Wave Arts
Master Restoration
Suite 6

User Manual

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1. Introduction

1.1 Master Restoration Suite Overview

Master Restoration Suite is a comprehensive set of restoration plug-ins for cleaning up tape, vinyl, and acoustic recordings. The tools give high-quality results with minimal tweaking, hence they sound great and are easy to use.

The MR Suite consists of 5 plug-ins:

MR Noise – Stellar sounding broadband noise reduction
MR Click – Click and crackle filter for vinyl or digital sources
MR Hum – Precise hum and buzz removal
MR Gate – Expander/gate for quick and simple background attenuation
Master Restoration – The all-in-one cleanup tool

1.2 Installation

Installers for Master Restoration Suite are found on the downloads page of the Wave Arts website. There are separate installers for Mac and Windows. You can download the full suite installer or individual plug-in installers. Mac installers are ".dmg" files which after downloading will expand into a ".pkg" installer file; double-click on the ".pkg" file to launch the installer. Windows installers are ".exe" files; double-click on the ".exe" file to launch the installer. The installers provide various options for selecting which plug-in formats to install and whether to use Pace/iLok or Wave Arts licensing.

1.3 Registration

We support two licensing methods – Wave Arts licensing and PACE/iLok. When installing the plug-ins you must select which version of the plug-ins you wish to use. When you purchase a plug-in, you will be e-mailed a serial number (looks like WA-PPP-XXXX-XXXX where PPP is a product code and X is a hex digit). Use the serial number to unlock the plug-in as described below.

1.4 Wave Arts licensing

Prior to registration, the plug-ins operate in demonstration mode; they are fully functional but stop operating after 30 days. To unlock the plug-in after purchasing, open the plug-in, select the Tools->Register option, and enter your name, email address, and serial number. The plug-in will contact our registration server and download a license file which will unlock the plug-in. You should see a message saying your registration was successful.

If your computer is not connected to the internet, use the Tools->Offline Register option and follow the instructions to generate a keyfile at our website registration page and import the keyfile.

If you have purchased a plug-in suite, when you unlock any one of the plug-ins within the suite, the entire suite will be unlocked.

1.5 PACE/iLok licensing

All our plug-ins support PACE/iLok. Prior to activation, the plug-ins will allow you to start a 30-day trial by creating an iLok account. To unlock the plug-in after purchasing, go to our Product Registration page, select "PACE/iLok", and enter your serial number. A PACE redeem code (looks like XXXX-XXXX-XXXX-XXXX-XXXX-XXXX-XXXX-XX) will be displayed and also emailed to you. There are two ways to redeem the code and generate a license. When opening the plug-in a dialog window will appear and give you the option to Activate the plug-in, you can paste the PACE redeem code there, and proceed to create or login to an iLok account and then transfer the license to an iLok or your machine. Otherwise, go to http://www.ilok.com, create an iLok account, and download and install the iLok License Manager. Within the manager, under the Licenses menu, select "Redeem Activation Code" and paste your redeem code. Then transfer the license to either an iLok dongle or your machine. The plug-in will run only if it can find a license on an iLok or the machine.

When purchasing a plug-in suite, the redeem code will generate multiple licenses, one per plug-in in the suite, but the licenses are grouped together.

2. Plug-in Control Operation

2.1 Knobs

Please refer to the following guide for information about the various ways you can use knobs:

Function	Mac	Windows
Increase/Decrease a parameter value (rotate clockwise/counterclockwise)	Click on the knob + drag up/down	Click on the knob + drag up/down
Fine adjustment — increase/decrease	Shift + click + drag up/down -or- Command + click + drag up/down	Right click + drag up/down -or- Shift + click + drag up/down -or- Control + click + drag up/down
Reset knob to default value	Double-click	Double-click

2.2 Text Entry

Many value displays are editable text. A text field is editable if your mouse cursor changes to an I-beam when moved over the text. Following is a table that fully describes how to use the text editing features:

Function	Mac	Windows
Enter text entry mode	Click in the display	Click in the display
Select text	Click + drag	Click + drag
Select entire text	Double-click	Double-click
Delete character to left of cursor	Delete	Backspace
Delete character to right of cursor	Fn+Delete	Delete

Move the cursor left/right	Left/Right arrow keys	Left/Right arrow keys
Extend the current selection	Shift + click + drag -or- Shift + left/right arrow keys	Shift + click + drag -or- Shift + left/right arrow keys
Exit text entry mode	ESC -or- Return/Enter	ESC -or- Return/Enter
Select next parameter to edit	Tab	Tab
Select previous parameter to edit	Shift + Tab	Shift + Tab

You'll find that many parameters, such as frequency, will recognize units typed into the text field. The following values, when typed into a frequency value box, are equivalent:

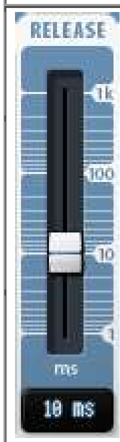
2k = 2 kHz = 2000 = 2000 Hz

2.3 Selector button

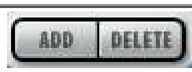
The selector button cycles through a number of fixed values. Click on the button to go to the next value. Click on the text to display a pop-up menu of the available values. The table below describes the functionality of the selector button:

Function	**Mac**	**Windows**
Go to next value	Click on the knob	Click on the knob
Go to previous value	Shift + click on knob	Shift + click on knob
Display pop-up menu of all choices	Click on text	Click on text

2.4 Sliders

	Function	**Mac**	**Windows**
	Increase/Decrease a parameter value	Click on the slider handle + drag up/down	Click on the slider handle + drag up/down
	Fine adjustment — increase/decrease	Shift + click + drag up/down	Right click + drag up/down -or- Shift + click + drag up/down
	Reset slider to default value	Command + click -or- Double-click	Control + click -or- Double-click

2.5 Buttons

	Lighted buttons show a toggle state. A green, orange or yellow light indicates "on" and a black (extinguished) light indicates "off." Click the button to toggle the state.
	Buttons that do not light up are used to activate certain commands.

2.6 Output Meters

	Output meters show the peak signal power in short time updates — in green from -96dB to -6dB, in yellow between -6dB and 0dB, and in red above 0dB. Peak hold levels are also drawn. The meter also stores the overall peak value for each channel, and displays these values in the peak indicator boxes above the meter. If the detector finds a peak value above 0dB, the text color will turn red as a warning. Click on either indicator box to reset them back to -96dB.

3. Menu Bar and Preset Manager

All Wave Arts plug-ins in the Master Restoration Suite Bundle have the following menu bar displayed at the top of the plug-in:

This section describes the operation of the menu bar, preset manager, and the other functions available in the menus.

3.1 Bypass

Clicking on the bypass button bypasses the effect, that is, audio will pass through the effect without alteration. The button is lit when the effect is bypassed.

3.2 Undo

Clicking the Undo button causes the parameters to revert to their settings prior to the last edit. Only one level of undo is available, so clicking the undo button again will restore the parameters after the edit. Both A and B buffers (described below) have their own undo buffers.

3.3 Copy

Clicking the Copy button copies the current set of effect parameters to the unused A/B buffer. Hence, if the A buffer is currently selected, the parameters are copied to B, and if the B buffer is selected, the parameters are copied to A. After clicking Copy, you can continue to make changes, and then revert to the original copied settings by clicking either the A or B buttons to switch buffers.

3.4 A/B buffers

The A/B edit buffers allow you to compare two different sets of parameters or presets. One of the A or B buttons is always lit; the button that is lit shows the current buffer. Clicking either the A or B button will switch to using the other buffer, thus changing the effect settings (assuming different settings are stored in A and B).

Here’s how to use the A/B buffers to compare two different presets. Select a preset from the Preset menu, then switch to the other buffer and select a different preset. Now switch between the two buffers to alternate between the two different presets.

3.5 Preset name and arrow controls

The currently selected preset name is displayed in the text field in the menu bar. Changing any parameters causes an asterisk (*) to be displayed at the end of the name. This indicates that changes have been made to the preset. In order to save the changes to a user preset you must select the "Save…" item in the Preset menu, described below.

The arrow controls to the left and right of the preset name cycle through the set of factory and user presets. Clicking the right arrow goes to the next preset, clicking the left arrow goes to the previous preset.

3.6 Preset menu

The Preset menu contains lists of factory and user presets for easy selection, and options for managing presets. The functions are described in the following sections.

3.7 Factory Presets

Factory presets are selected from a rolloff menu at the top of the Preset menu. Factory presets cannot be modified or deleted. The Default preset is always first in the list; it defines all default parameter settings.

3.8 User Presets

User presets are selected from a rolloff menu just below the Factory presets in the Preset menu. When you first run a Wave Arts plug-in, there will not be any user presets and the menu will be empty. When you save a preset using the “Save” option the preset is added to the User menu. All instances of a plug-in share the same set of user presets. So, after you save a preset with one instance of a plug-in, you can go to another instance and find that the preset can be found in its User preset menu too.

User presets are stored in a text file called “<plugin> Presets.txt”, where <plugin> is the name of the plug-in you are using. If the file is deleted, an empty preset file will be created automatically the next time the plug-in runs. User presets files are stored in the following directory, depending on the operating system, where <username> is your login name:

Mac OS-X:
/home/<username>/Library/Application Support/Wave Arts/<plugin>/

Windows:
C:/Users/<username>/AppData/Roaming/Wave Arts/<plugin>/

3.9 Save As…

When you have created an effect you want to save as a preset, select the “Save As…” option. You will be asked to name the preset and the preset will be saved in the set of User presets. If you supply the same name as an existing user preset, the preset will be overwritten with the new preset without any warning notice.

3.10 Save

The Save option is used to save changes to a user preset. The Save option is enabled when the current preset is a user preset and you have made changes, in which case an asterisk (*) will be appended to the preset name. Select the Save option to save the changes to the preset.

3.11 Import…

User presets can be written to files using the “Export” function, and read from files using the “Import” function. Selecting the “Import…” option will open a file chooser to select the preset file for importing. After selecting the file the presets are read and will appear in the User Presets menu. Importing does not delete existing presets; if an imported preset has the same name as an existing preset, you will get two presets with the same name.

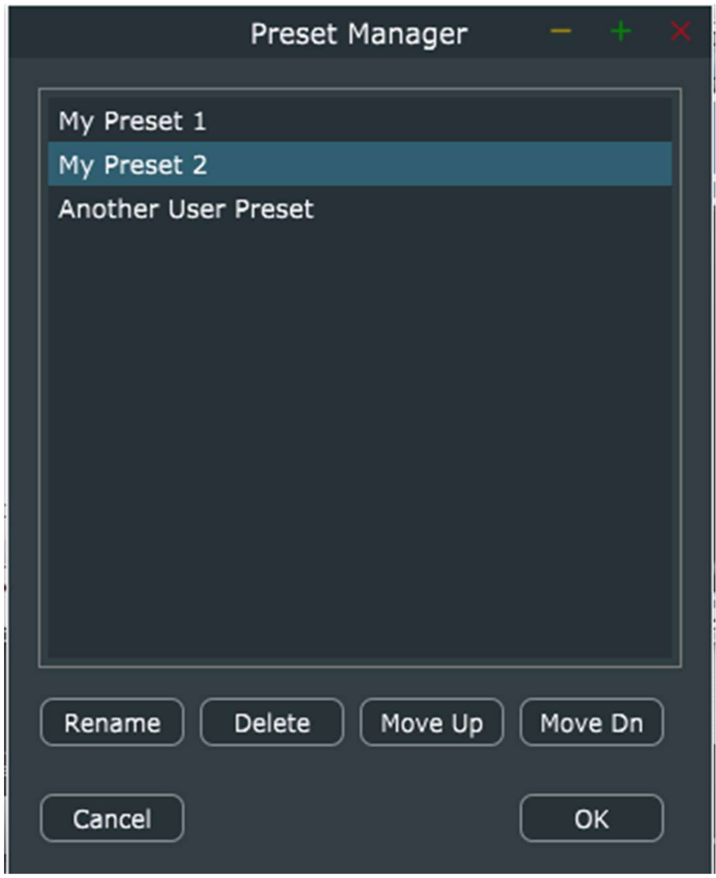
3.12 Export…

Selecting the “Export…” option will open a file chooser to specify the output file. Then your user presets are written to the file.

Preset Export is also useful for making backup copies of your user presets. If you have a large set of user presets, be sure to export them to a backup file.

3.13 User Preset Manager

The User Preset Manager provides a simple interface to organize user presets. You can rename presets, delete presets, and organize them by moving them up/down in the list.

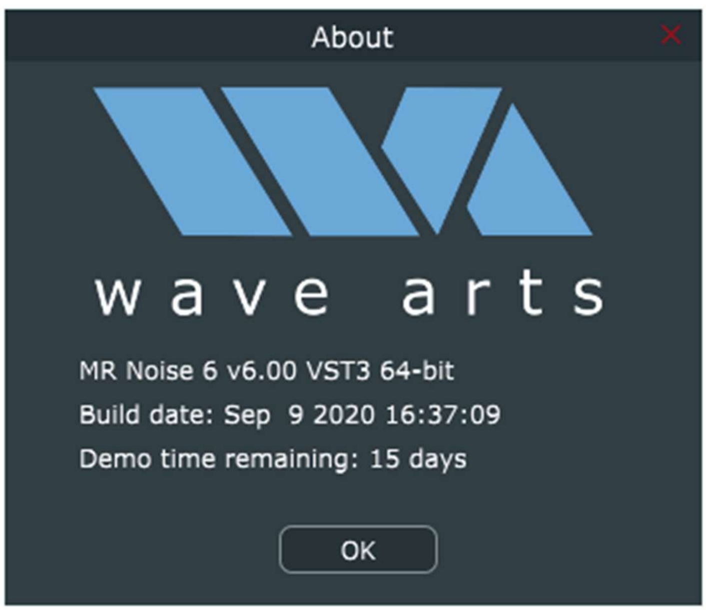


3.14 Tools menu

The Tools menu contains various important options, described below.

3.15 About…

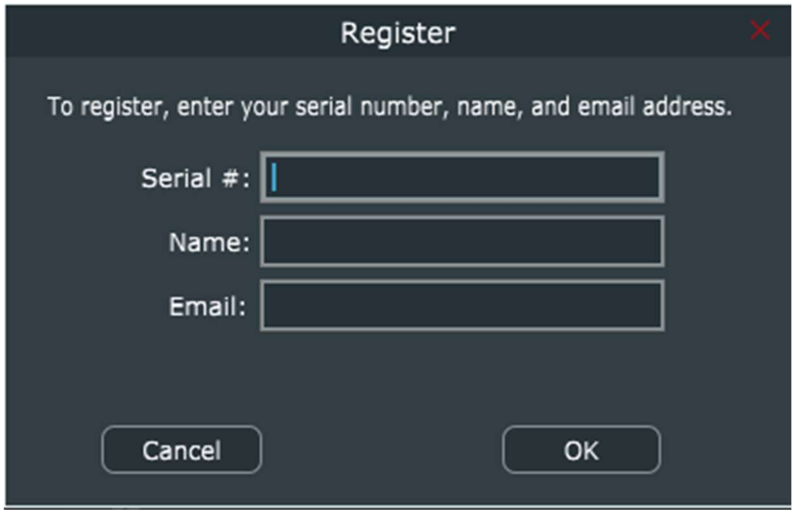
The About option displays important information about your plug-in. An example is shown below:



On the top line, the plug-in name and version are displayed, along with the current plug-in format (AAX, VST3, AU). This is useful if you aren't sure which format of the plug you are running. The build date of the plug-in is displayed on the next line. If the plug-in is using Wave Arts licensing, the registration status is displayed on the next line. If the plug-in is operating in demo mode, the time remaining (if any) is displayed. If the plug-in has been successfully registered (unlocked), the serial number is displayed. If the plug-in is using Pace/iLok licensing, it will display "Pace/iLok licensing".

3.16 Register…

Select this option to register (unlock) your plug-in. Enter your name, email address, and serial number, and click OK.



3.17 Offline Register…

Select this option to register the plug-in when your computer is not connected to the internet.

3.18 Open User Manual…

Select this option to open this user manual in a browser.

3.19 Preferences…

This option opens the Preferences dialog to customize the plug-in operation.

4. MR Hum

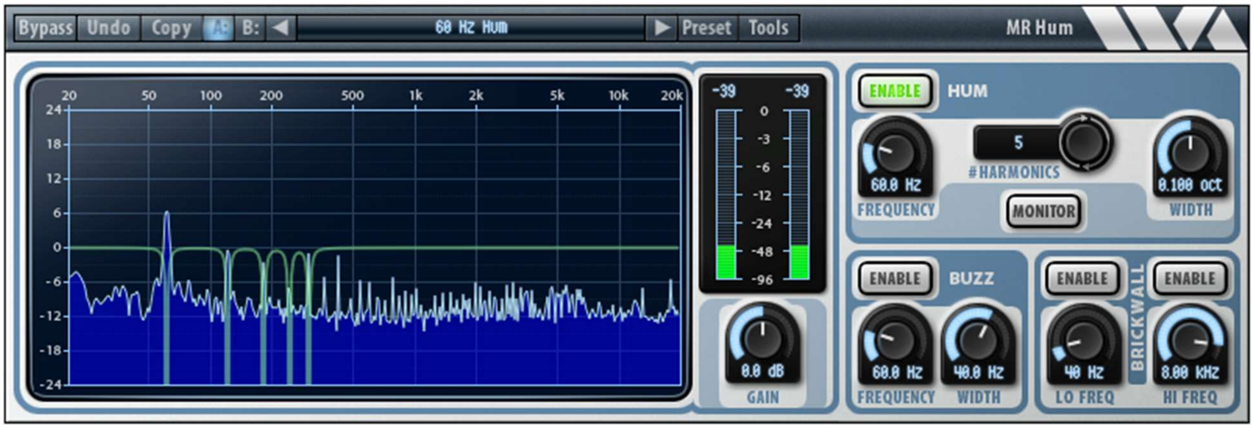


Figure 4-1. MR Hum user interface.

4.1 Overview

MR Hum combines hum removal, buzz removal, brickwall filtering, and spectrum analysis into one easy to use plug-in. The functions of MR Hum are also available in the Master Restoration plug-in. Here are some of MR Hum's key features:

- Hum removal using from 1 to 10 harmonic notch filters.
- Buzz removal for frequencies from 20 Hz to 200 Hz.
- Adjustable notch widths.
- Brickwall filtering for rumble and hiss removal.
- High resolution spectrum analysis.
- All sections can be monitored to hear removed signal.

MR Hum's audio routing and meter placement is shown in the diagram below:

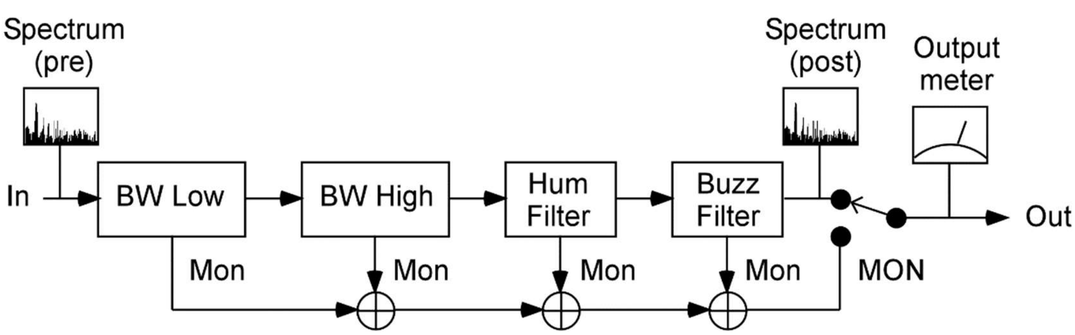


Figure 4-2. MR Hum audio routing diagram.

The input signal is first processed by the spectrum analyzer, if set to "pre" mode. The input signal is then processed by the brickwall filter. This allows the user to eliminate rumble using the low brickwall filter, or hiss using the high brickwall filter. The signal is next processed by the hum filter, which applies from 1 to 10 harmonic notch filters to cancel 50 Hz or 60 Hz hum and associated harmonics. The signal is next processed by the buzz removal filter. The result is input to the spectrum analyzer, if set to "post" mode.

When the monitor feature is enabled, the output consists of the signal that is removed by each of the enabled sections. For the low brickwall filter, the monitor will contain all frequencies up to the cutoff of the low brickwall. For the high brickwall, the signal will contain all frequencies above the cutoff of the high brickwall. For the hum filter, the monitor signal will contain only those frequencies that are notched by the hum filter; this is accomplished using a parallel sum of bandpass filters at the harmonics of the hum fundamental. For the buzz filter, the monitor signal will contain frequencies of the buzz fundamental and all harmonics. However, for the hum and buzz filters, the monitor feature does not perfectly reproduce the signal removed when the filters are enabled; it is merely an approximation. One can use the hum monitor feature to

4.2 About Hum, Buzz, and Brickwall filtering

Hum

Hum is caused by interference of audio circuitry with line power. Line power oscillates at either 50 Hz or 60 Hz depending on your world location. So in its simplest form, hum is a low frequency interference tone at either 50 Hz or 60 Hz. However, Hum often contains harmonics (multiples) of the fundamental frequency. For example, in a region with 60 Hz power, interfering hum would typically contain frequencies of 60 Hz, 120 Hz, 180 Hz, and so on. Below is a frequency spectrum of typical hum with a 60 Hz fundamental. It shows significant energy at 60 Hz, 120 Hz, 180 Hz, 300 Hz, and 420 Hz, i.e. the fundamental, 2d, 3d, 5th, and 7th harmonics.

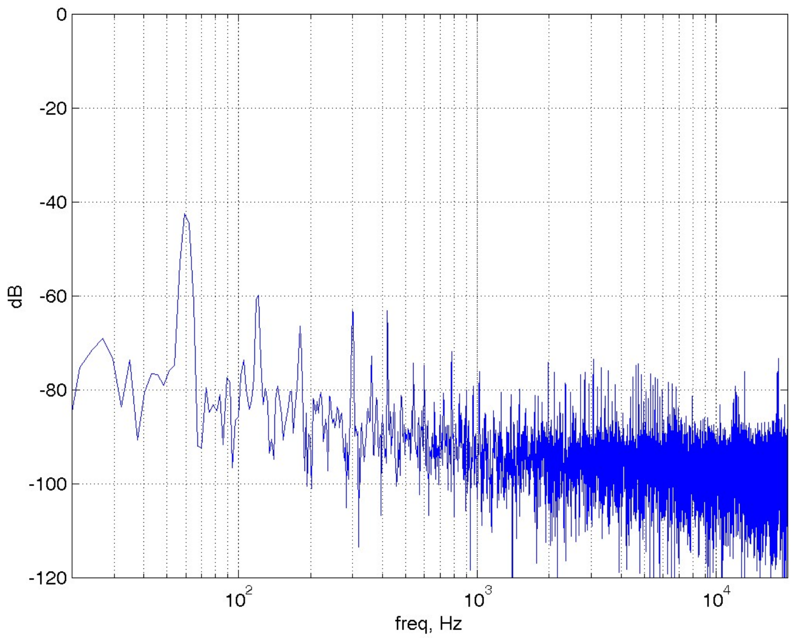


Figure 4-3. Frequency spectrum of typical 60 Hz hum.

Hum is generally caused by poor grounding of audio equipment. Ground loops, where the ground connections between different pieces of equipment create a loop circuit, form a loop antenna which is effective at picking up line power oscillations. A typical ground loop can be created when two pieces of grounded equipment, say a pre-amp and an amplifier, are connected with a shielded audio connector that completes a ground loop. Ground loops can be broken by breaking the ground connection on the audio path. It is also possible to lift the ground connection of a piece of equipment, but this is a dangerous practice because it eliminates the safety feature of a ground, which is to prevent electrocution. Even without ground loops, hum can be created when sensitive audio electronics share a ground with other equipment that may be discharging significant current through the ground. Hum harmonics are created when electronic equipment rectifies AC power to create DC, or by other devices such as dimmers.

Hum is best removed by using a set of sharp notch filters to eliminate the fundamental and harmonic frequencies of the hum. Sharp notch filters work well because the frequency of line power is very stable so the filters can be very narrowly tuned. MR Hum uses up to 10 notch filters to eliminate up to the 10th harmonic.

The frequency of hum depends on the frequency of the local line power. 60 Hz power is used in North America (except Greenland), Central America, most of South America, Saudi Arabia, and parts of southeast Asia including Korea. The rest of the world, including Europe, Asia, Africa, and Australia uses 50 Hz power.

Buzz

Buzz is a periodic impulsive contamination which like hum can be caused by poor grounding. The simplest example of a buzz is that of a very bad hum whose harmonics extend up to very high frequencies. However, buzz can also be caused by a periodic signal, like a camera motor, which is recorded along with the audio program. The plot below shows buzz created by very poor grounding with 60 Hz line power. The periodicity of the signal is about 16.7 msec.

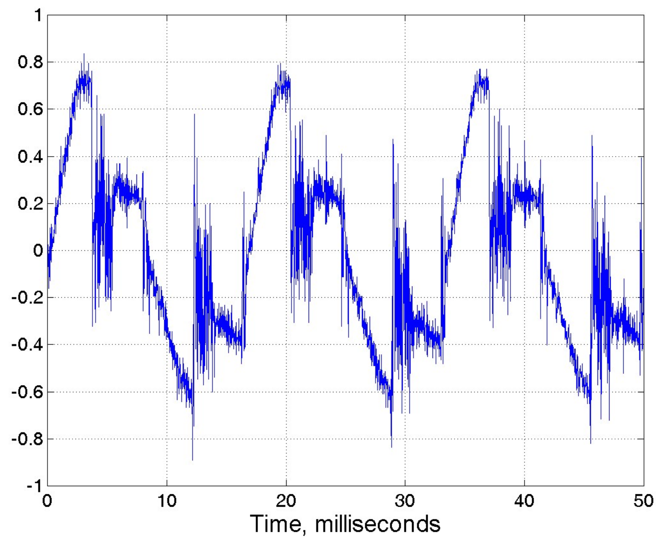


Figure 4-4. Time plot of 60 Hz buzz.

Buzz removal differs from hum removal in that the buzz harmonics extend to high frequencies, and the fundamental frequency might be something other than 50 or 60 Hz. Buzz processing consists of a set of notch filters located at the fundamental buzz frequency and at all harmonics in the audio range. The set of notch filters is created using a “comb filter” which essentially computes the difference of the input signal with a copy of the input signal delayed by one fundamental period. The difference calculation removes any periodic contamination. However, the side effect imparted to the signal is that of a comb filter, that is, an echo with a very short delay. This can sound like a hollow room ambience and may be objectionable. Hence, buzz removal imparts significant side effects.

Brickwall filters

Brickwall filters are lowpass or highpass filters with very steep cutoffs, used to pass all frequencies up to the cutoff frequency and eliminate all frequencies beyond the cutoff. The frequency range that is passed unaltered is called the "passband", the frequency range that is attenuated is called the "stopband". MR Hum's brickwall filters are implemented using 10th order elliptical filters, with at least 90 dB of stopband attenuation and less than 0.1 dB of passband ripple.

Brickwall filters are used to eliminate unwanted frequency ranges. One use of the low brickwall filter is to eliminate low frequency rumble which can occur from phonograph playback or due to wind or other low frequency noise during an acoustic recording. Brickwall filters can also be used when processing a noisy recording of an instrument sound that does not use the entire frequency range. The brickwall filter would be positioned at the edge of the instrument's frequency range to eliminate out-of-band noise. So for example, when processing voice, one could use the brickwall filters to eliminate all frequencies below 100 Hz and above 8 kHz.

The brickwall filters can also be used to zero in on a particular frequency range just for analysis purposes. For example, one could use the brickwall filters to listen to selected overtones in an organ, or to isolate the sibilants of a human voice.

4.3 Using MR Hum

MR Hum has three sections: the hum filter, the buzz filter, and the brickwall filters. Each section can be enabled with the corresponding ENABLE button. Hence the enable buttons can be used to compare the results of processing enabled with processing disabled (bypassed). The monitor feature is also extremely useful to tune the parameters by listening. The spectrum analyzer can be used to visualize the spectrum before and after processing.

Spectrum analysis

Spectrum analysis works by transforming the time signal into its constituent frequencies and displaying the resulting spectrum amplitudes as a function of frequency. You can see at a glance what frequencies are present in your signal. MR Hum uses a long time window when transforming blocks of the input signal; hence the resulting spectrum has very good resolution at low frequencies which makes it easy to see low frequency tones as spikes in the spectrum. See figure 4-2 for an example of a spectrum display. Because of the long time block sizes, the update rate is fairly slow, just a few updates per second. When using the spectrum analyzer to visualize hum or buzz, it is best to audition a portion of your signal which contains only the hum or buzz.

MR Hum spectrum analysis can be set to “pre” mode, meaning it runs before hum processing, or “post” mode, meaning it runs after hum processing; see figure 4-1. You can select between the two modes by clicking on the vertical axis of the display; this causes a popup menu to appear containing the two choices. By default the post mode is selected. The vertical range of the spectrum analysis is fixed: the bottom of the display is -120 dB and the top of the display is 0 dB, hence each division is 15 dB.

Brickwall

Each of the low and high brickwall filters has an enable button and a knob for setting the cutoff frequency. When one of the bands is enabled, a corresponding brickwall control handle appears in the frequency response display. You can edit the brickwall frequency by dragging the control handle or dragging the frequency knob. The monitor feature can be used to hear the signals that are removed by the brickwall filters. If both the low and high brickwall filters are enabled, the monitor signal will contain both low and high frequencies.

Hum

If your signal has an obvious hum, the first place to start is to enable the Hum section, then select either the 50 Hz or 60 Hz hum preset depending on the frequency of the hum. In the case where you are processing a signal of unknown origin, you should be able to see the fundamental of the hum frequency in the spectrum analysis display. You can also hear the hum frequency using the monitor feature. Press the MONITOR button to enable monitoring. This will pass only the frequencies of the hum fundamental and the selected harmonics. Then alternately select the 50 Hz or 60 Hz preset. One of these will pass the hum.

The default notch width is 0.1 octave. This is the bandwidth of the notch filters defined as the width in octaves between the half-power frequencies. Hence, with a width of 0.1 oct, the notch at 60 Hz will have half power (-3 dB) response at 58.0 Hz and 62.1 Hz. Obviously these filters are very sharp. The notches are infinite depth; at the center frequency the response is 0 which equals -inf dB.

Very narrow width filters have a long time response. Consequently, a narrow notch filter will take some time to reject the hum. When applying the hum filter to a signal with hum, if you notice that the hum persists at the very start of the signal, you can increase the width of the notch filters and this will make the filters respond faster to the onset of the hum.

Deciding how many harmonics should be used can be done by examining the spectrum display which is overlaid with the hum frequency response showing the notch filters. As you increase the number of harmonics you can see if the spectrum has a corresponding spike at that frequency. If there is no corresponding spike in the spectrum, then the newly added notch will have little effect. In some cases you may have to use a large number of notches in order to cancel a high order harmonic. For example, in Fig 4-3, one would select 7 harmonics in order to reach the 7th harmonic which has significant energy. Although the 4th harmonic of the hum is not present, there is no way to disable the 4th harmonic of the notch filter.

One can also listen to the harmonics using the monitor feature. With monitor enabled, try increasing the number of harmonics starting with 1, and continue adding harmonics if the hum gets progressively louder.

If your audio has been resampled or recorded on magnetic tape, it’s possible that the hum fundamental will no longer be exactly 60 Hz or 50 Hz. You can use the monitor feature and slowly adjust the hum fundamental until the loudness of the monitored hum is maximized. Hold down the right mouse button while adjusting the hum frequency knob to get super fine tune mode.

The frequency response of the hum filter is shown in the MR Hum display. This reflects the fundamental frequency, the number of harmonics, and the width of the notch filters. However, the parameters cannot be edited by clicking in the display.

Buzz

If the hum is bright and buzzy, you should try processing with the buzz filter instead. In this case you should disable the hum filter. The buzz filter can also be used for any sort of periodic contamination. After enabling the buzz section, the first step is to determine the buzz frequency. If you know the buzz is created by line power, then enter the fundamental in the buzz frequency control by adjusting the knob or by typing in the value. You can also examine the spectrum display and look at the frequency of the buzz fundamental. You can also use the monitor feature while slowly adjusting the frequency to scan for the proper fundamental. When the frequency parameter is adjusted close to the fundamental, the monitor will begin to pass the buzz sound. Similarly, with monitoring off and with the spectrum analysis set to “post” mode, you can scan for the buzz fundamental and see when the buzz harmonics are canceled by the buzz filter; this should be plainly audible as well.

Once the fundamental is determined, turn off the monitor and proceed to adjust the width control. Generally, larger widths will reduce the resonant sound of the comb filter. Now try enabling/disabling the buzz section to hear the effect of the buzz processing. It’s possible the comb filter effect imparted by the buzz filter is more annoying than the buzz. If so, you can seek other methods to remove the buzz. Broadband noise reduction as provided by MR Noise can be extremely effective in removing buzz. Also the buzz can be gated in quiet sections using MR Gate.

4.4 Parameters

The table below lists all the internal parameters of MR Hum and shows the range of values as would be displayed by a generic parameter-value plug-in interface. Most of these parameters have a one to one correspondence with controls on the user interface.

Parameter name	**Values**
Enable	0 = Off, 1 = On
Monitor Enable	0 = Off, 1 = On
Brickwall Low Enable	0 = Off, 1 = On
Brickwall Low Freq	20 – 20000 Hz
Brickwall High Enable	0 = Off, 1 = On
Brickwall High Freq	20 – 20000 Hz
Hum Enable	0 = Off, 1 = On
Hum Frequency	20 – 200 Hz
Hum Width	0.05 to 0.2 octaves
Hum Num Harmonics	1 - 10
Buzz Enable	0 = Off, 1 = On

Buzz Frequency	20 – 200 Hz
Buzz Width	10 – 100 Hz
Gain	-12 to +12 dB
Monitor Gain	-12 to +12 dB

The parameters are described in more detail below.

Enable — This parameter is only available in the standalone MR Hum, where it functions as the global bypass control. The Master Restoration plug-in does not display an enable control for the Hum&Buzz section and the Hum&Buzz section is always enabled.

Monitor Enable — This enables/disables the monitor feature. When monitoring is enabled, the output signal consists of all the signals that would be removed by the MR Hum filters.

Brickwall Low Enable — Enables or bypasses the brickwall lowpass filter.

Brickwall Low Freq — Sets the brickwall lowpass filter cutoff.

Brickwall High Enable — Enables or bypasses the highpass brickwall filter.

Brickwall High Freq — Sets the brickwall highpass filter cutoff.

Hum Enable — Enables/disables the hum removal filter.

Hum Frequency — This parameter sets the fundamental frequency of the hum, from 20 to 200 Hz. Typically you will use 50 Hz or 60 Hz which can be selected using the presets.

Hum Width — This parameter sets the bandwidth of the hum notch filters in octaves relative to the hum fundamental, where bandwidth is defined as the difference between the half-power (-3 dB) points of the notch response.

Hum Num Harmonics — Sets the number of notch filters used in the hum filter, from 1 to 10. This should be set to the number of prominent harmonics in the hum for maximum cancellation.

Buzz Enable — Enables/disabled the buzz removal filter.

Buzz Frequency — This parameter sets the fundamental frequency of the buzz, ranging from 20 to 200 Hz.

Buzz Width — This parameter sets the bandwidth of the buzz notch filters in Hz, where bandwidth is defined as the difference between the half-power (-3 dB) points of the notch response.

Gain — Output gain in dB.

Monitor Gain — Monitor output gain in dB.

4.5 Presets

MR Hum factory presets are listed in the table below. In the standalone plug-in these appear in the preset menu in the menu bar. In Master Restoration, these presets appear in the Hum&Buzz Preset selector.

Name	**Description**
Default	Default settings, 60 Hz hum removal
60 Hz Hum	60 Hz hum removal
50 Hz Hum	50 Hz hum removal
60 Hz Buzz	60 Hz buzz removal
50 Hz Buzz	50 Hz buzz removal
Rumble	Low frequency rumble filter
Speech band	Isolate speech range

4.6 Specifications

Description	Hum and buzz suppression filters, brickwall filters for rumble and hiss suppression, and spectrum analysis
Operating Systems	Windows 10; Mac OS X 10.11 or higher
Plug-in Formats	VST3, AU, AAX
Sampling Rates	up to 192 kHz
Latency	none
I/O Formats	mono-mono, stereo-stereo

5. MR Click

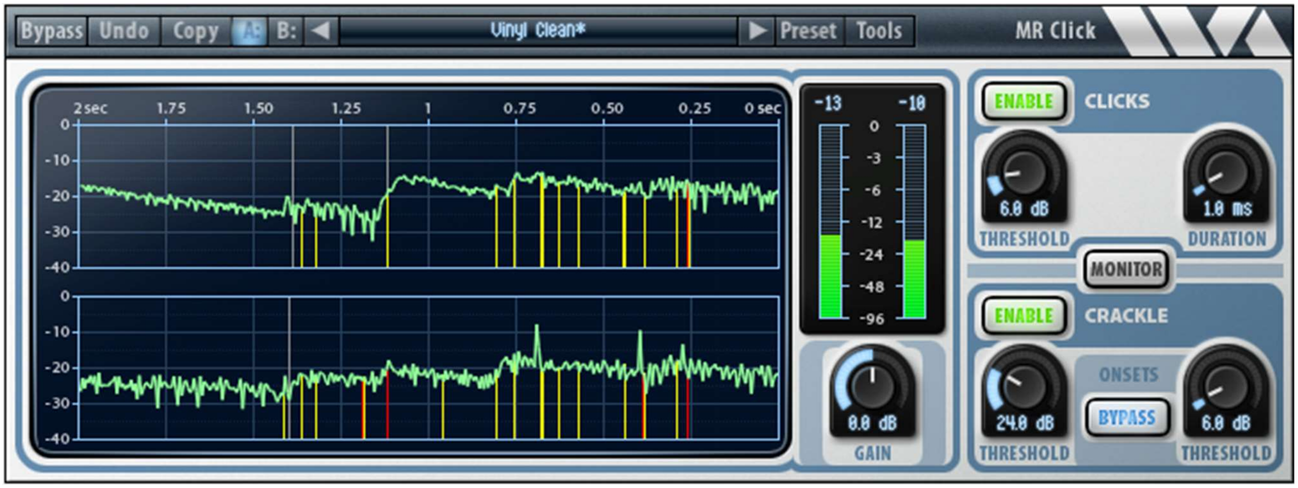


Figure 5-1. MR Click user interface.

5.1 Overview

MR Click combines click and crackle filtering specifically intended for restoring phonograph recordings. The functions of MR Click are also implemented as part of the Master Restoration plug-in. Some of MR Click's features include:

- Separate algorithms optimized for click and crackle processing.
- Sophisticated detection algorithms for clicks and crackle focus processing on contaminating events.
- Onset detection allows crackle processing to bypass onset events, keeping attacks bright.
- Monitoring feature for hearing eliminated clicks and crackle.
- Comprehensive stereo display of signal level, clicks, crackle, and onset events.

A diagram of MR Click's audio routing and meter placement is shown below.

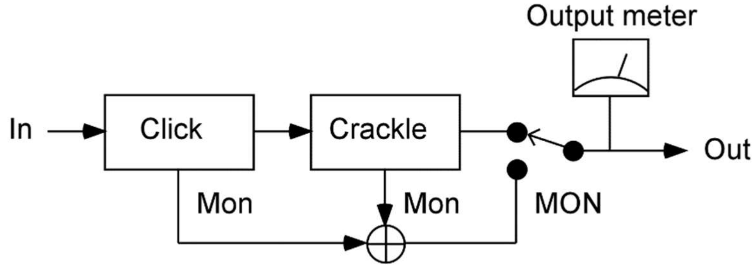


Figure 5-2. MR Click audio routing diagram.

The input signal is processed by the click filter first, then the crackle filter. When the monitor feature is enabled, the output consists of the signal that is removed by each of the enabled sections. If both the click and crackle processors are enabled, the monitor signal will contain both removed clicks and crackle.

5.2 About Clicks and Crackle

The playback of phonograph recordings often contains noticeable clicks and crackle caused by degradation of the recording grooves, surface scratches, or accumulation of foreign matter in the grooves. We use the term “click” to mean a large disturbance, such as caused by a scratch or an obstruction in the groove. Clicks might also be called pops. The disturbance that causes a click is always so large that both stereo channels will be similarly affected. We use the term “crackle” to mean the frequently occurring, lower amplitude disturbances due to groove degradation or accumulated dust. These are small enough to affect just one wall of the recording groove, and hence crackles will often occur on just one stereo channel at a time.

The plot below shows a typical phonograph click. The click is considerably louder than the surrounding signal.

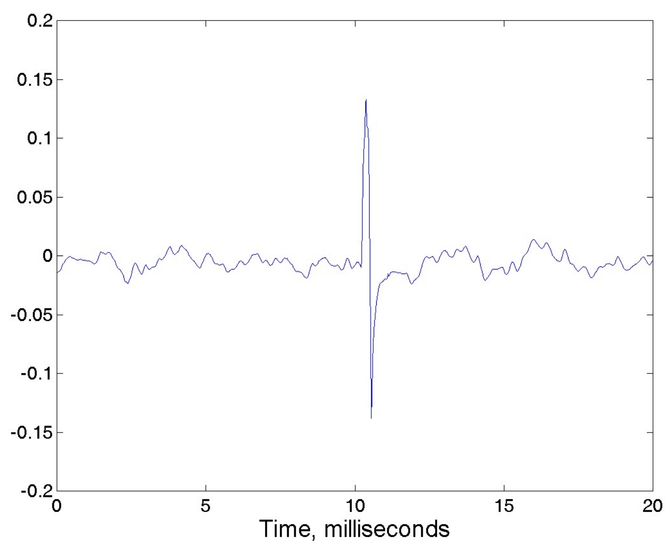


Figure 5-3. Plot of a vinyl phonograph click during a signal. Only one channel is shown; however both channels were affected by the click event.

Clicks may also be present in digital audio signals that have undergone some kind of error in transmission or processing. If the contamination results in various samples taking on random values, these will sound similar to phonograph clicks and can be effectively suppressed using MR Click.

The plot below shows a typical phonograph crackle event. Although not shown, this crackle event affects only one channel. The level of the crackle is similar to the level of the surrounding signal. Typical crackle events have a shorter duration than clicks and have a correspondingly higher frequency content.

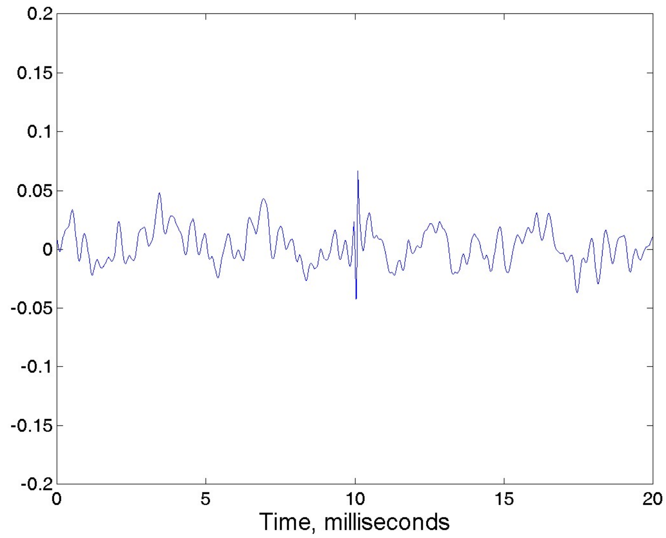


Figure 5-4. Time plot of a vinyl phonograph crackle event. Only one channel is shown, and only one channel was affected by the crackle event.

Processing clicks and crackle

The strategy for processing clicks and crackle is to detect the occurrence of click and crackle events, and then alter the signal around the event to suppress the sound of the click or crackle. Both the detection strategy and the suppression strategy differ for click and crackle events.

Clicks are detected as a rapid onset and offset of energy. Clicks are suppressed by overwriting the signal with a linear interpolation during the click event. The time of the interpolation is a user parameter; typically 1 msec is ample to completely mask the click.

Crackle is detected as a rapid onset of high frequency energy. Crackle is suppressed by smoothing the signal around the crackle event using a lowpass filter.

In many cases, crackle caused by degradation of the phonograph groove is not distinguishable from the content of the original recording. Typical percussive events, such as snare drum hits, ride cymbal hits, etc., contain rapid onsets of high frequency energy that are indistinguishable from crackle events. If the crackle processing was applied uniformly, the percussive attacks would be dulled by the crackle smoothing. To alleviate this problem, MR click has an option to bypass crackle processing during note onsets. MR Click contains a broadband onset detector that will trigger on note onsets. When the bypass onsets option is engaged, the crackle processing will be bypassed during note onsets.

5.3 Using MR Click

MR Click Display

Operation of MR Click is greatly facilitated by the event display shown below:

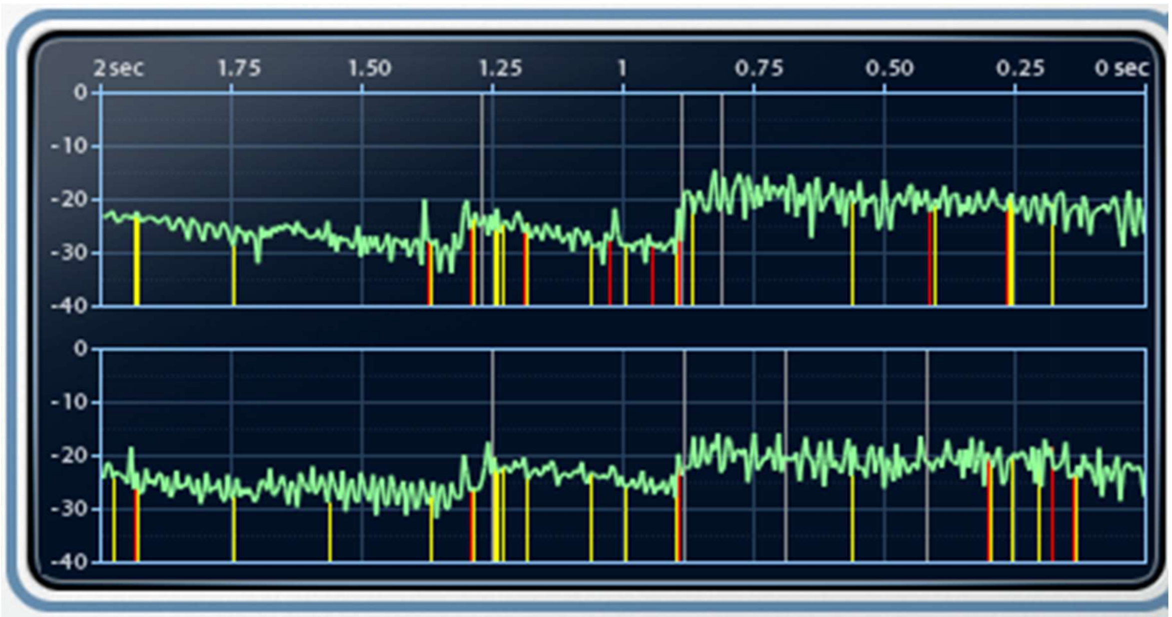


Figure 5-5. Click and Crackle event display. The green plot shows signal level, red lines are clicks, yellow lines are crackle, and grey lines are onset events.

The display is active when MR Click is enabled. The display shows the past few seconds of signal history and scrolls from right to left as audio playback proceeds. The green plot shows input signal level. Red lines indicate detected click events. Yellow lines indicate detected crackle events. White lines indicate detected onset events. These are summarized in the table below.

Green	Input signal level
Red	Click events
Yellow	Crackle events
Grey	Onset events

Click events are shown only if the click processor is enabled. Crackle and onset events are shown only if the crackle processor is enabled.

To change the time range, right-click (SHIFT-click on Mac) the horizontal time axis and a popup menu will appear with different choices for the time range in seconds.

To change the vertical range, right-click (SHIFT-click on Mac) the vertical amplitude axis and a popup menu will appear with different choices for the amplitude range in dB.

MR Click Operation

When processing phonograph recordings to remove clicks and crackle, the strategy is to first remove the loud clicks and pops and then remove the lower level crackle.

You should first audition the material to be processed and listen for offending clicks. Then, rewind your track and enable the click processor by clicking on the ENABLE button. Start playback of the track and adjust the click threshold knob until all the offending clicks are detected. Lower threshold values will detect more click events, and higher threshold values will detect fewer click events (only the loudest clicks will be detected at high threshold settings). You can see the clicks appear in the event display as red lines. You can also enable the monitor feature and listen to the detected clicks in the monitor signal. When the monitor feature is off, you can also enable/disable the click processor to compare the results of click removal versus no processing.

A click duration of 1 msec usually works well, but in some cases you might want to lengthen the duration, if there is still a residual thump sound after click removal.

Once you are satisfied that the loud clicks and pops have been suppressed, you can proceed to process the crackle. Again, it is good to audition the track ahead of time to get a good idea of the amount of crackle present. Then enable the crackle processor and begin playback. You will see the detected crackle events appear as yellow lines in the display. Adjust the threshold to change the number of crackle events detected; lower thresholds will detect fainter crackle and will create more crackle events, while higher thresholds will detect louder crackle and will create fewer crackle events.

The monitor feature is particularly useful for hearing the detected and removed crackle events. Note that if the click processor is still engaged you will also hear the removed click events. With lower thresholds, the crackle processor will detect high frequency portions of your signal and process these as crackle; this will be apparent when monitoring because you will be able to hear parts of your signal in the monitor signal. This is a good indication that the threshold is set too low. Basically, the monitored crackle signal should sound like random crackle events and should not resemble your input signal. If your signal is fairly percussive, it is likely that the crackle processor will detect the percussive onsets as crackle events. This will be audible in the monitor signal because you will hear the underlying rhythm. Your choices here are to increase the threshold, or to engage the onset bypass mechanism.

To use the onset bypass, adjust the onset threshold until you see onset events (white lines in the display) that correspond to the underlying rhythm of your track. Lower thresholds result in more onsets events; higher thresholds reduce the number of onset events. Now enable the onset bypass by clicking on the BYPASS button. When monitored, you should hear that the crackles corresponding to musical onsets are no longer present, or at least reduced, when the onset bypass is engaged.

It is unrealistic to expect that you will be able to remove all audible clicks and crackle without adversely affecting the sound of the original signal. A better strategy is to remove the most offending events, leaving the signal as untouched as possible. Because crackle is most audible during silent portions, say between tracks, one might be tempted to lower the threshold until all of the crackle is suppressed during these silent portions. But this is probably overdoing it, and there are other techniques which can be used to clean up the quiet portions. Both broadband noise reduction, as provided by MR Noise, and noise gating, as provided by MR Gate, can clean up the quite portions. Hence you should focus on removing the clicks and crackle that are audible while the music is playing.

5.4 Parameters

The table below lists all the internal parameters of MR Click and shows the range of values as would be displayed by a generic parameter-value plug-in interface. Most of these parameters have a one to one correspondence with controls on the user interface.

Parameter name	**Values**
Enable	0 = Off, 1 = On
Monitor Enable	0 = Off, 1 = On
Click Enable	0 = Off, 1 = On
Click Threshold	6 – 20 dB
Click Duration	0.5 – 10 msec
Crackle Enable	0 = Off, 1 = On
Crackle Threshold	12 – 40 dB
Onset Bypass	0 = Off, 1 = On
Onset Threshold	0 – 40 dB
Gain	-12 to +12 dB
Monitor Gain	-12 to +12 dB

The parameters are described in more detail below.

Enable — This parameter is only available in the standalone MR Click, where it functions as the global bypass control. The Master Restoration plug-in does not have an enable for the Click & Crackle section.

Monitor Enable — This enables/disables the monitor feature. When monitoring is enabled, the output signal consists of the click and crackle signals that are removed by MR Click.

Click Enable — Enables or bypasses the click processor.

Click Threshold — Sets the click detection threshold. Lower values result in many faint clicks detected, high values result in only a few loud clicks detected.

Click Duration — Sets the interpolation time for each click. MR Click will overwrite the click event with a linear interpolation starting 0.5 msec before the click and lasting a duration set by this parameter.

Crackle Enable — Enables/disables the crackle removal filter.

Crackle Threshold — This parameter sets the crackle detection threshold. Lower values result in many faint crackles detected, high values result in only a few loud crackles detected.

Onset Bypass — When onset bypass is engaged, crackle processing is bypassed during detected onset events. This is useful to prevent decrackling of percussive onsets that can easily be recognized as crackle.

Onset Threshold — This parameter sets the onset detection threshold. Lower values result in many faint onsets detected, high values result in only a few loud onsets detected.

Gain — Output gain in dB.

Monitor Gain — Monitor output gain in dB.

5.5 Presets

MR Click factory presets are listed in the table below. In the standalone plug-in these appear in the preset menu in the menu bar. In Master Restoration, these presets appear in the Click & Crackle Preset selector.

Name	**Description**
Default	Default settings with nothing enabled
Vinyl Clean	Typical vinyl phonograph settings, both click and crackle
Vinyl Light Clean	Light cleaning settings for vinyl
Vinyl Hard Clean	Hard cleaning settings for vinyl
Clicks	Click removal
Crackle	Crackle removal

5.6 Specifications

Description	Click and crackle removal filters
Operating Systems	Windows 10; Mac OS X 10.11 or higher
Plug-in Formats	VST3, AU, AAX
Sampling Rates	up to 192 kHz
Latency	click: 10.5 milliseconds when enabled crackle: 3 milliseconds when enabled
I/O Formats	mono-mono, stereo-stereo

6. MR Noise

Figure 6-1. MR Noise user interface.

6.1 Overview

MR Noise is broadband noise reduction processor. Here are some of MR Noise’s key features:

- Great sounding with minimal artifacts.
- Low latency (under 20 msec) noise reduction can be used in live situations.
- Fast noise floor learning time (down to 50 msec), hence noise looping is not required.
- Auto dynamics mode keeps transients sharp while avoiding squirrelly artifacts.
- Monitor feature to hear removed noise.
- Very easy to use.
- Comprehensive display shows spectrum of input signal, output signal and noise floor.
- Parameters can be edited as function of frequency.

A diagram of MR Noise's audio routing and meter placement is shown below.

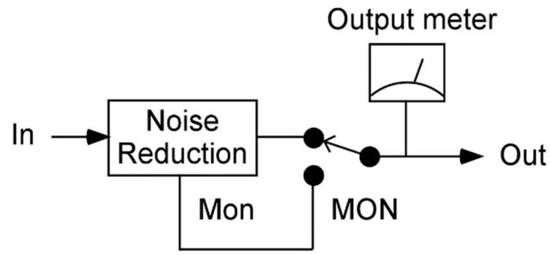


Figure 6-2. Audio routing in MR Noise.

In normal processing, the noise processor removes noise from the signal and outputs the noise reduced signal. When the monitor feature is enabled, the output consists of the signal, i.e. the noise, that is removed by the noise processor.

6.2 About Noise Reduction

Noise

Noise is a general term that can apply to various types of unwanted artifacts in an audio signal, including hum, buzz, clicks, crackle, rumble, and hiss. We use the term "broadband noise" to describe noise consisting of a random signal which sounds like hiss. This type of noise is present in every audio signal. There are many sources of broadband noise:

Digital quantization – in digital signals, the quantization of the signal to a certain bit resolution creates broadband noise called quantization noise.
Acoustic recordings – there is always some acoustic noise, which can be due to ventilation sounds or other background noises.
Electronics – all electronics, particularly high gain amplifiers, will add noise to a signal due to thermal variation in the amplifier elements.
Magnetic tape recordings – magnetic tape is composed of individual grains of magnetic material which are each magnetized differently to record an audio signal. The finite number of grains and limited tape speed causes random noise during playback.
Phonograph recordings – surface irregularities cause playback noise.

An important characteristic of the noise is its spectrum, that is, the distribution of energy at different frequencies. The noise spectrum is also called the noise "floor" , because it defines the lowest signal level at each frequency that can be distinguished from the noise. In order for a signal to be audible, it has to be above the noise floor. The noise floor is also called the noise "profile" in other noise reduction literature.

For many sources of noise, the noise floor spectrum will remain fairly constant over time. This is typical of quantization noise, electronic noise, magnetic tape noise, and phonograph noise. Acoustic noise, however, can either be constant or varying. We would expect the acoustic noise in a quiet recording room to be fairly constant. But the acoustic noise in an outdoor environment can vary significantly over time due to traffic noise, wind, or other background sound.

Noise reduction

The noise reduction algorithm used by MR Noise is very effective at removing noise with a constant spectrum. The MR Noise algorithm operates in the frequency domain. At each frequency, it compares the signal level with the noise floor. If the signal level is well above the noise floor, it is unchanged, but signals that are close to the noise floor are attenuated. This is like having a noise gate running at each frequency, where the gate threshold is set to the noise floor level at that frequency.

Learning the noise floor

MR Noise must first learn the noise floor spectrum. This is done by finding a portion of the recording where there is just noise without signal. The noise sample is played through MR Noise with the LEARN option enabled. When learning, MR Noise computes the noise spectrum, and when learning is complete, MR Noise remembers this spectrum and uses it for subsequent noise removal. The MR Noise algorithm requires only a short noise sample, down to 50 msec, to learn the noise spectrum effectively, although longer learning times give more accurate results.

To handle varying noise floors, there are two approaches. One can learn a long varying portion of noise and use this average floor, or one can segment the audio into different clips and learn different noise floors for each clip. The latter method is more effective.

Reduction amount

It is often necessary to limit the maximum amount of attenuation. This is done via the Amount parameter, which specifies the maximum attenuation. By limiting the maximum attenuation, we limit the amount of noise reduction, but also limit processing artifacts.

Knee shape

As described above, the noise reduction algorithm is essentially a dynamics processor applied to each frequency band in the spectrum. Maximum attenuation is applied when the signal level falls equal (or below) the noise floor. Signal levels above the noise floor are attenuated less because they are less likely to be noise. When the signal rises to about 20 dB over the noise floor, there is very little attenuation, since it is very unlikely the signal could be noise. So this is like a gate or expander dynamics curve where there is no attenuation at high signal levels with respect to the noise, but attenuation is applied as the signal level gets lower. In the MR Noise algorithm, the curve of attenuation versus level above noise floor is defined by the Knee parameter. There are three knee shapes: soft, medium, and hard. These actually represent well defined theoretical approaches to noise reduction: Soft is spectral magnitude subtraction, Medium is Weiner filtering, and Hard is spectral power subtraction. But we can also think of them as just different knee shapes that control how fast the attenuation is applied as the signal approaches the noise floor.

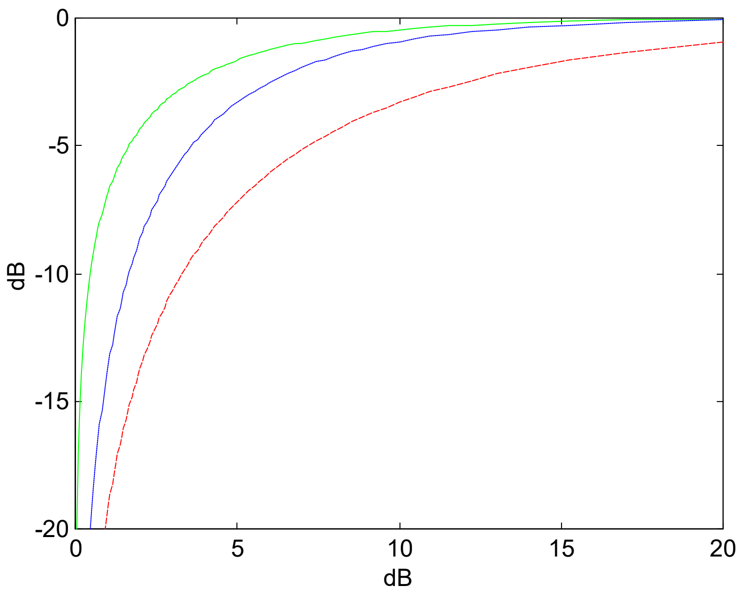


Figure 6-3. Plots of attenuation as a function of level in dB above the noise floor. The three knee shapes are shown: red dashed = Soft, blue dotted = Medium, green solid = Hard.

Musical noise

If the noise reduction is applied to noise alone, then portions of the noise will survive the noise reduction processing because by chance they are distributed above the average noise floor. The result is a set of tone bursts occurring at random times and frequencies, called *musical noise*, which despite being an interesting phenomenon, also sounds fairly annoying. It sounds like trickling water, but with lots of resonance, as if one was listening to it through a tube. In order to prevent musical noise from occurring it is necessary to raise the knee point above the noise floor so that more attenuation is applied to signals just above the noise floor. In the MR Noise plug-in, the Threshold parameter specifies an offset that raises the knee point above the noise floor. About 6-9 dB of noise floor offset is usually sufficient to completely silence any musical noise.

Dynamics

Another artifact of noise reduction is the possibility of individual harmonics turning on and off as they decay approaching the noise floor. We can reduce the possibility of individual frequencies gating on and off by introducing some time dependent filtering to each frequency gain, in the form of attack and release rate controls. The release control determines the maximum rate of attenuation for decaying signals.

The attack control determines the maximum rate of boost for increasing signals. Using short attack and release times means the frequency band gains will respond very quickly to the input signal. Using longer attack and release times means the frequency band gains will respond slower to changing inputs.

6.3 Using MR Noise

Learning the noise floor

The first step is to identify a section of your track that contains only background noise. Often this can be found at the very start of the track before the signal comes in. However, if the track has already been cropped to eliminate leading and trailing signal, then you will have to find noise during a silent section in the track. MR Noise can learn a noise profile with as little as 50 msec of noise, so this could be found between words in a dialog track for example. If you can't find any part of the track which is quiet then the noise floor will have to be estimated by learning the quietest portion you can find and then manually editing the floor parameters to lower and smooth the floor. Editing the noise floor is described later.

Figure 6-4. Learn time and Learn button.

The learn time is specified using the LEARN TIME parameter. It has various choices from 50 msec to 5 seconds, and also has a manual mode. Manual mode would be used for doing very long averages. Choose the longest learn time that is shorter than the portion of noise you have identified.

If you are using the Master Restoration plug-in, first enable the noise section by clicking on the ENABLE button. If you are using the MR Noise plug-in it should already be enabled unless it happens to be bypassed, in which case you should turn off the bypass.

Now click the LEARN button. It will light in a lovely purple color. Now start playback of your track. During learning you can see the noise floor evolve in the spectrum display as the averaging takes place. When learning is complete the LEARN button will extinguish and the displayed noise floor spectrum is smoothed as it is converted from a spectrum to a set of 31 noise floor parameters spaced at 1/3-octave bands. For stereo inputs, MR Noise learns independent noise floors for the left and right channels.

At this point the noise reduction is active. You can adjust the noise reduction parameters as described below.

Manual learning can be used for learning very long portions of noise, or for controlling the start of learning. When the LEARN TIME is set to Manual, learning will continue for as long as the LEARN button is on. You must click LEARN again to turn off learning.

Adjusting the noise reduction

Assuming you are starting from the default preset values, the noise reduction parameters will be set to something sensible. Recommended parameters are: AUTO mode enabled, Soft Knee type, Threshold from 6 to 9 dB to eliminate musical noise, and Amount from 3 to 40 dB based on preference.

Adjusting the Threshold can be done easily if you have a quiet portion of the track that contains only noise. When this is processed adjust the threshold as low as possible until you hear some musical noise (random tones), and then raise the threshold by 2-3 dB so the tones disappear.

The Amount control is the most important for adjusting the tradeoff between noise reduction and processing artifacts. When Amount is set to 0 dB, there is no noise reduction and the output signal should exactly match the input. Increasing the Amount increases the amount of noise reduction, but also increases the possibility of processing artifacts. Possible processing artifacts include:

- smearing of attack transients,
- harmonic tones gating on and off, causing a watery effect,
- musical noise during quite portions,
- loss of signal content, particularly at high frequencies.

MR Noise is designed to minimize artifacts and give you really great noise reduction performance. But you may encounter difficult input signals that can reveal artifacts under heavy noise reduction. Using moderate Amounts, say 6-12 dB, can leave a bit of noise in your recordings, but this may be more desirable than trying to fully suppress noise and causing artifacts.

Another useful technique when processing extremely noisy recordings is to do multiple passes. Instead of doing one pass with a large amount of noise reduction, do several passes using gentle noise reduction settings. This technique will usually result in fewer bad sounding artifacts.

Sometimes a large Amount value is needed to fully suppress noise during quiet portions. However, it may be better to focus instead on what sounds best when the music is playing. Suppressing noise during quiet portions can also be done using a basic noise gate, such as MR Gate.

Adjusting attack and release times

For transient input sounds with sharp attacks you will want to use short attack and release times to preserve the transients. For slower changing legato sounds, you want to use long attack and release times to prevent harmonics from gating on and off rapidly as they decay near the noise floor. If the attacks seem to be smoothed out by the noise reduction, try using a faster attack time. If you hear individual harmonics gating on and off as they decay, try using a longer release time.

Auto dynamics mode

MR Noise features an AUTO dynamics mode that adjusts the attack and release times based upon transients in the input signal. When AUTO mode is enabled, MR Noise detects transients in the input signal and automatically uses short attack time during rapid onset events and uses short release times during rapid offset events. When there are no transients, MR Noise uses the attack and release times set by the user for maximum smoothness during slowly changing passages. We recommend enabling AUTO mode at all times.

Monitoring the removed signal

Enabling the MONITOR feature allows you to hear the signal that is being removed by the noise reduction processor. Ideally this would contain only noise, but in practice it is impossible to perfectly separate the unwanted noise from the desired signal, hence the monitor signal will contain some residue from your desired signal, usually at high frequencies and during transients. Listening to the monitor can be useful for adjusting the various parameters to optimize the noise reduction performance. Adjusting the Amount parameter directly controls the amount of noise signal you will hear in the monitor. Increasing the Amount also increases the amount of residue from the desired signal. You should also be able to hear the effect of adjusting the attack and release parameters. Slower attack and release times allow more of the transients to leak into the noise monitor. Using faster attack and release times (or enabling the AUTO mode) retains more of the desired signal during transients, so that less transient signal leaks into the noise monitor signal. If the monitor signal contains a lot of low or mid frequencies from the desired signal, you should consider adjusting the frequency scaling parameters to prevent these frequencies from being removed as noise. Try reducing the amount and threshold parameters at these frequencies.

Although the monitor feature is useful, you should base your decisions on the sound of the processed signal, comparing this to the original sound when the plug is bypassed. Listening to the noise monitor by itself ignores the fact that many low level details of the noise reduction process are masked by the desired signal. Thus, details in the noise monitor signal may be inaudible in the processed signal.

Spectrum display

MR Noise features a spectrum display which is shown below.

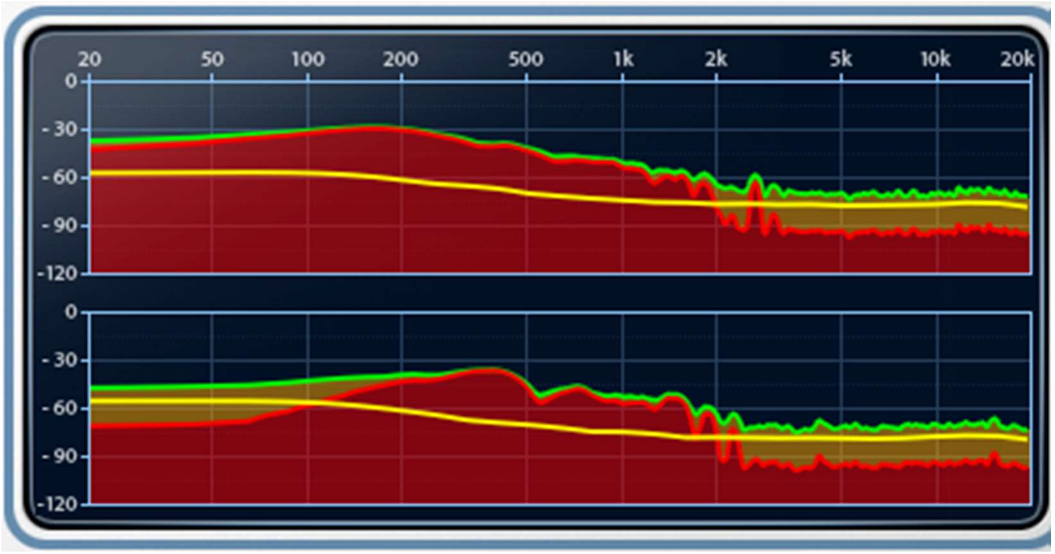


Figure 6-5. Noise reduction spectrum display. Green is input signal spectrum, red is output signal spectrum, and yellow is the noise floor spectrum. The orange region shows where noise reduction is occurring. For stereo inputs, the left channel is displayed on top, and the right channel is displayed below.

When processing a stereo signal, the display is divided into two sections; the top display is the left channel and the bottom display is the right channel. For greater detail you can look at just one channel by using the DISP MODE control. DISP MODE has values of "Stereo", "Left", and "Right". Use the Left or Right values to select just the left and right channels for display. When processing mono signals the DISP MODE control will show "Mono" and the display will show the single channel.

In the display, the input signal spectrum is shown in green, the output signal spectrum is shown in red, and the noise floor spectrum is shown in yellow. The difference between the input and output is drawn with orange, this is where noise reduction is occurring. The input and output spectra are shown only when the MR Noise is enabled and the plug-in is processing audio.

Green	Input signal spectrum
Red	Output signal spectrum
Yellow	Noise floor spectrum
Orange	Region of noise reduction

MR Noise uses a fairly short spectral analysis time, under 20 milliseconds, and consequently the spectrum display does not have good resolution at low frequencies. This is seen in the spectrum display as a very smooth curve at low frequencies. The choice of using a short analysis time was made to optimize the audible noise reduction performance, without regard to display resolution.

The vertical range of the spectrum display can be changed by clicking on the vertical axis; this will show a popup menu with the choices of vertical range: 40 dB, 80 dB, 120 dB, or 160 dB.

Editing the noise floor

It may be impossible to find a portion of sound that contains only background noise. In this case, the noise floor can be learned from a portion of your sound that has the lowest signal level, and then you can manually adjust the floor by editing. To edit the noise floor, click on the FLOOR button.

Figure 6-6. Clicking the floor button allows the noise floor to be edited manually. Click the floor button again to turn off editing.

When the FLOOR button is lit, a set of 31 control points are drawn on each of the stereo noise floor curves. The curves can then be modified by clicking and dragging in the display. After editing you may want to save a preset to save your noise floor edits. Click the floor button again to turn off editing. When the FLOOR button is lit, clicking RESET will reset the noise floor to -160 dB at all frequencies.

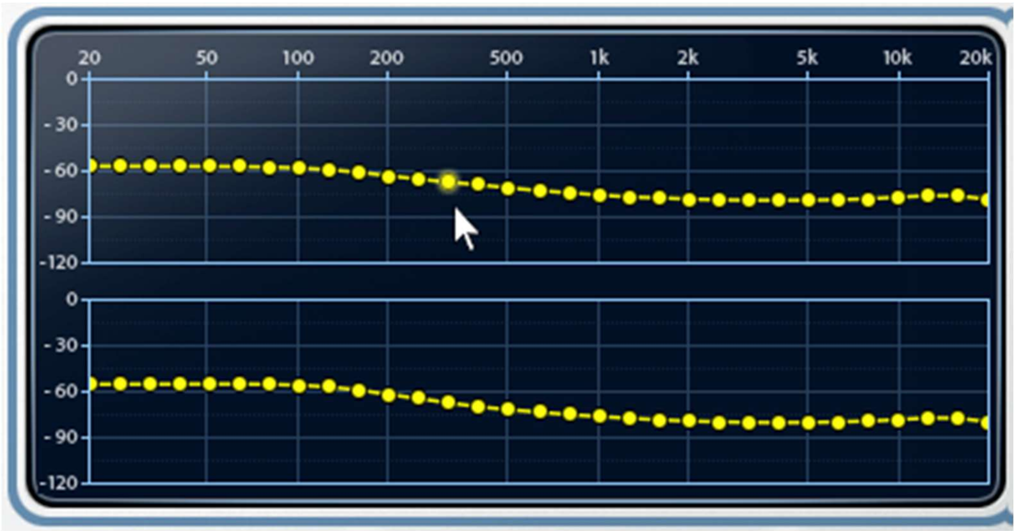


Figure 6-7. Editing the noise floor. Click and drag to draw a new floor.

Editing the frequency scaling curves

For challenging noise reduction problems, it may be useful to specify different threshold, amount, attack time, and release time parameters at different frequencies. This can be done by scaling the parameters as a function of frequency. MR Noise allows the user to specify a scaling curve for each of the above parameters. To edit the scaling curve, click on the corresponding edit button. The figure below shows the Amount button selected.

Figure 6-8. The Amount parameter is selected for editing as a function of frequency.

When one of the parameter buttons is selected, a blue scaling curve is drawn on the display. The scaling curve is specified using 11 control points at frequencies from 20 Hz to 20 kHz such that each point spans roughly one octave. Each control point can scale the parameter from 0% to 200% of the parameter's nominal value. The parameter value at any frequency is determined by the product of the value shown on the knob times the value of the scaling curve at that frequency. The figure below shows an example scaling curve. Although the display is stereo, the scaling curves are monophonic; the scaling parameters are the same for each channel. If you click RESET when editing a scaling curve, the curve will be reset to x1 at all frequencies.

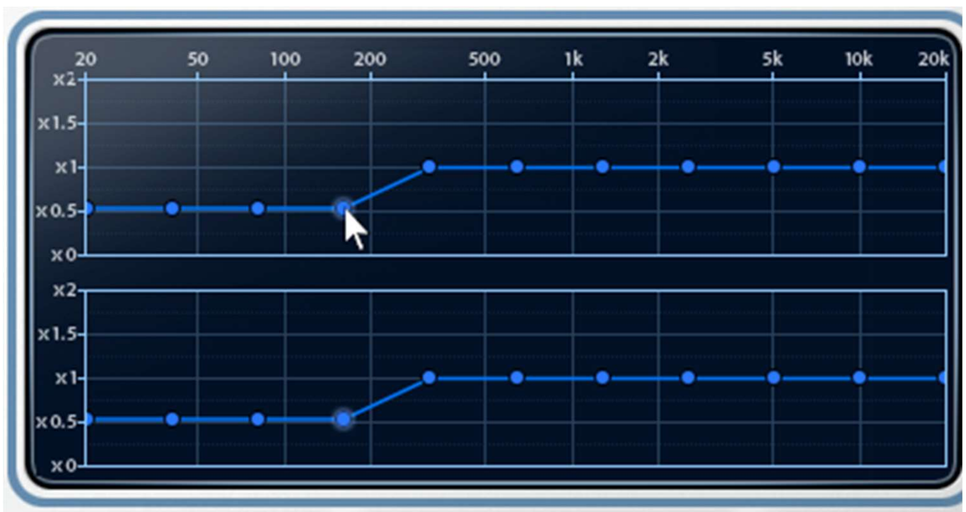


Figure 6-9. Example scaling curve for the Amount parameter. Frequencies below 160 Hz are at 50% value. If the Amount knob was set to 12 dB, then Amount would be 6 dB up to 160 Hz, and 12 dB above 320 Hz.

6.4 Parameters

The table below lists all the internal parameters of MR Noise and shows the range of values as would be displayed by a generic parameter-value plug-in interface, or by an automation interface. Most of these parameters have a one to one correspondence with controls on the user interface.

Parameter name	Values
Enable	0 = Off, 1 = On
Monitor Enable	0 = Off, 1 = On
Threshold	0 – 40 dB

Amount	0 – 60 dB
Attack Time	0 – 100 msec
Release Time	0 – 1000 msec
Knee Type	0 = Soft, 1 = Medium, 2 = Hard
Learn Enable	0 = Off, 1 = On
Learn Time	0 = 50 msec, 1 = 100 msec, 2 = 200 msec, 3 = 500 msec, 4 = 1 sec, 5 = 2 sec, 6 = 5 sec, 7 = manual
Auto Mode	0 = Off, 1 = On
Gain	-12 dB to +12 dB
Monitor Gain	-12 dB to +12 dB
Left Floor 0-30	-160 to 0 dB
Right Floor 0-30	-160 to 0 dB
Thresh Scale 0-10	0 to 2
Amount Scale 0-10	0 to 2
Attack Scale 0-10	0 to 2
Release Scale 0-10	0 to 2

The parameters are described in more detail below.

Enable — Enables or bypasses the noise reduction processing.

Monitor Enable — This enables/disables the monitor feature. When monitoring is enabled, the output signal consists of the noise signal that is removed by MR Noise.

Threshold — Sets the amount the noise floor is offset to start signal attenuation at higher levels than the noise floor. The threshold raises the knee point above the noise floor.

Amount — Sets the amount of noise reduction in dB. This is the maximum amount of attenuation for signals that are at or below the noise floor level.

Attack Time — This parameter sets the attack time constant of the dynamics processor for each frequency, that is, how fast a signal can increase at each frequency. A time of 0 means attacks are followed as quickly as possible. Longer time constants cause smoothing of attacks.

Release Time — This parameter sets the release time constant of the dynamics processor for each frequency, that is, how fast a signal can decay at each frequency. A time of 0 means releases are followed as quickly as possible. Longer time constants cause smoothing of decays.

Knee Type — The Knee determines how fast the attenuation is applied as the signal approaches the noise floor. The Soft knee applies attenuation slowly, Medium is a sharper transition, and Hard is the hardest transition.

Learn Enable — Enables learning. When learning is enabled, the input signal spectrum is averaged to form the noise floor spectrum.

Learn Time — Sets the learn time from 50 msec to 5 seconds. Learning will automatically stop after the learn time period. Manual learning means the learn enable must be turned off manually to stop learning.

Auto Mode — When enabled, Auto mode overrides the attack and release times with automatically determined times based on a broadband transient analysis. MR Noise will use fast attacks during rapid onsets and will use fast releases during rapid offset events. During slowly changing signals without transients, the usual attack and release times are used for maximum smoothness.

Gain — Output gain in dB.

Monitor Gain — Monitor output gain in dB.

Left Floor – Noise floor levels in dB for the left channel. There are 31 floor parameters ranging from 20 Hz to 20 kHz in roughly 1/3-octave intervals.

Right Floor - Noise floor levels in dB for the right channel. There are 31 floor parameters ranging from 20 Hz to 20 kHz in roughly 1/3-octave intervals.

Thresh Scale – Frequency scaling values for the Threshold parameter ranging from x0 to x2. There are 11 scaling parameters ranging from 20 Hz to 20 kHz in roughly one-octave intervals.

Amount Scale – Frequency scaling values for the Amount parameter ranging from x0 to x2. There are 11 scaling parameters ranging from 20 Hz to 20 kHz in roughly one-octave intervals.

Attack Scale – Frequency scaling values for the Attack Time parameter ranging from x0 to x2. There are 11 scaling parameters ranging from 20 Hz to 20 kHz in roughly one-octave intervals.

Release Scale – Frequency scaling values for the Release Time parameter ranging from x0 to x2. There are 11 scaling parameters ranging from 20 Hz to 20 kHz in roughly one-octave intervals.

6.5 Presets

Following is the list of presets that ship with MR Noise.

Name	**Description**
Default	Default settings, same as Clean
Clean	Default settings with learn enabled
Light Clean	Light cleaning settings with learn enabled
Vinyl	Vinyl phonograph settings with reduced threshold and amount at low frequencies, learn enabled

Tape	Magnetic tape settings, harder knee than Clean, learn enabled.
LP33 Floor	Example vinyl phonograph noise floor
Cassette Floor	Example cassette tape noise floor

6.6 Specifications

Description	Frequency domain noise reduction processor
Operating Systems	Windows 10; Mac OS X 10.11 or higher
Plug-in Formats	VST3, AU, AAX
Sampling Rates	up to 192 kHz, supported rates listed below
Latency	20 milliseconds or less, depending on sampling rate, listed below
I/O Formats	mono-mono, stereo-stereo

MR Noise is designed to work at the sampling rates listed in the table below. When processing audio at some other sampling rate, the closest sampling rate in the table below will be chosen. The table also lists the processing latency in samples and milliseconds for each sampling rate.

Sampling rate	Latency, samples	Latency, msec
8000	160	20
11025	160	14.5
16000	320	20
22050	320	14.5
32000	640	20
44100	640	14.5
48000	640	13.3
88200	1280	14.5
96000	1280	13.3
176400	2560	14.5
192000	2560	13.3

7. MR Gate

Figure 7-1. MR Gate user interface.

7.1 Overview

MR Gate is a full featured expander/gate that can be used to gate noise during quiet sections. Some of MR Gate's key features are:

- Adjustable lookahead
- Comprehensive metering
- Dynamic display shows recent history of input and gain levels
- Monitor allows you to hear gated signal

A diagram of MR Gate's audio routing and meter placement is shown below.

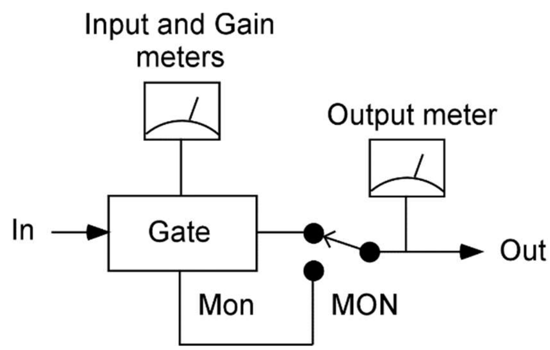


Figure 7-2. MR Gate audio routing diagram.

In normal processing, the gate mutes the audio during silent portions. When the monitor feature is enabled, the output consists of the low level signal that is removed by the gate.

7.2 About Gating

A noise gate is a conceptually simple device that passes high level signals and attenuates low level signals. This is useful for muting the noise that can be heard in the quiet sections between loud parts. A gate monitors the level of the input signal, and if the level falls below a predetermined threshold, the signal is attenuated. The threshold is set to be just above the level of the background noise.

The amount of attenuation applied depends on the ratio control. The definition of ratio is somewhat confusing. A ratio of 1 means no attenuation is applied as signals fall below threshold, while a ratio of infinity means the signal is maximally attenuated. For other ratios R, the attenuation is (R-1) times the difference between the input signal and the threshold. So for example, if the ratio is 5 and the signal is 10 dB below threshold, then 40 dB of attenuation will be applied. MR Gate has a maximum ratio of 50 which for practical purposes can be considered infinite. Because MR Gate has a ratio control it is more properly considered to be an "expander" than a gate.

The attack and release time parameters control how fast gain changes are applied. The release time controls how fast the gain is turned down and the attack time controls how fast the gain is restored. Consider the example of a drum recording containing background noise, assuming the threshold has been set just above the background noise and the ratio is large. When the signal goes below threshold the gate kicks in and begins reducing gain. A short release time will decrease gain rapidly, abruptly cutting off the decay of the drum. Longer release times cause the gain to decrease more slowly, which may sound more natural but also allow the noise to be audible at the end of the drum decay. On the next drum hit, the gate will restore gain according to the attack time. Using a short attack time is prudent in this case, otherwise the attack of the drum will be lost due to the slow attack fade-in of the gate. So, when gating, the attack and release times correspond to the attack and release times of the instrument you are processing.

Another important parameter for gating is the lookahead time. Lookahead allows the gate to respond to events in the signal before they occur. This is done by delaying the audio signal after level detection and before gain processing. Lookahead is particularly important for restoring gain just before onset (attack) events. A few milliseconds of lookahead is sufficient to allow the gain to ramp up before the attack is processed by the gain stage. Running a gate without lookahead risks cutting off all attack events.

MR Gate Input and Gain meters

MR Gate displays an input meter on the left and a gain reduction meter on the right. The input meter shows the peak signal level coming into the gate and the gain reduction meter shows the amount of gain reduction being applied.

Figure 7-3. MR Gate input meter (left) and gain reduction meter (right).

The input meter has a triangular control that lets you set the input threshold level. Drag the control up and down to change the threshold.

MR Gate display

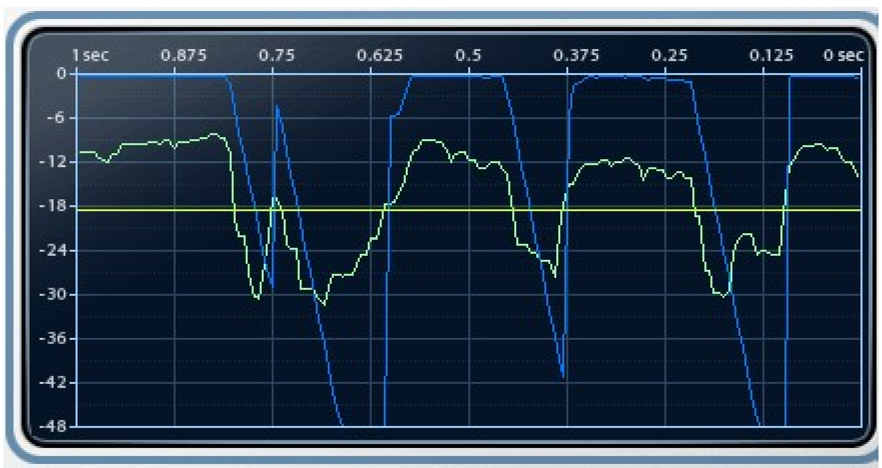


Figure 7-4. MR Gate time display. The green plot is the input signal level, the blue plot is the gate gain, and the yellow line is the threshold.

MR Gate features a dynamic display that plots signal level and the gate gain. The MR Gate display is shown above. Signal level is plotted in green, the gate gain is plotted in blue, and the threshold level is plotted as a horizontal yellow line. The display is mono for either mono or stereo inputs. This is because when processing stereo inputs, MR Gate applies the same gain to both channels based on peak level calculated across both channels. The display is only active if MR Gate is enabled.

The display plots the last few seconds. As processing continues, the display contents scroll from right to left. The vertical amplitude axis range can be changed by clicking on the vertical axis; this will display a popup menu of choices for vertical axis range: 24 dB, 48 dB, or 96 dB. Similarly the horizontal time axis range can be changed by clicking on the horizontal axis; this will display a popup menu of choices for horizontal axis range: 1 sec, 2 sec, or 4 sec.

7.3 Using MR Gate

A good place to start is to select the “Fast Gate” preset. If you are using the Master Restoration plug-in, click on the DISPLAY button to bring up the MR Gate display. Now start playback and adjust the threshold to be just above the level of the background noise during quiet sections. After the threshold is set, you may want to adjust the release time to get the most natural sounding decays into silence. For best results you want to match the release time of the gate with the release time of the sound you are gating.

If the noise gating is too obtrusive, you may want to reduce the attenuation during quiet sections. This can be done by using lower ratio values.

Selecting the MONITOR button allows you to hear the sound that has been removed by the gate. Typically this will contain background noise, and perhaps some reverberant decay tail from the sound you are gating. If you hear the leading portions of sound onsets, you may want to increase the lookahead time to prevent these from being gated.

7.4 Parameters

The table below lists all the internal parameters of MR Gate and shows the range of values as would be displayed by a generic parameter-value plug-in interface, or by an automation interface. Most of these parameters have a one to one correspondence with controls on the user interface.

Parameter name	Values
Enable	0 = Off, 1 = On
Monitor Enable	0 = Off, 1 = On
Ratio	1 to 50 (1:1 to 1:50)
Threshold	-72 to 0 dB
Attack	0.1 to 1000 msec
Release	1 to 2000 msec
Lookahead	0 = off, 1 = 1 msec, 2 = 2 msec, 3 = 5 msec
Knee	0 = Soft, 1 = Medium, 2 = Hard
Gain	-12 to +12 dB
Monitor Gain	-12 to +12 dB

The parameters are described in more detail below.

Enable - Enables/bypasses MR Gate. In the standalone MR Gate plug-in, this parameter is implemented with the bypass control. In the Master Restoration plug-in, this parameter is implemented with the ENABLE button.

Monitor Enable - Enables/disables monitoring. When monitoring is enabled, the output signal contains only the signal that is being removed (gated).

Ratio - Controls the amount of attenuation. Higher ratios cause more attenuation as the signal falls below threshold.

Threshold - Input signals are attenuated when they fall below the threshold level.

Attack Time - Sets the time it takes to increase gain.

Release Time - Sets the time is takes to decrease gain.

Lookahead - Sets the lookahead time. Lookahead is useful when gating to provide a bit more time for the gate to restore gain prior to an onset.

Knee - Sets the dynamics knee shape; options are soft, medium and hard.

Gain — Output gain in dB.

Monitor Gain — Monitor output gain in dB.

7.5 Presets

Following is the list of presets that ship with MR Gate.

Name	**Description**
Default	Default settings with MR Gate disabled
Fast Gate	Fast release gate
Slow Release Gate	Slow release gate
Medium Gate	Small ratio downward expander, use for medium ducking during silence
Mild Gate	Very small ratio downward expander, use for mild ducking during silence

7.6 Specifications

Description	Expander/gate
Operating Systems	Windows 10; Mac OS X 10.11 or higher
Plug-in Formats	VST3, AU, AAX
Sampling Rates	up to 192 kHz
Latency	0-5 msec depending on lookahead setting
I/O Formats	mono-mono, stereo-stereo

8. Master Restoration

Figure 8-1. Master Restoration user interface.

8.1 Overview

Master Restoration combines the functions of MR Hum, MR Click, MR Noise, and MR Gate into one plug-in. The audio routing diagram for Master Restoration is shown below.

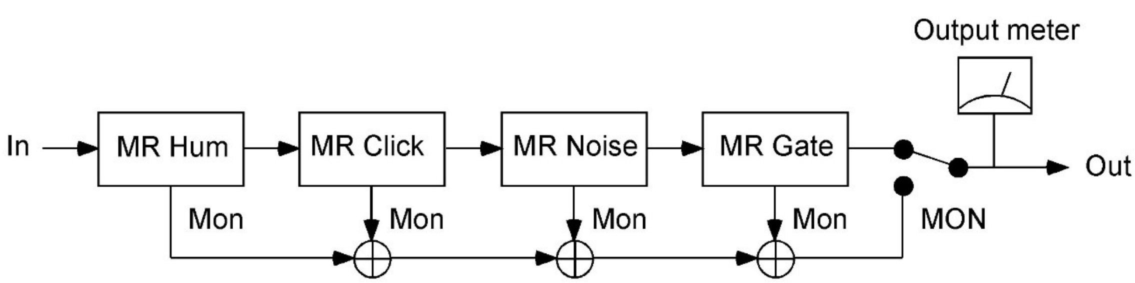


Figure 8-2. Master Restoration routing diagram.

The signal processing order is MR Hum, MR Click, MR Noise, and MR Gate. When any of the processors is monitored, the output consists of a mix of all the monitor outputs from the processors. Hence, you can monitor any combination of the processors to hear a mix of all the signals that have been removed by the individual processors.

8.2 Using Master Restoration

The individual processors are described in detail in the preceding chapters, and the descriptions will not be repeated here. This section will describe some issues specific to using Master Restoration.

Switching the display.

The processors share the same common display area. There are three ways to switch the active display:

1. Click on the tabs at the bottom of the display,
2. Click on the DISPLAY button in each section
3. Click on the panel in each section.

Monitoring sections

Each section has a MONITOR button which can be selected to hear the signal that is removed by the processor. In Master Restoration, it is possible to select multiple sections to be monitored, in which case the output contains a mix of the monitored sections. Master Restoration keeps has separate gains for monitored and non-monitored output. Hence, when monitoring, you can increase the output gain control to more easily hear the monitor signal. When monitoring is turned off, the normal output gain is applied.

Section presets

In Master Restoration, each section has a preset control as shown below. Click on the button to advance to the next section preset (SHIFT-click moves backwards to the preceding preset). Click on the text area to display a popup menu of preset options. There are rolloff menus for factory and user presets, and Save and Save As options for saving a section preset. This menu works like the main preset menu in the menu bar, which is described in Chapter 3.

Figure 8-3. Section preset control.

Sharing presets between processors

User presets created in the various processor sections of Master Restoration can be imported into the corresponding standalone plug-in, and vice-versa. Click Preset->Import and select the preset file of the plug that created the preset you want. Navigate to the user preset file. The location of the preset files is discussed in section 3.8 User Presets. After importing the preset file, the user presets will appear in the global preset menu of the individual plugs when importing from Master Restoration, and will appear in the section preset menus of Master Restoration when importing from the individual plug-ins.

8.3 Presets

Master Restoration factory presets are listed in the table below.

Name	**Description**
Default	Default settings with nothing enabled
Tape clean	Magnetic tape
Vinyl clean	Vinyl phonograph

8.4 Specifications

Description	Hum, buzz, click, crackle, and broadband noise removal with gating, brickwall filters, and spectrum analysis
Operating Systems	Windows 10; Mac OS X 10.11 or higher
Plug-in Formats	VST3, AU, AAX
Sampling Rates	up to 192 kHz
Latency	Sum of latency of individual sections. See specifications for individual plug-ins.
I/O Formats	mono-mono, stereo-stereo

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