To the left and right of the Freq field, there are two additional fields. Amp or Amplitude shows the amplitude of the fundamental frequency, while Σrms shows the RMS energy of harmonics plus fundamental covered by all active selected areas.

The amplitudes of spectral components are normalized in such way that Σrms shows 0 dB when processing white noise having a maximum line level of 0 dBFs, and the marking rectangles covers the whole frequency range from 0 to maximum.

The "Process between marked areas" check box, at the bottom of the Harmonics Selection window, allows an interpolation which is complementary to the harmonic removal process. In this case harmonics are preserved and the space among them is cleaned. It is especially useful for cleaning sound samples used for keyboards and sequencers.

Note that the Automatic Selection of Harmonics tool can also be used for the Copy & Paste function. This is especially advantageous if you want to shift a tone, a fundamental with accompanying harmonics, while keeping the ambience integrity untouched. It also helps to correct the beginning of a tone if it has been not precisely played.

1.3.9 Application Tips

reNOVAtor is an easy to use tool, even if, at first glance, you may feel uncertain because of the unusual interface. However, after reading the Getting Started chapter, you will be able to carry out repairs with speed and quality you have never experienced before.

This manual gives you just an overview of the reNOVAtor functionality, but there are still a lot of tricks and "secrets," allowing previously impossible sound repair operations. Indeed, it is very difficult to formulate general statements on how to proceed. Every impairment, in connection with specific audio material you want to retain, creates a unique situation. We recommend you spend some time trying to localize and remove certain audible features, or copy audio events to other destinations. Together with all zooming, resizing and precise playback cursor-placing possibilities, you'll get a good feel for controlling reNOVAtor in a short time. Below are some additional remarks which may help to speed your learning process.

reNOVAtor repairs are always undo–able at any time by selecting NoNOISE II > Restore Click. However, before loading a sound file to be processed, we recommend you always make a backup of your original audio material to have the opportunity to start again if, for any reason, the NoNOISE metadata, the ".cd" and ".rl" files, are lost.

For removing transient disturbances like clicks, use the hor (horizontal) interpolation type and for removing longer tones or harmonics, use vert (vertical) interpolation type. The interpolation types left or right are a special form of hor and should be used if one side of the disturbing click is not suitable for proper interpolation such as when the material includes a strong percussive beat. Similarly, the top and bottom are a special form of vertical interpolation and should be used if one side of the impairment includes strong tones or harmonics. However, note that the results of one sided interpolations are not as accurate as symmetrical repairs.

Be aware of the proper selection of the blksize parameter. In general, interpolation of short disturbances, like clicks, requires smaller block sizes, while a frequency selective interpolation, like removal of discrete tones or harmonics, requires larger block sizes. In difficult cases, such as with loud, short duration sounds mixed with low-frequency reverberation, a step–by–step removal process works better. Instead of a wide, vertical area selection, begin with a narrow one to concentrate on the main energy and then a wider horizontal but vertically smaller selection to remove the reverberation part. In addition, for the first step select a smaller block size, such as 256, and for the second step, a larger one like 2048.

If the disturbance is a wideband signal that strongly overlaps with the signal you want to retain, it helps to mark and process carefully selected small regions, step–by–step, instead of drawing one big selection around the critical area.

In extremely difficult situations, especially if you do not have enough clean material surrounding the unwanted noise, we recommend using the copy and paste technique. This method is also ideal for correcting timing problems in instrumental or vocal performance. You can grab a whole tone and shift it a little bit backwards or forwards.

reNOVAtor is also a perfect tool for improving distorted signals. It can remove distortions caused by overdriving analog equipment and works as an excellent de-esser.

If you cannot locate the disturbance you hear, switch the spectrogram to a different Channel if possible. Also, try a different Blocksize and adjust the Range parameter to get proper amplitude-to-color mapping. It may also help to increase the Accuracy, since this will lead to a higher time resolution of the spectrogram while maintaining its frequency resolution, as determined by Blocksize.

1.3.10 reNOVAtor Key Equivalents

General

space bar	play
shift-p	process
command-z	undo
shift-command-z	redo
shift-c	copy selection
shift-v	paste selection
shift-drag	constrain selection in f (frequency)
` (accent grave)	toggles smoothing

Zoom

0	zoom to entire

1	zoom 10x
2	zoom 5x
3	zoom 2x
4	zoom 1x
5	zoom 0.5x
up arrow	zoom 0.5x
down arrow	zoom 2x
<	zoom ƒ by 0.5x
>	zoom ƒ by 2x
, (comma)	zoom ƒ to top half
. (period)	zoom ƒ to bottom half
; (semicolon)	zoom ƒ to middle half
/ (forward solidus or slash)	zoom ƒ to entire

Presets

^ (accent circumflex)	set preset 1
& (ampersand)	set preset 2
* (asterisk)	set preset 3
(	set preset 4
6	load preset 1
7	load preset 2
8	load preset 3
9	load preset 4

Play Cursor Placement

y	set play cursor before selection, 6 sec.
u	set play cursor before selection, 3 sec.
i	set play cursor before selection, 1.5 sec.
o	set play cursor before selection, 0.5 sec.

Accuracy

}	select next accuracy
{	select previous accuracy

Channel

shift-l	select left channel
shift-r	select right channel
c	select current channel
a	select all channels

Type

h	horz
v	vert
l	left
r	right
t	top
b	bottom
d	type 2-dim (2 dimensional)
g	gain

Block Size

=	select next block size
-	select prev block size

Note that, for the PDF version of this Index, only the page numbers, *not the preceding descriptive subject text*, are hyperlinked.

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Comments, corrections and suggestions regarding this manual are always welcome. Please contact us at <support@ sonicstudio.com>.

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