



T-RackS 5

USER MANUAL

T-RackS 5 is available in different versions.

The difference between the different versions is in the number of gear models included and in the features available.

While consulting this manual, please refer to the specific gear models and features available in the T-RackS 5 version you are using.

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Via dell'Industria 46
41122 Modena
Italy

What's New in T-RackS 5

T-RackS 5 is the complete redesign and improved version of the all in-one-mastering software known for its high standards, clarity and performances.



The GUI is redesigned and resizable from a minimum of 1330x670 px (non-retina/high density display value) to 4K resolution and has now 8 dedicated parts to make your workflow faster and easier.

The Mastering Chain is now improved, making possible to have 16 modules per session (see computer minimum requirements).

New modules are added: ONE module, EQual module, DynaMu module and the MasterMatch Module.

A new Assembly window lets you edit and create industry standard audio montage, and the improved export possibilities now include DDP 2.5 and Wav Cue and PQ sheet.

Improved and even more precise metering suite with separate floating window, for an even more accurate visual reference, the best in its category.

Chapter 1 – What is T-RackS 5

1.1 - Introduction

IK Multimedia pioneered analog-modeled mastering with the first all-in-one software solution, the original standalone version of T-RackS, that became a cult hit in studios around the world.

T-RackS 5 raises the bar again in DAW mastering and mixing with a suite of analog-modeled and digital processor plug-ins with many new features and unrivalled audio quality.

T-RackS 5 processors include high-end analog and digital EQs, compressors, limiters, reverbs, delays and saturation units, that offer amazing sonic performance with a similar working environment to that of professional mastering stations - right on your desktop - both as a plug-in for all the most popular DAWs or as a standalone application.

1.2 - T-RackS 5 Features

- Modular mastering/mixing suite of dynamics and EQ processors including: Configurable mastering/mixing chain – runs up to 16 parallel/serial processors
- Complete built-in metering section with Peak, Phase and RMS meters plus RTA analyzer, Spectrogram, VU and PPM meters, Dynamic Range and Perceived Loudness with standards and genres suggestion, all bold following the ITU-R BS1770-4.
- High-fidelity oversampling with high transparency for extremely high-quality audio processing throughout the entire signal path.
- Standalone and plug-in versions available in the same package for use in all the most popular DAWs.
- SCC™ (Sonic Character Cloning) technology coupled with IK's unique DSM™ (Dynamic Saturation Modeling) technology to provide the most realistic software emulation of vintage gear to date .
- Extremely easy-to-use interface with style-based presets, full chain visualization, one-click single module or chain bypass, Equal Gain function, multiple audio file processing in standalone mode and much more.
- Native compatibility with 64-bit applications and operating systems.

1.3 - The Custom Shop

T-RackS 5 includes the Custom Shop functionality as its predecessor, which allows you to log into a virtual online shop full of powerful models which you can choose from. This modular system lets you build your personal collection of gear that you can tailor to your specific mixing/mastering needs.

1.3.1 - Entering the Custom Shop

The construction of your custom setup starts by simply installing the T-RackS 5 CS Shell which comes with Classic EQ and the simple Metering modules included as a startup base package (the modules also work as single plug-ins within DAW environments) and is available as a free download from the IK Multimedia web site upon registration.

Just like T-RackS CS (hence the 4th version of the software), it works for both the plug-in and the standalone version.

The first thing you have to do is access the Custom Shop by clicking the “CART” button.



You can also launch the Custom Shop application from each single module within a DAW.

NOTE: you **MUST** be connected to the internet to do this as all of the following operations done in the Custom Shop are web-based.

1.3.2 - Gear Credits, the currency of the Custom Shop

The Custom Shop is based on a dual currency system: regular currency and credits. In order to buy credits, click the “ADD CREDITS” button in the main browser window. You will find Credit Packs of several sizes.

Welcome IK User, you have 1180 credits left

AMPLITUDE T-RACKS SAMPLETANK

Search...

ADD CREDITS GEAR ACCOUNT HELP

Buy Gear Credits

Buy Gear Credits whenever you need them. Use your Gear Credits to unlock Sound Libraries, AmpliTube and T-RackS gear models or Collections. Gear Credits are used to purchase Sound Libraries, AmpliTube and T-RackS gear models or Collections but cannot be used to purchase Hardware or Books.

Gear Credits	Price	Per Credit Cost	Buy Credits
25	\$ 24.99	as low as \$ 1 per credit	BUY CREDITS
50	\$ 49.99	as low as \$ 1 per credit	BUY CREDITS
75	\$ 74.99	as low as \$ 1 per credit	BUY CREDITS
100	\$ 99.99	as low as \$ 1 per credit	BUY CREDITS
150	\$ 134.99	as low as \$ 0.9 per credit	BUY CREDITS
200	\$ 159.99	as low as \$ 0.8 per credit	BUY CREDITS
300	\$ 224.99	as low as \$ 0.75 per credit	BUY CREDITS
500	\$ 349.99	as low as \$ 0.7 per credit	BUY CREDITS
1000	\$ 549.99	as low as \$ 0.55 per credit	BUY CREDITS
1500	\$ 749.99	as low as \$ 0.5 per credit	BUY CREDITS

Top Seller

1. Black 76 Limiting Amplifier
2. Mic Room
3. Opto Compressor
4. Classic T-RackS Clipper
5. Tape Echo
6. Precision Comp / Limiter
7. EQ PG
8. CSR Room Reverb
9. EQ PA
10. CSR Inverse Reverb

Edit Billing and Shipping information, then click PROCEED TO CHECKOUT.

You can then choose the preferred payment method. The first one is a standard credit card procedure where you enter your personal data. You can also choose to use PayPal to buy Credit Packs. In this case, you will

be redirected to the PayPal main page to login with your PayPal account info. If you don't have a PayPal account, you will have the opportunity to create one by following the on-screen instructions. Once the checkout procedure is complete, your total credits will be displayed in the browser's main window. Now you can go back to the Custom Shop main page and start choosing your gear!

If you have purchased a Credit Pack outside of the Custom Shop (from the IK Multimedia online store, from a dealer, through a promotion, etc.), you must enter the Serial Number on the ADD CREDITS page of the Custom Shop. Click ADD CREDITS, then enter your serial number at the bottom of the page in the field below "Redeem Credit Pack Serial Number." Now click the REDEEM CREDITS button to add the Credit Pack to your account .

Redeem Credit Pack Serial Number

If you have purchased a Credit Pack outside of the Custom Shop (from the IK Multimedia online store, a dealer, through a promotion, etc.) enter the Serial Number here, and then click the Redeem Credits button.

Serial Number:

REDEEM CREDITS

1.3.3 - Buying Gear

As you can see, each of the available pieces of gear listed in the various categories (EQ, Dynamics, Reverbs and so on) have their own price shown in credits or regular currency, together with a description of the item. You can now select and buy the modules you want limited only by the total amount of credits in your profile (in case you decide to pay with credits).

Buying gear is simple: just click on the BUY button next to the item you want and choose your payment method, then confirm your purchase in the following window.

Welcome IK User, you have 1180 credits left

AMPLITUDE T-RACKS SAMPLETANK

Home > Dynamics 19 gear models available

	Saturator X Multi-Mode Harmonic Saturation Module 80 Credits	\$79.99	TRY	BUY
	Stealth Limiter Ultra-transparent mastering inter-sample peak limiter 125 Credits	\$124.99	TRY	BUY
	Black 76 Limiting Amplifier FET Compressor/Limiter 100 Credits	\$99.99	TRY	BUY
	Brickwall Limiter Intelligent, Multi-Algorithm Limiting 80 Credits	\$79.99	TRY	BUY

Top Seller

- Black 76 Limiting Amplifier
- Mic Room
- Opto Compressor
- Classic T-RackS Clipper
- Tape Echo
- Precision Comp /

EQ

DYNAMICS

REVERB

METERING

DELAY / ECHO

HARMONICS

OTHER

COLLECTIONS

AC. CORRECTION

In case you chose to pay with credits, the item's price in will be subtracted from your total shown in the main browser's window. The module you just purchased will be immediately available within T-RackS. It will also be marked as "purchased" in the MY GEAR section of the Custom Shop with the date of purchase.

All the modules (as well as the collections) can also be purchased on the IK online store.

1.3.4 - Trying Gear

You can also choose to try out a piece of gear (clicking the TRY button) to audition it in your existing rig before actually buying it. Every item in the Custom Shop has a 14 days unrestricted trial time. When an item is selected for trial, the time left for its evaluation is shown in the corresponding item menu.

Once the trial time has expired, the item will be locked – it will be grayed out. Once the trial time has expired the module will only be purchasable on the Custom Shop or IK online store.

NOTE: multiple instances of the same item will not affect the total time of the trial period. The counter starts with the first one. Modules can be tried within the T-RackS 5 suite as well as singles in third party DAWs.

1.3.5 - Restore Purchased Gear

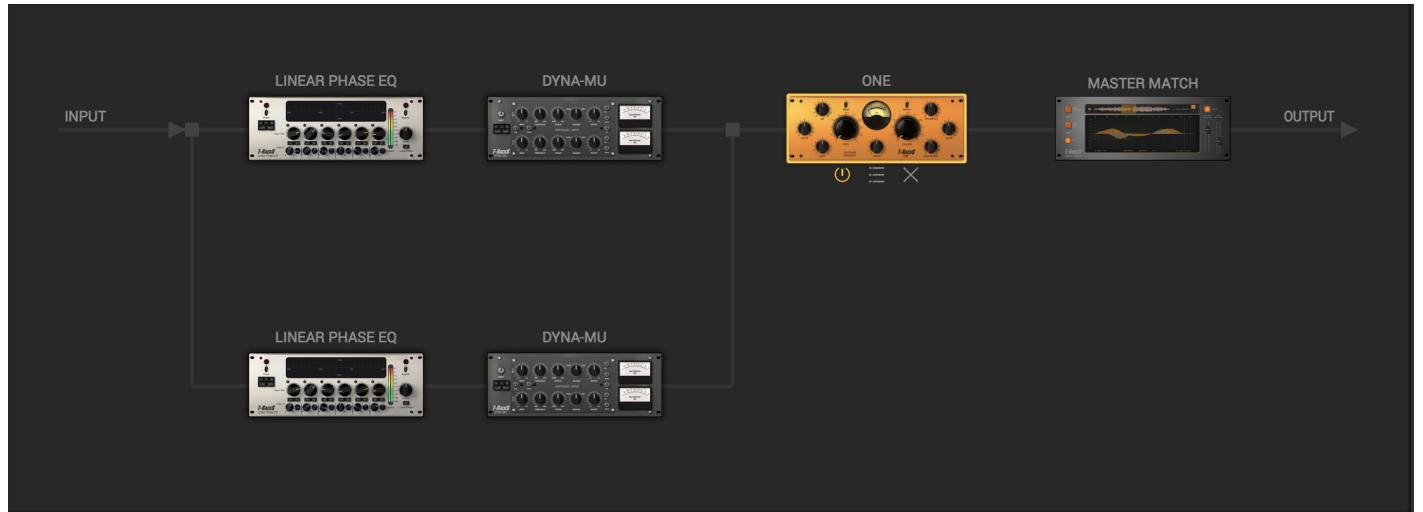
If you have upgraded your computer hardware or OS, or if you are working with a dual setup (desktop, laptop), you may need to have T-RackS Custom Shop installed on more than a single machine.

Since the Custom Shop is web-based, it's easy to recover all your purchased gear and get it back online ready for use in your session. Just click the "RESTORE MY GEAR" button, and all of your gear will be loaded back into the T-RackS plug-in (again, an internet connection is necessary for this as the Custom Shop is web-based).



1.4 - 16 processing slot mastering chain

Audio chain can be filled with up to 16 selectable processing modules that can be arranged with extreme flexibility in both parallel and series.



A module is insert in the chain by drag and drop from the Module Browser or by double click on its icon; the audio path follows the positioning of the module in the chain.

When a module is insert, it is possible to move it and change its path position in both parallel and series; also it is possible to substitute a module when selected, by pressing ALT and Click on the module icon you want to insert in the Module Browser.

It is possible to bypass, cancel and browse the single module preset with fast icons that appears when the mouse is positioned over the module in the chain, or when the module is selected.

The Master Match is the only module which has a fixed position in the chain; that is because it's purpose is to fine match your work with a reference tracks and finalize it, or to help you visualize possible errors in your chain or easily get inspired by really complex equalization suggestion.

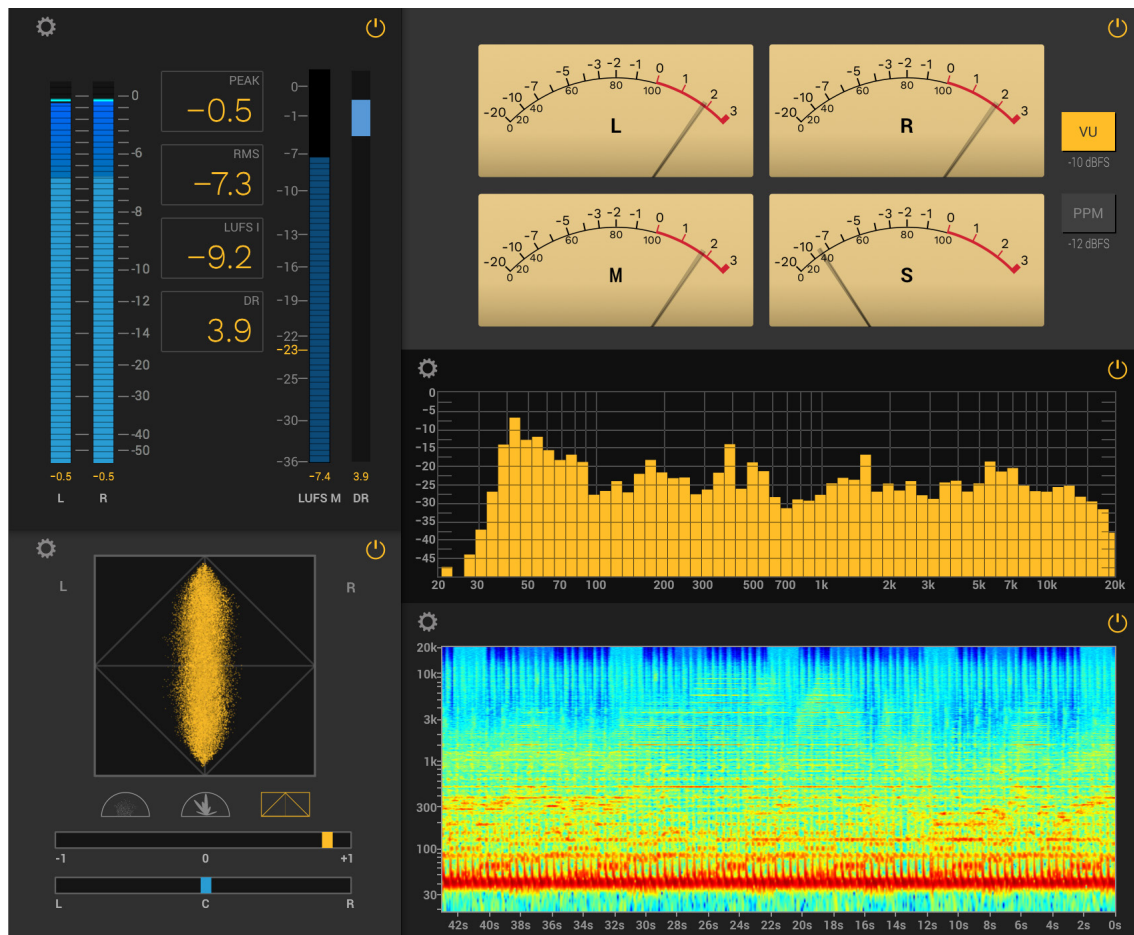
The Master Match can be bypassed or removed from the audio path with the fast icon below it.

“Equal Gain” function allows comparison between the sound of the processed version and the original with like all the most famous mastering engineers are used to - decrease the listening volume of the mastered version

1.5 - Accurate, complete and truly useful metering

T-RackS 5 includes a high-quality, high-precision, complete metering section that includes:

- A precision Peak Meter with accurate sample indicators and real digital “Over” indication.
- RMS meter.
- Phase Scope with options to match the style and indication you’re used to.
- Phase Correlation.
- RTA analyzer with Peak, RMS and Averaging indicators.
- Loudness and Dynamic Range metering with reference levels and targets indication selectable by standard and genre. We also added suggestion and references for the new standards in music and broadcasting, in order to have your records ready for the present and future industry.
- VU and PPM meters with adjustable reference level.
- Spectrogram.
- Spectrum analyzer.



1.6 - High transparency, extremely high quality audio processing

You may think digital processors are transparent in general and that all of them are high enough quality for mixing and mastering professional audio. That's not always true. Very simple processing like a filter or a dynamics effect change can literally ruin your sound beyond rescue if it is not performed at the highest possible quality by applying techniques that are not so commonly used.

T-RackS 5 products deliver incredible purity, transparency and fidelity far superior to the vast majority of plug-ins you can compare it to. If you are familiar with the sound of top-quality analog and digital hardware limiters that are used daily in top mastering houses, you'll immediately recognize here with T-RackS 5 the quality and purity typical of those units. If you're not familiar with the hardware, you'll experience what high-quality dynamics processing means.

Many digital processors use oversampling to achieve better sound. Unfortunately, many of them – under critical listening sessions – achieve no significantly better (or even slightly worse) results than the regular sampling rate option.

This is because the upsampling and downsampling processes are not done at a sufficient level of quality. In other words, typically what is improved by running a processor at double sampling frequency is partially compromised by the loss of quality in the upsampling and downsampling process.

There are only a few pieces of gear, mainly hardware – and, until now, all of them very expensive – that have really realized oversampling the way it should be done. T-RackS 5 is one of them. We wanted to offer the same quality of the highest caliber mastering tools, and the result is a process that has no side effects at all on the sound: total transparency.

So what you hear is exactly the same sound of the processor with NO side effects.

1.7 - SCC™ technology

With over 15 years of experience in analog-modeled software, IK has pioneered this technology and accumulated more experience than any other plug-in manufacturer in the field. This has resulted in several improvements like our unique DSM™ (Dynamic Saturation Modeling) and now the new SCC™ (Sonic Character Cloning). SCC™ has been developed to “clone” the true behaviour of an analog vintage processor down to its finest level of sonic detail for a faithful reproduction of all its tonal characteristics. This new emulation technology produces results that are indistinguishable from the originals, even for the professionally trained ear.

Many pieces of vintage gear tend to have original and interesting circuit topology, due to the creative solutions that engineers were sometimes forced to put into place in order to overcome the limitations of components' technology.

Vintage component technology (many components can't be found anymore because their technology became obsolete and uneconomical) as well as their aging (some of the best sounding devices are 40-50 years old) played fundamental roles in giving these devices a unique musicality that is what the purist would define to be “unmatched” by even the most advanced processors today.

And finally, many of the unique sonic characteristics of these magnificent vintage units sometimes depended on erratic and unpredictable behaviour of their electronics that would be considered unorthodox and unusable by today's science and standards.

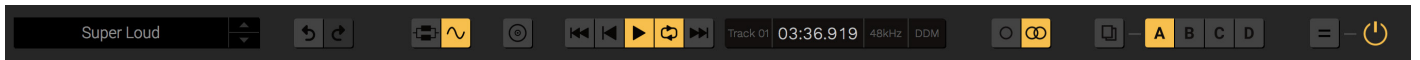
SCC™ technology has the ability, through a new analysis method developed by IK, to integrate all of the above and turn it into mathematical algorithm that, for the first time, is able to take into consideration all of these complex variables and get an exact “clone” of the original; software emulation evolved to a sound quality never heard before. The results are so close to the original units that you will not be able to tell the difference.

1.8 - Interface

T-RackS 5 interface has 8 separate areas, positioned and designed to give you a fast and easy workflow:

Top Bar

In the top bar is it possible to: fast browse presets, undo and redo actions, select between chain and waveform view and open the Assembly window, using all the transport commands (when in standalone mode), select between mono and stereo mode, compare between 4 different chains, Copy and paste those chains between clips and activate the Equal gain output possibility and bypass the entire chain.

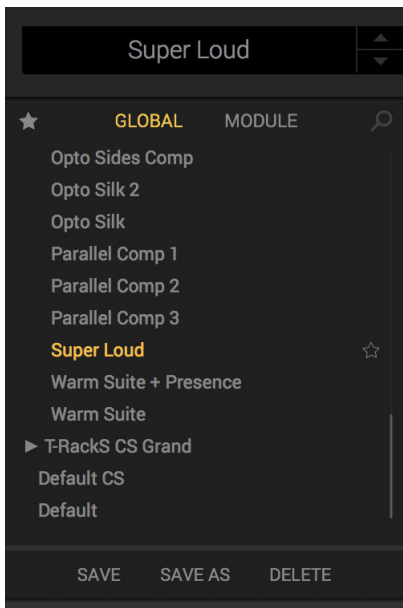


Presets Browser

This area, in the upper left, is used to save and load T-RackS presets.

Presets can be saved/loaded as GLOBAL (the entire mastering chain) or by MODULE (selected module processor).

It is also possible to mark a preset as favorite for fast recall: press the Star icon to filter the preset by the favourite option.



A search engine is also available: start writing the name of your preset and it will automatically filter the presets by the name.

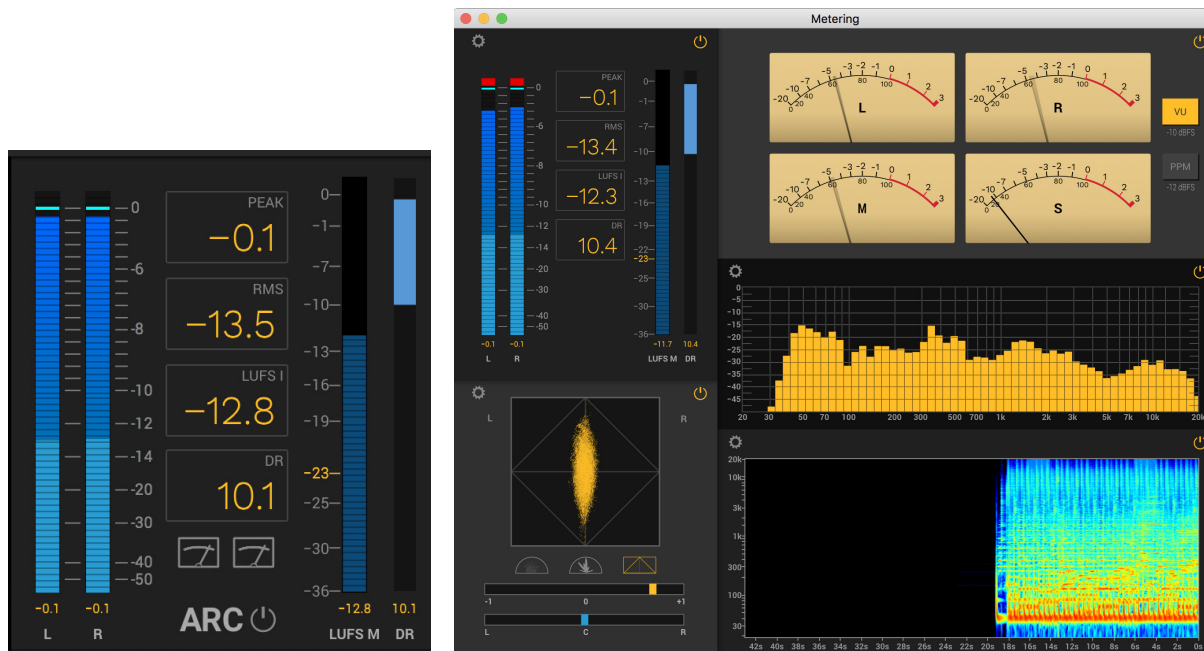
Processor Interface

This area shows the selected processor with all its controls.



Metering section

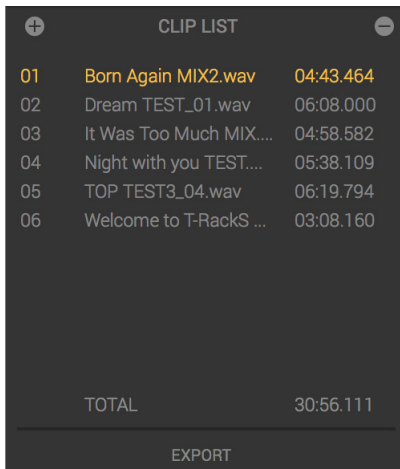
All T-RackS 5 meters are conveniently placed in this area and are always visible to keep the most important sonic signal variables under control. This section can be opened in a separate floating window by pressing the related icon, right above the ARC button.



Also, it is possible to access and activate the ARC correction, to focus on mastering without guessing.

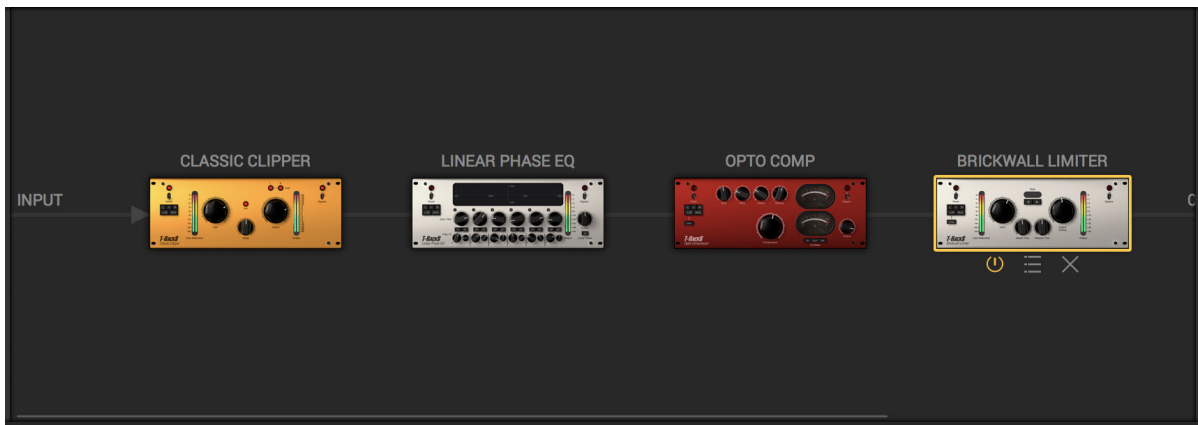
Cliplist

This panel shows the loaded audio clip in the standalone, showing the length or total length when multiple tracks are loaded. Simply press the Plus or Minus icon to add or remove clips.



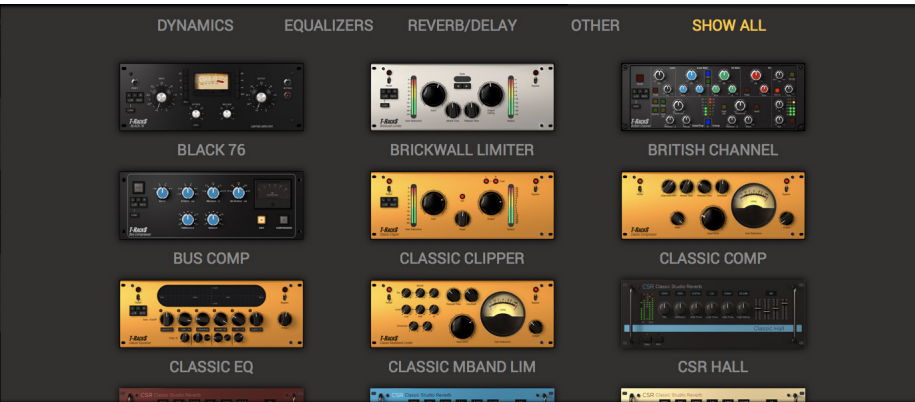
Chain View/Waveform view

In this panel, right below the module section, the module chain or the waveform are editable.



Module Browser

In this section you may select any of the modules. You can drag and drop these into the chain, or double-click to load into the chain. There is also a Filter to help search by type.



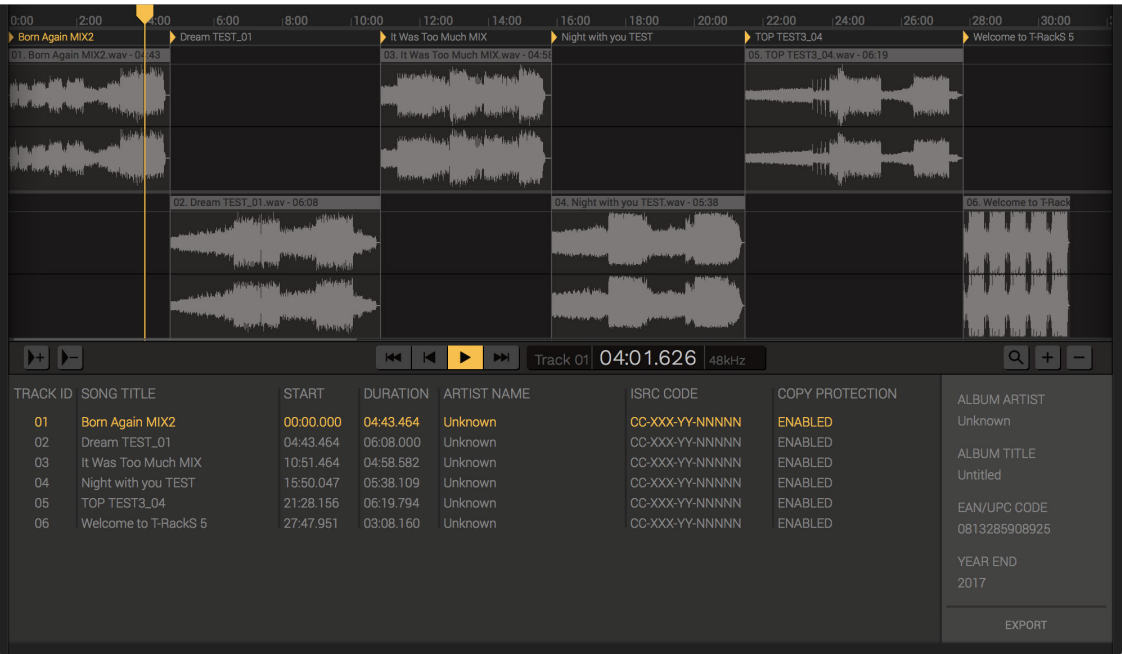
Bottom Bar

Shows the buttons to manage software authorization, information, custom shop, preferences and automation settings when in the plug-in version.



Assembly window

This area is present only in the standalone version of T-RackS 5 and allows you to load and edit multiple audio files to be processed with T-RackS 5 and exported for CD burning or to other media.



Each file's waveform is displayed on the top, and various basic editing features are available such as FADE IN/FADE OUT.

Once loaded, every track will have its own track ID and start snapshot, and the Track ID can be modified in the assembly to create the correct metadata of your track.

It is also possible to insert other Track ID markers to easily export audio montage in the audio standard quality/specifics.

Chapter 2 – Getting started

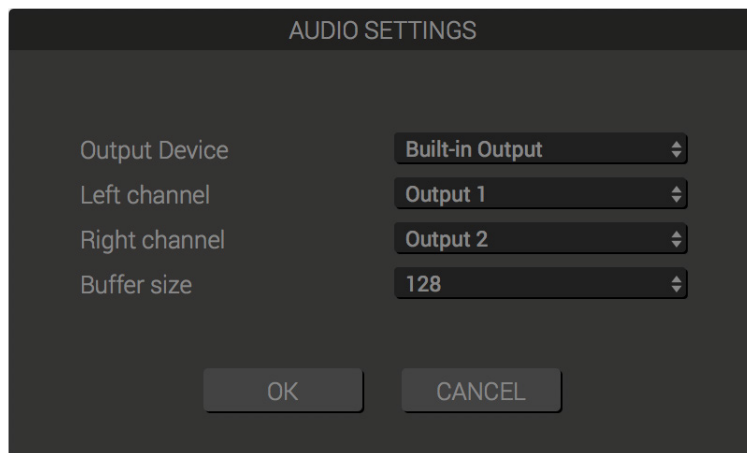
T-RackS 5 can be used as a standalone application or as a plug-in; also all the single modules can be used as single plugins in your DAW.

2.1 - Using T-RackS 5 as a Standalone Application

2.1.1 - Audio Hardware Settings window

To use T-RackS 5 in standalone mode, you need to set up the audio interface used by the software for its stereo audio output.

This window can be found under the T-RackS menu in the Audio Setup section, in Menu bar of the standalone application. Here you will select the audio interface and buffer size that will be used in T-RackS 5.



Audio Driver: T-RackS CS is compatible with any ASIO or DirectX audio interface in Windows and any Core Audio compatible audio interface in Mac OS X.

Windows

T-RackS 5 is compatible with ASIO or DirectX compatible audio interfaces on Windows. Use the drop-down Technology and Output Device menus to select your audio interface.

Left/Right Channel pop-up menus are used to route the T-RackS 5 stereo output to specific audio outputs on the chosen audio interface.

Buffer Size allows you to change the buffer size of the audio interface. Click on Panel button to change ASIO settings for your audio interface, like buffer size, sync option etc.

Mac OS X

T-RackS 5 is compatible with Core Audio audio interfaces in Mac OS X. Please select your audio interface from the Output Device drop down menu.

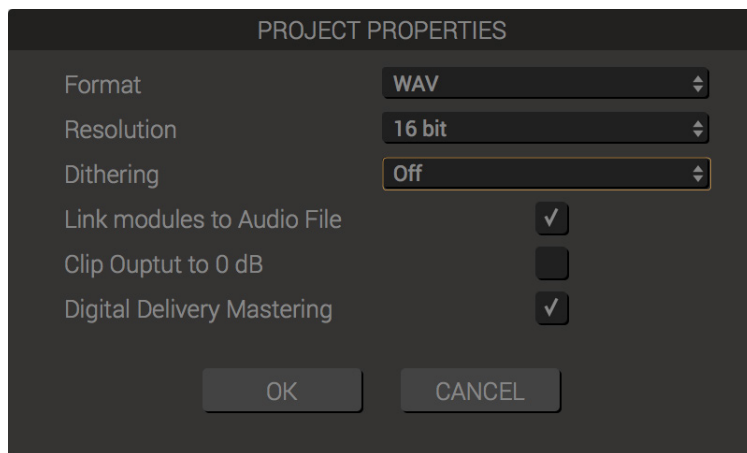
Left/Right Channel pop-up menus are used to route the T-RackS 5 stereo output to specific audio outputs on the chosen audio interface.

Buffer Size menu allows you to change the buffer size of the audio interface. These settings are dependent on the driver of the audio interface itself.

Project Properties window

The Project Properties window (File > Project Properties) allows you to adjust the T-RackS 5 Project resolution and sample rate as well as enabling the Digital Delivery Mastering (DDM).

These settings are found also when a new project is created.



2.1.2 - Loading and playing files

To load audio files into the T-RackS 5 standalone application, simply drag them over the Interface or use the (PLUS) button to open the system dialog.

The audio files will appear in the CLIPLIST panel with their relative duration.

By clicking on the file, it will be selected, and its chain and waveform will be shown in the relative views.

By Double clicking on the file it starts playback.

Pressing PLAY, a PLAYHEAD will be displayed over the waveform with the COUNTER display indicating its position in the file in minutes/seconds/frames format.

Press PLAY again to stop playback.

Both in PLAY or STOP mode you can skip to a different points of the clip by clicking on the desired position on the waveform.

To loop a part of the file, simply select the desired part with the mouse, and the LOOP will automatically turn on. To turn it off use its button on the transport on the TOP BAR or double click on the selected part of the timeline. The loop can be delete by closing it on itself.



2.1.3 - Inserting processors

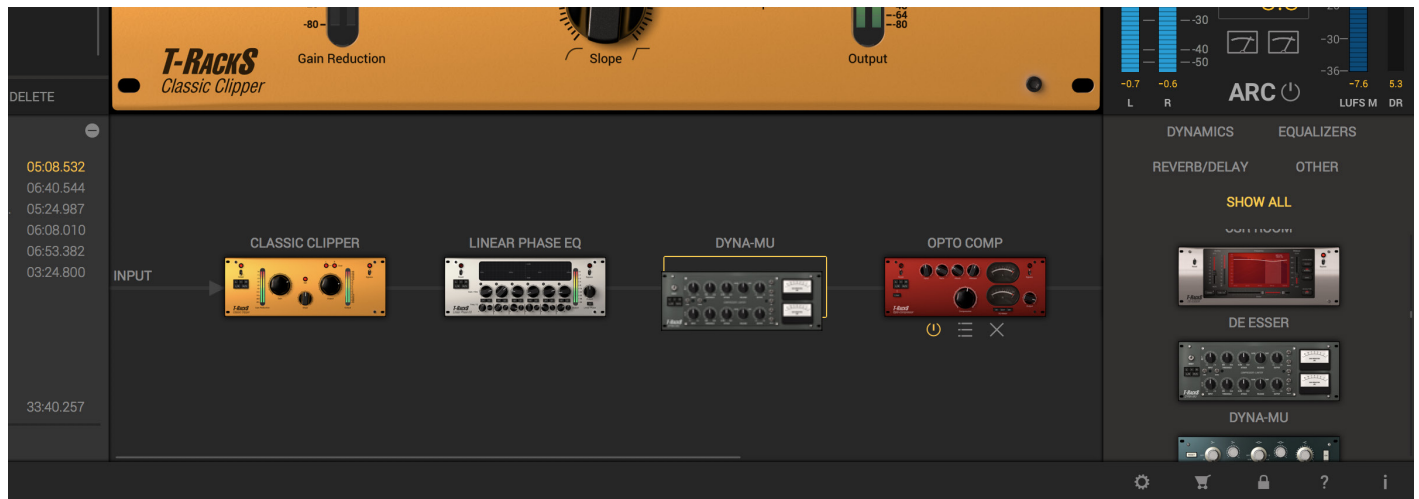
To start mastering your files you will need to insert T-RackS processors in the audio chain for the selected file.

To do this, simply search for the desired processor in the Module Browser and drag and drop it in the mastering chain, or double click on it to load it at the end of the chain.

It is possible to open the module related preset clicking the icon below the module: the preset browser will show the relative presets.

Now the audio will be played through the T-RackS processor inserted in the chain.

To insert more processors, you can simply repeat the operation up to 16 times.



Now you can edit every control of the processors to achieve the desired mastering results and check your results on the T-RackS metering section.

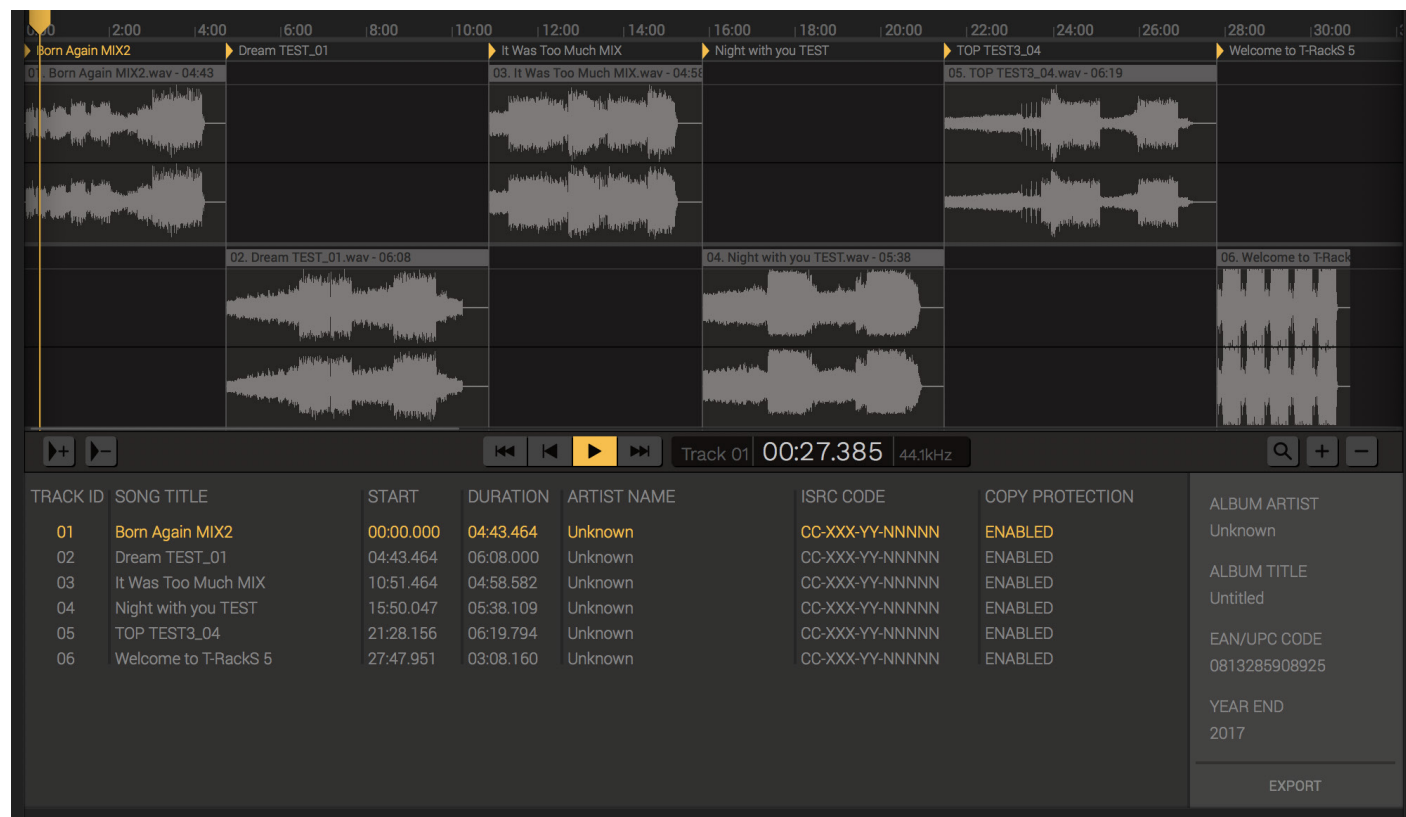
By selecting a different audio file you can insert a totally different chain, or copy a chain from a Clip to another, you can use the ABCD chain compare buttons, which can be used to compare different chains.

Within the same audio file you can apply different T-RackS settings by using the available SNAPSHOTS.

2.1.4 - Edit your files: Assembly window

When you are satisfied with the results of your processing, you can edit all the information of the files or your album or song.

To open the assembly window, press the “ASSEMBLY” icon on the Top Bar, and a floating window will appear.



The assembly window is divided in:

Waveform/Album Edit

Displays files in chronological order. You can graphically edit the files, track IDs, pauses, and crossfades.

The transport function and behaviors remain the same of the other window.

Edits made to the audio file are not destructive, so you can always revert to the original, if necessary.

Files are alternately displayed in the upper or lower Region lanes in the Assembly, making it easier to view any overlapping areas between two regions to create perfect crossfades.

Once a file is added to your project, it is automatically assigned a Track ID marker, which contains the track number displayed by a CD player during playback.

This Track ID marker is mirrored by a newly created CD track that uses the new files, and appears in the Cliplist.

Please keep in mind that the Clips and the Track ID are not completely related: The Track ID are metadata position that allows the creation of industry standard files as DDP and Wav CUE, while a clip is the audio file itself.

A Track ID is linked to the clip until the track ID or the clip are edited otherwise, for example deleting the track ID. Doing so, it is possible to have clips with more than 1 Track ID and the other way around: this allows you to edit and create a live CD from the 1 file recorded audio directly in the assembly.

To select a file, simply click on it or on the Track ID on the tracklist; to rename the track ID, double-click on the name onto the Tracklist. To delete the file, select and press DELETE on your keyboard or right-click and select DELETE.

To reorder a track, vertically drag the track name in the tracklist.

To adjust the position of a track in the timeline, without affecting the following tracks, hold Option before selecting and dragging the track.

Middle Tool Bar

Here its located the transport bar and the editing tools, such as marker creation or delete, zoom and export.

Tracklist

List the files in chronological order and it is possible to rename and reordered in the tracks list..

It is possible to edit all the info relative to the file and the album, in order to be inserted in the WAV cue file and the DDP file.

2.1.5 - T-RackS 5 Project

T-RackS 5 standalone version can save its current state, a list of loaded audio files, T-RackS processing chain settings, snapshots, fades and other settings into a Project file.

When you launch T-RackS 5 standalone, an “untitled” project is loaded by default. You can start to load audio files and tweak the T-RackS 5 audio processing chain in this Project.

To save your Project so that it can be loaded later on, select Save Project from File menu.

As you have been working on a untitled Project, you will be asked for a file name and location on your hard drive. To continue working on this Project later on, select Open Project... and locate the file you previously saved (T-RackS 5 projects have a “.tr5prj” extension).

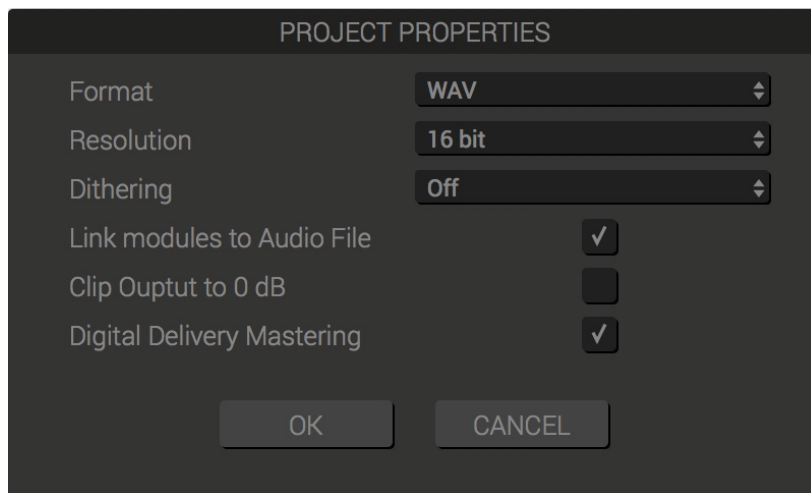
The T-RackS 5 default Project has the following properties:

Output File Format: WAVE

Output Sampling Rate: 44.1 kHz

Output Resolution: 16 bit

DDM: OFF



The T-RackS audio processing chain settings will be linked to the selected Audio File.

To create a new project, simply select New Project in the file menu, and a new Project window will appear allowing you to specify a Project name and set all Project properties.

To save a copy of the Project with a different name and / or to a different location, select Save As... from File menu.

2.2 - Using T-RackS as a Plug-in

In addition to the standalone application, T-RackS also works as a plug-in effect. T-RackS plug-in functions will be exactly the same as the standalone version apart from the file player/editor that is available only in standalone mode. Please refer to your DAW’s user manual on how to instantiate and operate T-RackS inside your session.

Chapter 3 – General controls

3.1 - Flexible Chain

T-RackS 5 Flexible Chain allows you to insert up to 16 parallel/serial processors, including the Master Match module, which is fixed at the end of the chain, and will be bypassed by default.

You can insert any of the available T-RackS processors into the remaining 15 slots. To load a processor, simply drag and drop it from the modules browser or double click on it.

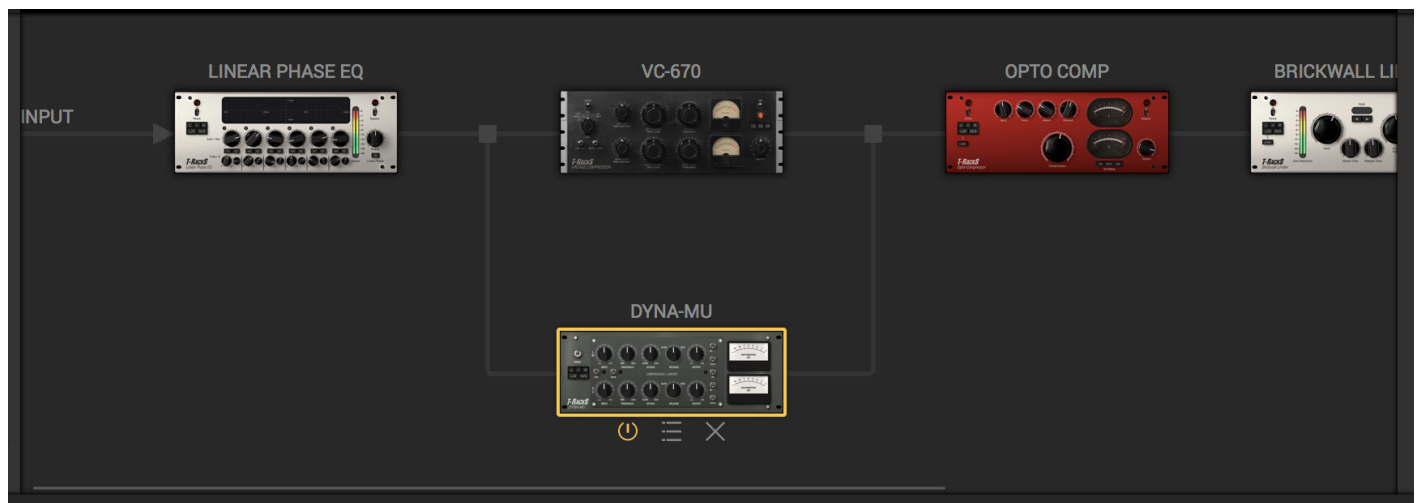
The dragged module will then become selected and the relative 3 function icons (bypass, preset, remove) will appear..

The module is in serial if dragged it the upper half of the panel, and in parallel if dragged in the lower half of the panel.

The modules in parallel can be moved in the chain at will, and also the chain nodes can be used to move the position of the parallel signal start and end.

A module in the chain can be swapped with a module in the browser by selecting it and Alt-Click on the one in the browser.

It is possible to change the position of a loaded module at anytime with a max 16 modules at a time.

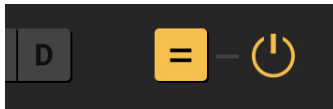


3.2 - Equal Gain Output

In T-RackS 5 it is possible to audition the track both processed by the chain and not processed by chain, at the same perceived level.

We want to give you the experience of a real mastering studio, automatically lowering the listening volume of the processed clip to match the unprocessed file; that is done to help you compare without distortion caused by increasing the unprocessed file to wrong levels, like most other software programs do, and help in discover error that you might not notice at higher levels.

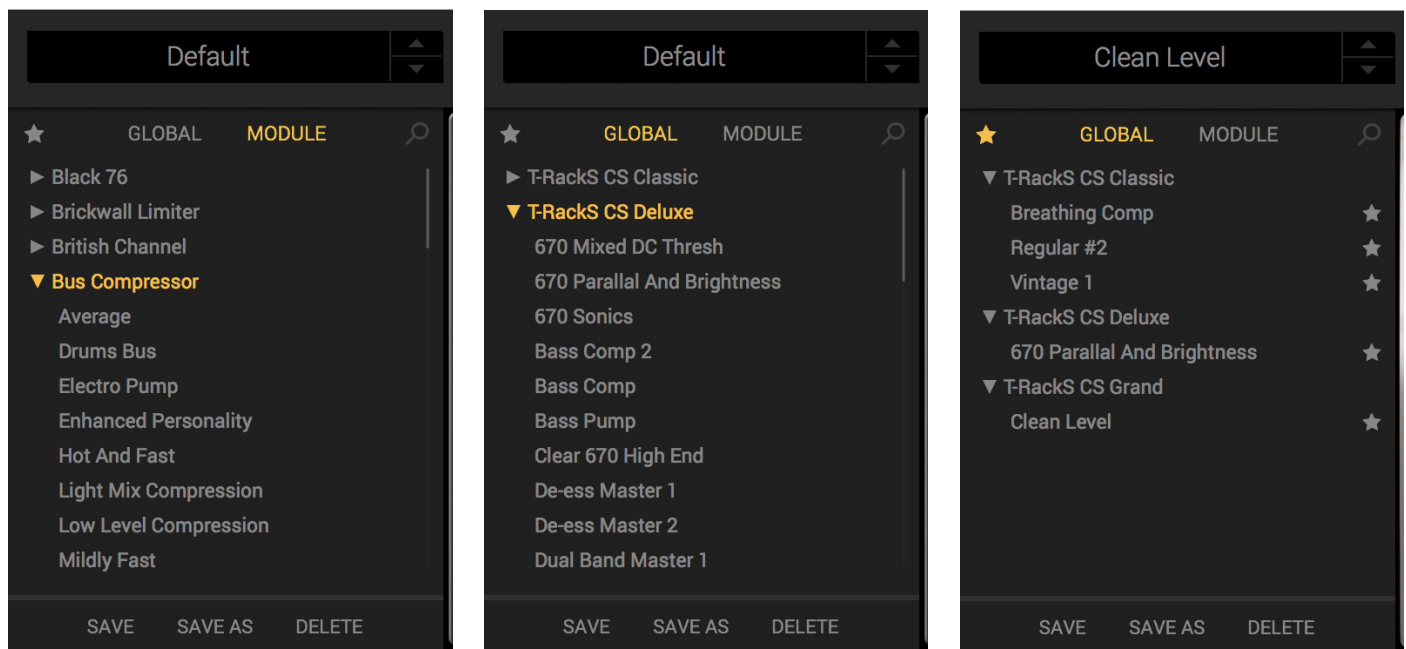
To do so, simply press the "EQUAL" icon on the left of the bypass icon, in the top bar; the software will automatically balance the levels and using the general Bypass to listen the processed or the unprocessed file.



3.3 - Presets

You can save the entire T-RackS configuration or single module processor settings into presets that can be stored, edited and recalled at will.

To load a preset in T-RackS, simply go to the preset browser panel on the left if you are using the standalone application, or press the preset window if you are using the plug in; then select if you want to search a GLOBAL or a MODULE preset, and search between the folders or search by name by selecting the "LENS" icon and typing in the preset name you are looking for.



To load a preset, simply click on the name of the preset.

The SAVE button saves the current T-RackS settings as a preset. Saving a preset with no previously loaded presets shows the Save As window to allow you to give your preset a name.

The SAVE AS button works like the SAVE button presenting the Save window with the name of the current preset displayed in the preset name field, at the same time giving the option to save the preset to another one with a different name.

DELETE preset will delete the currently loaded preset.

The module presets can be opened from the icon that appears below the modules when passing the mouse above it, or when a module is selected. The function Button, such as the Save, Save As.. and the Delete, will not work for the preset modules.

3.3.1 - Presets backward compatibility

If you are upgrading from previous versions of T-RackS (or you have them still installed on your computer), it is possible to both read and load older T-RackS CS presets and they will work perfectly.

However, the old user presets need to be copied manually from the T-RackS folder to the T-RackS 5 folder.

3.4 - A-B-C-D chains

The A-B-C-D chains are 4 different T-RackS chain and processors that can be saved and recalled on the fly while working on the audio.

To copy a chain into another A-B-C-D chain simply click on the COPY TO button. The A-B-C-D buttons will start flashing and clicking on any of them will save the current settings in the chosen one.

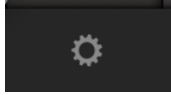
A chain can also be copied in another audio clip with the same method.

You will be able to quickly compare between up to four different configurations without pausing your audio, a function that is particularly useful during fine-tuning sonic details. Please note that these settings are not stored in the T-RackS 5 preset files or DAW session.



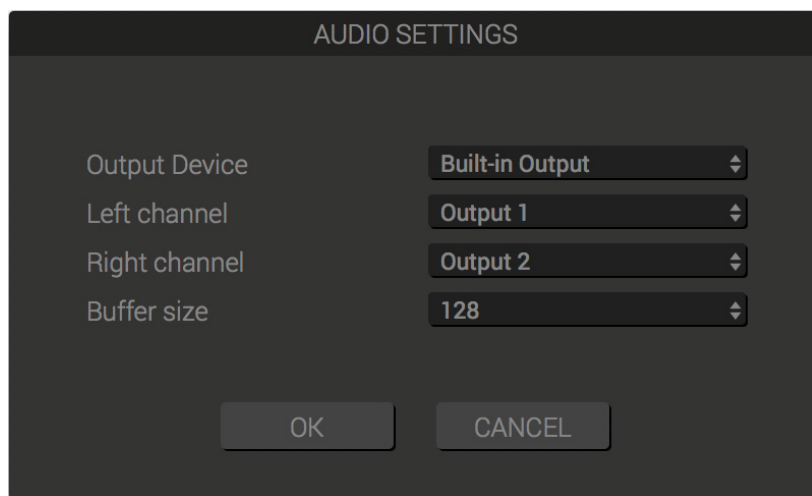
3.5 - Preferences

By clicking on the “GEAR” icon on the bottom bar, the T-RackS audio preferences window will be displayed.



Preferences are also accessible from the File menu.

In this dialog it is possible to adjust the Audio preferences:



Output Device

Select the core audio/asio compatible audio interface you want to use.

Left Channel/Right Channel

Select the physical output, of the selected interface, you want to use for the Left and Right Channel

Buffer Size

Select the buffer size here. A larger buffer size can be used here if you hear clicks or pops during playback.

Clip Output to 0 dB

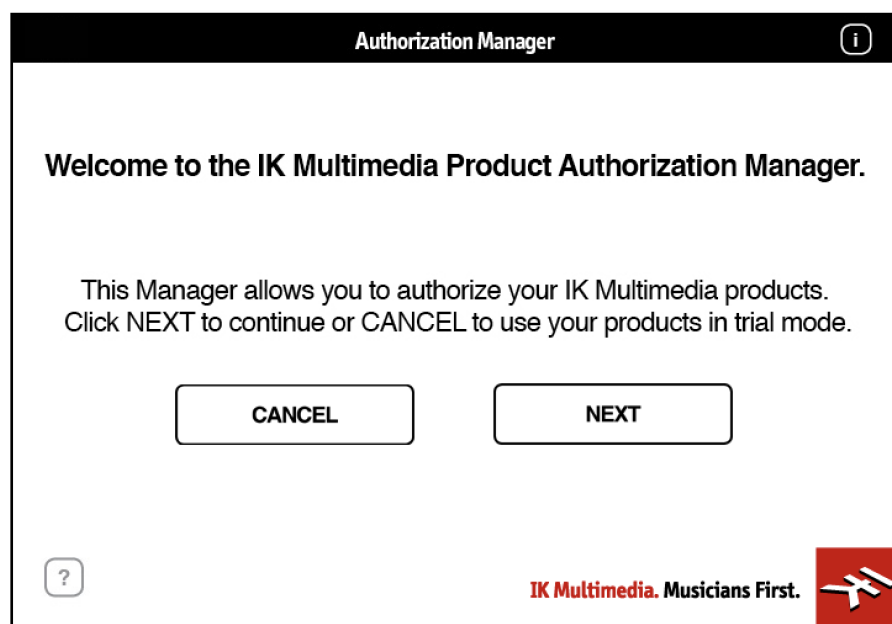
Use this option to have T-RackS 5 to limit at 0 dB the signal at the output of the processing chain by hard clipping. If this is unchecked the signal is allowed to go beyond the 0 dB digital level.

3.6 - Info

By clicking on the INFO button the T-RackS info panel will be displayed. You will find the product name, its version number, and the related copyright notes.

3.7 - Lock

If you are running the software in demo mode, and you click on the Lock button, the Authorization Manager will appear. However, if you already registered and authorized your product, clicking on the Lock button will show you the product Serial Number.



3.8 - General controls (standalone)

When launching T-RackS 5 in standalone, the T-RackS 5 interface will show 2 modules in the chain by default: the Classic Equalizer and the Master Match.

The Classic Equalizer will be selected and shown by default. The preset browser will show the global presets section by default, and the cliplist will be completely empty, read for your files to be imported.



To load audio files in the standalone application. Drag and drop them into the interface or use the ADD (“+”) button. The file loaded will appear in the cliplist, the tracklist, and its waveform will appear in the waveform view.

To select an audio file, simply click on it, and the selected file will be highlighted in orange.

To remove an audio file from the browser select it press the delete button on your keyboard or use the “-” button.



The types of files that can be loaded, played and processed from the standalone version of T-RackS are Wav, Aiff, Caf, Sd2, Mp3. Once a file is loaded you can use the transport buttons on the T-RackS 5 standalone top bar to play, stop and activate/deactivate the loop playback.

A time counter will show the current playing position time. A yellow PLAY CURSOR will be moving on the waveform during play.

You can click anywhere in the waveform to set a different playback position, or you can click and roll on the COUNTER display for the same result.



You can also change your view of the audio file using the zoom functions. By clicking on the LENS, you can select the desired audio portion with your mouse. There will also be +/- buttons in the waveform display to precisely select specific points of the audio file. The scroll bar underneath your wave form will allow to easily scroll your audio file's time line.



When the LOOP play button is enabled, a selection, delimited by an orange stroke. Will appear on the waveform. This loop selection can also be dragged and resize at will using your mouse.

You can apply FADE IN/OUT to your file by positioning the grey FADE IN/OUT handles, just below the timeline, at the extremities of the waveform displayed.

The FADE IN/OUT areas will be highlighted in grey and the waveform will assume the form of the fade, which can be selected on the bottom utility bar of the waveform view.

In the panel you can also select the type of FADE IN/OUT curve. This is done by clicking on the curve icon and selecting here the type you prefer. There will be linear, exponential, and logarithmic curves available.



In T-RackS 5 standalone you can also use snapshots to apply different settings of T-RackS processors during playback. To insert a new settings snapshot, simply change one or multiple parameters of your inserted processor chain, and then click on the PLUS (“+”) button. A snapshot marker will then appear.



By default a TRACK ID and a “Start” SNAPSHOT are always present at the start of the audio file. These are added when the file is loaded in the player/editor. This means when you take a snapshot for the first time this will be listed as SNAP 2.

The SNAPSHOT memorizes and applies the modified settings you have applied to your inserted processor chain. You cannot change the start snapshot’s position. Only the following snapshots can be moved.

You can take multiple different snapshot settings and apply them any time you want over the time line of the audio file

A COPY function is available so you can apply previous SNAPSHOTs in different positions of your audio file (for example to re-insert the same snapshot in the various chorus/verse parts of your song). To do this, simply select the snap you want to copy, then create a new snap where you want and press PASTE to paste the settings.

To delete SNAPSHOTs simply click on the DELETE button in the utility bottom bar after you have selected the snapshot that you want to delete.

If you want to temporarily “suspend” the SNAPSHOT’s automation. Just turn them OFF by clicking on the ON button in the SNAPSHOT. Your previously saved snapshots automation will be restored when you enable this section again.

Chapter 4 - T-RackS 5 module processors

T-RackS CS comes as a free download from the IK Multimedia web site and as a startup base package that includes the Classic EQ and a simple metering.

All other modules can be purchased by entering the Custom Shop (or as singles on IK Multimedia web store) or, in case you already have several modules purchased and registered to your account, they can be retrieved with the “Restore my Gear” procedure (see chapter 1.3.5 for reference).

Following is a description of all available modules.

4.1 - Vintage Tube Compressor/Limiter model 670



This processor is based on the “Holy Grail” of compressors/limiters, the Fairchild® 670. With a faithful reproduction of every control, this is an incredibly accurate model that captures every nuance of one of the best Fairchild units available.

Controls:

- **AGC:** sets the compressor channels mode. Set it to **Left/Right** for unlinked stereo (independent channels) operation. Set it to **Link** for linked stereo operation (both channels will have identical settings except for Input Gain, and the same amount of gain reduction will always be applied to both channels to ensure stereo image stability. Set it to **Lat/Vert** to have the compressor two channels to work on Mid and Side portions of the Stereo signal. With this mode you’ll be able to process independently your Center (Lat) and Sides (Vert) image of the stereo program.
- **LEFT-LAT DC Threshold:** exactly like on the original unit, set it fully counterclockwise for maximum softness of the compression knee; Set it fully clockwise for maximum hardness of the compression knee.
- **RIGHT-VERT:** same as LEFT-LAT DC Threshold for right channel.
- **LEFT-LAT TIME CONSTANT:** sets the compressors release time. Positions 1 from 4 are single time constant, from quick to slow, positions 5 and 6 are dual time constants and are especially useful for mixed stereo material.
- **RIGHT-VERT TIME CONSTANT:** same as LEFT-LAT TIME CONSTANT for right channel.
- **LEFT CHANNEL INPUT GAIN:** sets the compressor input gain on Left channel. When no compression is happening set this knob to 10 for unity I/O gain.
- **RIGHT CHANNEL INPUT GAIN:** same as LEFT CHANNEL INPUT GAIN, for right channel.
- **LEFT-LAT THRESHOLD:** sets the amount of compression. Fully counterclockwise means no compression, fully clockwise give the maximum possible amount of compression for that signal.
- **RIGHT-VERT THRESHOLD:** same as LEFT-LAT THRESHOLD for right channel.
- **IN/OUT/GR METERS:** when set to IN the VU meters will indicate the audio level at the inputs of the

compressor. When set to OUT they will indicate the audio level at the outputs of the compressor. The most important indication is GR (default) where the amount of compression in dB is shown. Always use the GR position when dialing the desired compression amount with THRESHOLD knobs.

- **OUTPUT:** sets the output level within a +/- 15 dB range.
- **BYPASS:** completely bypasses the whole unit.
- **RESET:** brings the compressor back to its default state where a compression of around 2 dB is happening on moderate level mixes.

4.2 - Tube Program Equalizer model EQP-1A



This processor is based on what is universally regarded as one of the best and most musical program EQs ever made, the Pultec® EQP-1A. Extremely accurate modeling of two real world units known to have “the sound” with sonic performance that is indistinguishable from the originals. Stereo or MS operation.

Controls:

- **LOW FREQUENCY:** sets the frequency intervention point for the low shelving boost and atten filters: 20, 30, 60 or 100 Hz. Please note that because the Pultec’s original circuit was based on a passive equalization network the boost and atten filters will not cover the same frequency span and will not cancel one each other if set identically.
- **LOW FREQUENCY BOOST:** applies a low frequencies boost with a shelving shape on the selected frequency range. Set it to 0 for no boost, 10 for max boost.
- **LOW FREQUENCY ATTEN:** applies a low frequencies attenuation with a shelving shape on the selected frequency range. Set it to 0 for no attenuation, 10 for max attenuation.
- **BANDWIDTH:** widens (clockwise position) or narrows (counterclockwise position) the range of frequencies covered by the high frequency bell curve. This control does not affect the sound of the low frequency filter nor the one of the high frequency attenuation.
- **HIGH FREQUENCY:** sets the center frequency of the high frequency bell boost filter: 3, 4, 5, 8, 10, 12, 16 kHz. (The high frequency boost and attenuation filter have independent frequency settings).
- **HIGH FREQUENCY BOOST:** applies high frequency boost with a bell shape, centered on the selected frequency. Set it to 0 for no boost, 10 for max boost.
- **HIGH FREQUENCY ATTEN:** attenuate high frequencies with a shelving shape starting from the selected value on ATTEN SEL. Set it to 0 for no attenuation, 10 for max attenuation.
- **ATTEN SEL:** sets the frequency intervention point for the high frequency attenuation filter: 5, 10 or 20 kHz.
- **OUTPUT:** sets the global equalizer gain, from -15 to + 15 dB.
- **ON/OFF:** completely bypasses the unit when on the OFF position.

- **RESET:** resets the equalizer to its initial, flat, state.
- **L=R:** linked channels mode. Set it this way when you want to control both left and right channels at the same time, by applying the exact same equalization to them.
- **L/R:** click on L or R icons when you want to tweak the equalization independently on the Left or Right channels.
- **M/S:** this option uses a Mid-Side matrix to insert the equalizers on your processing chain. This means that you will be able to equalize independently the Center or the Sides of your stereo program. The L and R icons will be replaced by M and S when in this mode. Click on M to equalize the Center (Mid) of your stereo image, or on S to equalize the Sides.

4.3 - Opto Compressor



Optical compression has a special sonic character that true audiophiles appreciate. This analog-modeled module faithfully recreates that particular (and extremely musical) “movement” that characterizes only the best opto compressors.

Controls:

- **INPUT:** sets the input level of the compressor, from -24 to +24 dB.
- **RATIO:** sets the Ratio of the compressor. Use low values for gentle and invisible mastering applications and higher values for more powerful and obvious compression. Ratio range is from 1:1 (no compression) to 30:1 (almost limiting).
- **ATTACK:** sets the compressor attack time from 0 to 50 ms. Extremely fast attack values could deliver slight distortion on certain sounds.
- **RELEASE:** sets the compressor release time from 30 ms to 5 seconds.
- **COMPRESSION:** sets the amount of compression that is applied to the signal. Fully counterclockwise means no compression, fully clockwise means full compression.
- **IN/OUT/GR METERS:** when set to IN the VU meters will indicate the audio level at the inputs of the compressor. When set to OUT they will indicate the audio level at the outputs of the compressor. The most important indication is GR (default) where the amount of compression in dB is shown. Always use the GR position when dialing in the desired compression amount in with the COMPRESSION knob.
- **OUTPUT:** sets the compressor output level within a range of +/- 24 dB.
- **BYPASS:** completely bypasses the opto compressor module.
- **RESET:** brings the compressor back to its default state where a compression of around 2 dB is happening on moderate level mixes.
- **L=R:** the same settings will be applied to L and R compressors. The L and R compression will be exactly the same if the LINK button is ON, to avoid stereo image shifts.

- **L/R:** click on L or R icons if you want to dial in specific compression setting for L and R channels, independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo compressor on your processing chain. This means that you will be able to compress independently the Center or the Sides of your stereo program. The L and R icons will be replaced by M and S when in this mode. Click on M to set the compression for the Center (Mid) of your stereo image, or on S to set compression for the Sides.

4.4 - Intelligent, multi-algorithm Brickwall Limiter



This Brickwall Limiter can really make your masters loud without squashing them to a lifeless sound. Accurate and transparent, digital brickwall limiting doesn't get any better than this. Selectable algorithms allow you to match the best limiting style for each program type.

IMPORTANT: this is the processor that should be always used at the end of your T-RackS mastering chain because it has special features to keep the final output level of your master under control.

Controls:

- **GAIN REDUCTION:** shows the amount of limiting that is happening in the gain reduction element of the limiter. Limiting obtained by saturation or clipping is NOT shown by this meter.
- **INPUT:** sets the input level for the limiter, and therefore the volume increase the limiter will give.
- **ATTACK TIME:** sets the limiter attack time. Depending on how much limiting you're applying it may be necessary to increase this value to preserve percussive material's sense of impact.
- **RELEASE TIME:** sets the limiter release time. The longer the release time the softer the limiter will be. Short and very short release times will make the limiter more suitable for rock, R&B, pop etc, while longer release times will make the limiter more suitable for music that is generally mastered quieter, like jazz, classic, fusion etc.
- **STYLE:** sets the style the limiter will use. Options are:
 - **Clean:** this is the cleanest and most transparent style, useful for genres where the complete absence of distortion is a must.
 - **Advanced 1, 2, 3, 4:** these styles use a "mix" between warm and smooth saturation and digital limiting. All offer more punch and color than "Clean".
 - **Sat 1, 2, 3:** these styles use various forms of saturation as limiting. The Gain Reduction meter will not show reduction because the gain reduction element is not working on these styles.
 - **Clipping:** straight digital 0 dBFS clipping, no gain reduction shown.

- **OUTPUT CEILING:** sets the top level the audio will never be allowed to exceed. Typically set to -0.1 or 0.2 dB. If “D/A Distortion Protection” is selected in the Preferences this can be left to 0 dB.
- **OUTPUT:** peak indication of the limiter output audio level.
- **BYPASS:** completely bypasses the Brickwall limiter.
- **RESET:** brings the limiter back to its default state.
- **L=R:** click on the L or R icons if you want to tweak the limiter’s L or R channels independently.

4.5 - High precision, high definition Linear Phase Equalizer



This is accurate, high definition, high precision EQ that can be trusted for delicate Mastering tasks with 6 completely identical bands that can be set to all available filter types and that can span the whole spectrum range (any kind of overlap is possible). Switchable between minimum and linear phase characteristics. Stereo or MS operation.

Controls:

- **GAIN/RES 1-6:** sets the gain from -15 to +15 dB when the band is set to Peaking or Shelving. When the band is set to High or Low pass filters this knob will control the resonance.
- **FREQ 1-6:** sets the center or cutoff frequency for the band, all bands can span from 10 Hz to 20 kHz.
- **Q 1-6:** sets the bandwidth of the Peaking filters, Q ranges from 0.2 to 20. When bands are set to Shelving this knob will set the shape of the shelving filter from gentle to steep and resonant, spanning from 0.2 to 3.
- **BAND ON/OFF:** each band can be turned On or Off independently. All bands are Off at startup. When the band is ON the button is green.
- **BAND FILTER TYPE:** each band can be set to work as a Peaking filter (default for bands from 2 to 5), Shelving filters (defaults for bands 1 and 6) and high or low pass filters.



- **OUTPUT:** sets the equalizer output level, from -15 to +15 dB.
- **OUTPUT METER:** peak meter for the equalizer output audio signal.
- **LINEAR PHASE:** sets the phase characteristic of the equalizer. When Off, the equalizer performs with minimum phase characteristics, when On the equalizer is completely linear phase.

- **CURVE DISPLAY:** this display shows the resulting equalization curve for both channels. Individual bands EQ points can be dragged to set Gain and Frequency points without using the equalizer knobs.
- **BYPASS:** completely bypasses the whole equalizer.
- **RESET:** brings the equalizer back to its default, flat, state.
- **L=R:** linked channels mode. Set it this way when you want to control both left and right channels at the same time, by applying the exact same equalization to them.
- **L/R:** click on L or R icons when you want to tweak the equalization independently on the Left or Right channels. Left channel curve will be displayed in green, right channel in red.
- **M/S:** this option uses a Mid-Side matrix to insert the equalizers on your processing chain. This means that you will be able to equalize independently the Center or the Sides of your stereo program. The L and R icons will be replaced by M and S when in this mode. Click on M to equalize the Center (Mid) of your stereo image, or on S to equalize the Sides.

4.6 - Classic T-RackS Compressor



The T-RackS classic tube compressor emulates that classic, analog, hi-end vintage gear used in mastering applications. This will give you that big, warm sound typical of highly acclaimed hardware devices. This compressor has some unique features that make it very flexible but always very musical and creative.

Controls:

- **SIDECHAIN HPF:** this control adds a high pass filter to the detector stage of the compressor. The detector stage is where the compressor analyzes the incoming signal to determine how gain reduction must be applied. This variable high pass filter prevents, when turned up, that compression pumping effect mainly caused by the material's low end content. The higher this knob the softer and more inaudible the compression character will be. At the minimum or at very low settings the compression will be much more aggressive and noticeable. Use lower settings when you want to "hear" the compression effect and use higher values for a gentle compression, as you may want in acoustic mixes.
- **ATTACK TIME:** adjust this control to change the speed of compression in the attack phase. With higher values, transients will pass unaltered through the compressor. With lower values the gain reduction will respond quicker and transients will be more affected. Value range is from 15 ms to 80 ms. While adjusting, check the release time value above the knob to see the attack time value you're setting.
- **RATIO:** use this control to define the strength of the compression. Lower values give a gentler and less noticeable compression effect, more suitable for 2-tracks mixes and acoustic sounds. Higher values give a more effective compression effect with, at times, all the side-effects you might expect from a over-compression process. These effects are sometimes desirable, that's why T-RackS' compressor ratio control sweeps up to a value of 5. If you are not sure which ratio control value is suitable for your needs, start setting this control at 2 - 3.
- **RELEASE TIME:** adjust this control to change the speed of compression in the release phase. Using higher values, the recovery time will be longer and the compression will be less noticeable. Using lower values, the recovery time will be shorter and the average loudness will be higher. Value range is from 70 ms to 1.5 s.
- **INPUT DRIVE:** this knob controls the amount of signal that drives the compressor's input. This is no

threshold point leveller so the amount of compression is determined by the strength of input signal. Range is from -18 dB to +18 dB.

- **STEREO ENHANCEMENT:** this control will affect the stereo imaging of your mix. Turning it up you will increase the stereo width and vice versa. While adjusting this knob check the value you're setting (in dB). Value range is from -5 to +5 dB, where 0 dB has no effect.
- **GAIN REDUCTION VU:** this analog VU shows the amount of Gain reduction in dB. When compression does not occur, the VU shows 0 dB. When compression is affecting the signal the VU indicates the level of compression in dB.
- **OUTPUT:** this knob controls the compressor output level.
- **BYPASS:** this switch allows you to bypass the Compressor module. When the LED is lit, the Compressor is ON. To turn OFF the Compressor module, click the switch so that the LED is OFF.
- **RESET:** brings the compressor back to its default state.

4.7 - Classic T-RackS Multi-band Limiter



This three-band peak limiter can make the mix even louder by reducing unwanted peaks. The peak limiting action is made separately on the three bands (low, mid, high). You will have a very loud master, without clipping, though as with any of these processors, use care not to over-do it. Leaving the output knob set to 0 dB means that the limiter output signal can't go beyond -0.05 dBfs.

Controls:

- **SINGLE BAND LEVEL:** these 3 knobs set the level (volume) of the three bands. This is very useful to get the correct level of each band after the limiting. Use this as a very broad tone control tool on the mix. This is a global, three band, equalizer too.
- **SINGLE BAND THRESHOLD:** these 3 knobs set the limiting amount over the individual band. Lowering the value the threshold will be lower and the limiting more frequent. Raising the threshold will make the limiter compress that band less. You can check this by looking at the analog VU on the right, this shows the average compression of the three bands.
- **CROSS OVER POINTS:** these two knobs set the two frequency split points, the low-mid split and the mid-high split. The default positions are fine in general but you can experiment with different settings. For example lower the low-mid point to, say, 70 Hz to only limit and tighten-up the low end.
- **RELEASE TIME:** adjust this control to change the speed of limiting in the release phase. This is the amount of time the limiter will take to return to normal gain after a peak limiting has occurred. While adjusting, check the current Limiter numeric display to see the release time you're setting. Value range is from 60 ms to 1.6 seconds.
- **INPUT DRIVE:** this knob controls the amount of signal that drives the multiband limiter input. Unlike the compressor module, this one has an exact threshold point where it starts to limit the signal. If you drive more signal through the limiter input, more peaks will be limited and the sound will be louder. If you want to preserve more peaks, set this control to a lower value. Values range is from -10 dB to + 15 dB. While adjusting Input Drive, check the values (in dB) by looking at the value above the Drive knob.
- **OVERLOAD:** this control will affect the way Limiter reduces peaks. Setting this knob to a lower value will

result in more frequent “pure” gain reduction by the limiter. Setting this knob to a higher value will result in less frequent gain reduction, but in more frequent clipping. Obviously setting this control at higher values will give you more loudness, but more clipping across the 0 dB level.

- **GAIN REDUCTION VU:** this analog VU shows the amount of Gain reduction in dB. When limiting does not occur the VU shows 0 dB. When limiting is affecting the signal, the VU indicates the level of limiting in dB. This VU shows the average value of limiting by all three spectral bands.
- **OUTPUT:** this knob controls the limiter output level.
- **BYPASS:** this switch allows you to bypass the Limiter module. When the LED is lit, the Limiter is ON. To turn OFF the Limiter module, click the switch so that the LED is OFF.
- **RESET:** brings the limiter back to its default state.

4.8 - Classic T-RackS Clipper



This clipping stage module is useful every time you need to cut out all the peaks that keep the average level of a signal too low. Peak clipping is very often used in mastering because it's in most cases more transparent than peak limiting.

Controls:

- **GAIN:** the Gain control adjusts the pre-level of this clipping stage. This is the level of the signal injected into the clipper and will determine the quantity of saturation applied to the audio. TIP: leave the OUTPUT now at 0 dB because doing this the clipper limiter will never go beyond -0.05 dBfs, regardless of the Gain level you dial in.
- **GAIN REDUCTION METER:** shows instant by instant the amount of signal that is attenuated by the clipping/saturation effect.
- **SLOPE:** the clipping shape of this stage is adjustable. It's continuously variable from a straight digital hard clip (fully clockwise) to ultra soft non-clipping mode (fully counterclockwise). Depending on music styles, the clipping can be less noticeable either with softer or harder shapes; that's why this control is continuously adjustable. Only your ears can detect the optimal clipping shape for the piece of music you're mastering. If you are unsure, start with a middle value, say 3 dB, as a starting point.
- **SAT:** when the signal is in the non-linear zone (saturation) the SAT LED lights up.
- **OUTPUT:** this is internally calibrated to ensure that the output of the clipper will never overload. This means that, regardless the input level of the clipper and the position of the GAIN knob, you'll never get samples over 0dB out from this module because the internal clipping has a -0.05 dBfs ceiling. This is only valid if the OUTPUT knob is set at 0 dB.
- **OUTPUT METER:** shows the peak output level on the same bar for L and R channels, L is on the top and R on the bottom.
- **OVER:** these two LEDs, separate for left and right channels, light up ONLY when a digital overload at the clipper outputs has occurred. In theory they should never turn on. Adjust the OUTPUT control to ensure they stay off, with as high a level as possible.

- **BYPASS:** this completely bypasses the Clipper module.
- **RESET:** brings the Clipper back to its default state.

4.9 - Classic T-RackS Equalizer



The T-RackS Classic Equalizer consists of a high quality six-band parametric EQ specially designed to achieve high-end performance in a mastering environment. It consists of:

- 1) 4th order High-pass filter, from 16 Hz to 5.3 kHz.
- 2) Low shelving type filter, from 30 Hz to 200 Hz.
- 3) Low-mid peaking type filter, from 33 Hz to 5.4 kHz, variable Q.
- 4) High-mid peaking type filter, from 200 Hz to 17.5 kHz, variable Q.
- 5) High shelving type filter, from 750 Hz to 8.5 kHz.
- 6) 4th order Low-pass filter, from 200 Hz to 18 kHz.

Controls:

- **LOW CUT band ON/OFF:** use this button to turn On and Off the band filter. When the filter is active the button is highlighted in yellow.
- **LOW CUT:** the knob adjusts this frequency of the high pass filter. Sweep range is from 16 Hz to 5.3 kHz.
- **LOW band ON/OFF:** use this button to turn the filter On and Off. When the filter is active the button is highlighted in yellow.
- **LOW FREQ:** the knob adjusts the frequency range of low-shelving filter. All frequencies up to this value will be boosted or reduced. You can set the cutoff frequency anywhere from 30 Hz to 200 Hz.
- **LOW GAIN:** use this control to boost or reduce the level of this band. The range is from -15 dB to +15 dB.
- **LOW MID band ON/OFF:** use this button to turn the filter On and Off. When the filter is active the button is highlighted in yellow.
- **LOW MID FREQ:** this knob adjusts the center frequency of the Low-Mid peaking filter. Range is 33 Hz to 5.4 kHz.

- **LOW MID GAIN:** use this control to boost or reduce the effect of this filter. Range is from -15 dB to 15 dB.
- **LOW MID Q:** sets the bandwidth of the peaking filter, from 0.2 (broad) to 20 (sharp).
- **HI MID band ON/OFF:** use this button to turn the filter On and Off. When the filter is active the button is highlighted in yellow.
- **HI MID FREQ:** this knob adjusts the center frequency of the Hi-Mid peaking filter. Range is from 200 Hz to 17.5 kHz.
- **HI MID GAIN:** use this control to boost or reduce the effect of this filter. Look at the value below the knob to check the value (in dB) you're setting. The range is from -15 dB to 15 dB.
- **HI MID Q:** sets the bandwidth of the peaking filter, from 0.2 (broad) to 20 (sharp).
- **HI band ON/OFF:** use this button to turn the filter On and Off. When the filter is active the button is highlighted in yellow.
- **HI FREQ:** this knob adjusts the frequency range of the high shelving filter. Range is from 750 to 8.3 kHz.
- **HI GAIN:** use this control to boost or reduce the effect of this filter. Its range is from -15 dB to +15 dB.
- **HI CUT band ON/OFF:** use this button to turn On and Off the band filter. When the filter is active the button is highlighted in yellow.
- **HI CUT:** this knob adjusts the cut-off frequency of the lo-pass filter. Range is from 200 to 18 kHz.
- **OUTPUT:** sets the equalizer output level from -15 to +15 dB.
- **CURVE DISPLAY:** this display shows the resulting equalization curve for both channels. Individual bands EQ points can be dragged to set Gain and Frequency points without using the equalizer knobs.
- **BYPASS:** completely bypasses the equalizer.
- **RESET:** brings the Equalizer back to its default, flat, state.
- **L=R:** linked channels mode. Set it this way when you want to control both left and right channels at the same time, by applying the exact same equalization to them.
- **L/R:** click on L or R icons when you want to tweak the equalization independently on the Left or Right channels. Left channel curve will be displayed in green, right channel in white.
- **M/S:** this option uses a Mid-Side matrix to insert the equalizers on your processing chain. This means that you will be able to equalize independently the Center or the Sides of your stereo program. The L and R icons will be replaced by M and S when in this mode. Click on M to equalize the Center (Mid) of your stereo image, or on S to equalize the Sides.

4.10 - Black 76 Limiting Amplifier



Black 76 is based on the most famous FET limiting amplifiers. The strong sonic character of Black 76 makes it one of the most versatile compressor - limiters. Based on all discrete transistor circuitry with true peak limiting and fast attack time, it is the go-to workhorse compressor for all instruments. Black 76 can be used on everything: drums, vocals, bass, guitars, piano, choirs, brass, strings... literally everything! Vocals, for example, will gain presence, consistency, clarity and punch when run through Black 76. Black 76 gives drums power, density, with total control on the level of “snap” you want, and it gives guitars a magic midrange zone that makes them really sing. This dynamics processing legend has been carefully modeled with IK’s maniacal attention to detail right down to the design of the VU meter for incredible accuracy.

Controls:

- **INPUT:** determines the amount of signal going into the Black 76.
- **OUTPUT:** determines the final level going out of the Black 76. Once the desired amount of limiting or compression is achieved with the use of the Input control, the Output control can be used to make up any gain loss due to the gain reduction process.
- **ATTACK:** sets the amount of time it takes the Black 76 to respond to an incoming signal and begin gain reduction. The Black 76 attack time is adjustable from 20 microseconds to 800 microseconds (both extremely fast). The attack time is fastest when the Attack knob is in its fully clockwise position, and it is slowest when in its fully counterclockwise position.
- **OFF:** when this switch is pressed the compression is disabled and the processor works as a signal amplifier.
- **RELEASE:** sets the amount of time it takes the Black 76 to return to its initial (pre-gain reduction) level. The release time is adjustable from 50 milliseconds to 1100 milliseconds. The release time is fastest when the Release knob is in its fully clockwise position, and it is slowest when in its fully counterclockwise position.
- **RATIO:** these controls set the behavior of the compressor related to the signal hitting the threshold. A setting of 4:1 means that signal exceeding the set threshold will be reduced by four times (4 dBs over the threshold will be reduced to 1dB). Likewise, 8:1. Higher ratios will make the Black 76 work as a limiter.

- **ALL:** this is the typical “ALL BUTTONS IN” setting that makes the Black 76 behave like no standard compressor due to the fact that some parameters like attack and release times are not fixed, but change constantly. This results in more distortion and very pronounced “pumping” effect that can be effectively used for some creative effects on room or ambient drum tracks, or to add distortion and sonic character in parallel compression to bass, guitars and vocal tracks.
- **METER:** these buttons will allow to monitor either the applied gain reduction or the plug-in’s output level. When the +4 button is selected a reading of 0dB on the metering will correspond to a -12dBFS level. When the +8 button is selected the 0dB on the meter will correspond to a -8dBFS level.
- **OFF:** this button power off the processor. Note that the signal flow will be interrupted.
- **RESET:** clicking once on this switch brings all controls back to their original default state where a moderate amount of compression is already applied (depending on the level of the input signal).
- **BYPASS:** completely bypasses the processing in the plug-in to compare with the unprocessed signal.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific compression settings for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo compressor on your processing chain. This means that you will be able to independently compress the Center or the Sides of your stereo program. The L and R icons will be replaced by M and S when in this mode. Click M to set the compression for the Center (Mid) of your stereo image or S to set compression for the Sides.

4.11 - White 2A Levelling Amplifier



Based on the most famous vacuum tube optical limiter Leveling Amplifier, the White 2A compressor is an impressive reproduction of one of the most popular and great sounding dynamics processing units in music recording history. Its tube driven electro-optical gain reduction circuit is well-known for a very smooth and natural compression, yet it is very vibey and incredibly musical when driven hard. This character has been faithfully captured in the White 2A plug-in — right down to the accurate movement of the meter — in order to provide you with one of the most useful tools for your compression needs. It is a must-have for tracking and mixing vocals and anything that needs gentle, warm, consistent compression. White 2A is also great for winds, bass, guitar, piano, organ, synths and more!

Controls:

- **GAIN:** the gain control should be set after the desired amount of compression is determined using the Peak Reduction control. Once the Peak Reduction control is set, adjust the Gain Control to set the desired output level.
- **METER SELECTOR:** sets the Metering display to show the actual output level with two different scales: +10dB and +4dB. When set to Gain Reduction, you can monitor the amount of compression applied.
- **PEAK REDUCTION:** sets the amount of compression applied to the signal processed by the plug-in. There is no compression when the knob is turned fully counterclockwise, and there is maximum compression when it is turned fully clockwise.
- **RESET:** clicking once on this switch put all the controls back to their original default state where a moderate amount of compression is applied (depending on the level of the input signal).
- **LIMIT/COMPRESS:** changes the compression ratio between two fixed settings: COMPRESS has a more gentle curve, while LIMIT has a more pronounced effect.
- **BYPASS:** completely bypasses the processing in the plug-in to compare with the unprocessed signal.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.

- **L/R:** click on the L or R icons if you want to dial in specific compression setting for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo compressor on your processing chain. This means that you will be able to independently compress the Center or the Sides of your stereo program. The L and R icons will be replaced by M and S when in this mode. Click M to set the compression for the Center (Mid) of your stereo image or S to set compression for the Sides.

4.12 - British Channel



The British Channel module is based on one of the most popular and universally acclaimed British mixing consoles which hit the market with great impact in the 80's and forever changed the way records were made. The British Channel carefully reproduces all of its op-amp driven EQ and VCA Compressor/Expander-Gate behavior and, as for the real thing, it can be used on all kinds of sources across a DAW to shape and finely craft the mix's sound. Its wealth of controls will make getting into each mix's detail easy, and all instruments will gain their own sound and dimension thanks to a super-precise EQ section (with selectable circuit mode) and a dynamics section capable of going from subtle level control to extreme squashing and effective gating.

Controls:

EQ Section:

LOW Band:

- **Gain - dB:** determines the amount of boost or cut in Decibels for this frequency band.
- **Freq - Hz:** sets the frequency in Hertz at which the boost or cut happens.
- **PEAK:** when this switch is pressed, the LOW band changes from low shelving type to peaking type with fixed Q.

LOW-MID Band:

- **Gain - dB:** determines the amount of boost or cut in Decibels for this frequency band.
- **Freq - KHz:** sets the frequency in Kilohertz at which the boost or cut happens.
- **Q:** sets the width of the frequency band where extreme left results in a narrow and sharper band, while extreme right results in a wide and broader band.

HI-MID Band:

- **Gain - dB:** determines the amount of boost or cut in Decibels for this frequency band.
- **Freq - KHz:** sets the frequency in Kilohertz at which the boost or cut happens.

- **Q:** sets the width of the frequency band where extreme left results in a narrow and sharper band, while extreme right results in a wide and broader band.

HI Band:

- **Gain - dB:** determines the amount of boost or cut in Decibels for this frequency band.
- **Freq - KHz:** sets the frequency in Kilohertz at which the boost or cut happens.
- **PEAK:** when this switch is pressed, the HI band changes from high shelving type to peaking type with fixed Q..

FILTERS Section (LOW-PASS/HI-PASS):

- **FLT IN:** this switch enables the filter section which can be used independently from the rest of the EQ section.
- **Freq - Hz:** sets the frequency of the High Pass filter rolloff point. The filter has a 12 dB/Octave cut (18 when in Black mode). In the OUT position the filter is inactive: this effectively allows to use the two filters independently.
- **Freq - KHz:** sets the frequency of the Low Pass filter rolloff point. The filter has a 12 dB/Octave cut. Just like the other, in the OUT position the filter is inactive.
- **FLT S/C:** this switch makes the filter section act as the detector for the sidechain circuit of the dynamics section. When engaged the HP and LP filters are not usable for their primary purpose.
- **IN:** this switch turns the whole EQ section (minus filters) on or off.
- **B:** this switch allows to choose which EQ mode the British Channel will turn in: when disengaged the mode is Brown, meaning that the EQ behavior will be the one of the Brown EQ circuit, while when engaged the EQ mode will be the one of the Black EQ circuit with its distinctive frequency shifts and boost/cut/Q differences. This two different circuit designs offer sonically distinct results, and having them both available at the press of a switch is a very handy resource for the ultimate tone tweaking.

DYNAMICS Section:

- **IN:** this switch enables the dynamics section.

GATE/EXP:

- **EQ PRE:** this switch modifies the sequence of EQ and dynamics sections. When engaged the compressor section is shifted before the EQ section, effectively changing how the whole module reacts to the incoming signal in relation to the parameters settings.
- **EQ S/C:** this switch enables the sidechain circuit, meaning that the EQ section is used as a detector for the dynamics section. This is a very effective way of compressing a particular frequency which is selected with the EQ section. When this switch is enabled the EQ section will not work the standard way and could not be used to EQ the incoming signal (you will have to instantiate another British Channel EQ/Dyn plug-in or use another EQ).
- **THRESHOLD:** this control sets the level (in dB) below which the incoming signal is attenuated by the gate/expander circuit.
- **RANGE:** this parameter controls the amount of gain reduction (in dB) applied to the incoming signal when this falls below the threshold level. The extreme counterclockwise position will have none to minimal gain reduction, while more effective gating (for more noisy or problematic tracks) is obtained turning progressively this control towards the 40 dB value (fully clockwise).
- **RELEASE - S:** this control determines the time (in seconds) it takes the expander/gate to release the

signal to its normal level once it has gone above the threshold.

- **FAST ATT:** this control switches the attack time it takes to the gate/expander circuit to open, to 1ms; this can be useful to produce some very particular effects like snappy and percussive sounds with very marked attack transients.
- **EXP:** this control activates the expansion mode for the gate/expander section.

COMP:

- **THRESHOLD:** this control sets the level in dB which has to be passed by the input signal coming into the British Channel module for the compression to start working. At the 0 position the module is not compressing. Turning the knob clockwise will lower the threshold and, depending on the input signal, start compressing. Turning the knob counterclockwise towards the +10 value will lift the threshold so that hotter signals can be easily managed. This control and its behavior can be affected by the Master section's I/O level controls. As an example, try driving the compressor's input level harder with the Master's section IN knob and set the threshold low for some serious squashing and pumping compression effects. The OUT knob will make calibrating the final output of the module easy in order to avoid clipping. A note on Auto Makeup Gain: this compressor circuit design is provided of an auto makeup gain control; this will progressively compensate the output level in relation to the threshold control avoiding a drop in overall output level as a result of gain reduction. The lower the threshold, the more you will hear it raise the output level, so pay attention about setting a correct gain staging among EQ, Compressor and Master I/O levels.
- **RATIO:** this control determines the gain reduction amount applied to the signal passing above the level set by the threshold control. Lower values of this control (1:1, 1.5:1) will result in none to very light compression, while turning the knob clockwise will progressively make for a more pronounced compression effect as you approach higher values; the extreme clockwise position will result in a limiting effect.
- **FAST ATT:** this switch shifts the attack time it takes the circuit to start compressing once the incoming signal has passed the set threshold to a faster 1 ms. It can be quite handy to emphasize attack transients on percussive instruments like snare or kick drums and the likes.
- **RELEASE - S:** this control sets the time(in seconds) it takes the compressor to release the signal back to uncompressed state after it has gone below the threshold level. Shorter release times will make the compressor snappy to downright "pumping" when lots of compression is applied. Longer release times will result in a more natural sounding effect and increased sustain on particular instruments.
- **HARD:** this switch changes the detecting mode of the compressor from RMS to peak.

MASTER Section:

- **Ø (Phase):** this control shifts the phase of the incoming signal by 180°, useful to check phase compatibility of sources or to reverse phase in one of two coupled tracks (like snare drum top and bottom).
- **IN:** this control lets you trim the input level of the signal going into the British Channel module. It can be used to correct the level of the incoming signal when needed (in case of a particularly weak or hot track) since it has a range of +/- 15 dB. As described above (in the compressor's threshold parameter section) it can be used to drive the compressor's section harder by boosting the input signal, thus producing some very particular compression effects which can be further emphasized by ratio, attack and release time settings.
- **OUT:** this control lets you trim the master output level of the plug in. It can be used to avoid clipping and overloading of the output stage in case some very radical EQ boosting is applied, or to simply control the level of a track that needs to be adjusted.

- **RESET:** this control brings the setting of all of the parameters back to their default status once clicked.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific compression settings for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo compressor on your processing chain. This means that you will be able to independently compress the Center or the Sides of your stereo program. The L and R icons will be replaced by M and S when in this mode. Click M to set the compression for the Center (Mid) of your stereo image or S to set compression for the Sides.

4.13 - Quad Comp



The Quad Comp is one of the four modules in the Multiband series of IK Multimedia plug ins for T-RackS. Multiband processing at the core of this plug-in splits the incoming signal into up to four frequency bands and treats each one separately with a dedicated set of controls. The width of each band can be finely adjusted so that the processing can be very selective (in case of a narrow band) or spread across a wider range. Each band is loaded with an optical compressor of particular effectiveness but with a very smooth sound and a full set of adjustable parameters.

Controls:

- **FREQUENCY Display:** this big display shows the whole frequency spectrum and how it's divided in the four bands by the vertical frequency bars. To modify the width of a band click and drag on the top square marker and adjust to the desired frequency (which will appear during movement). Clicking on a band's area will automatically select it and this will be reflected by the set of underlying controls which will change values accordingly. Click and drag on the central dot to adjust the band's gain (it will be reflected by the band's underlying gain control).
- **METERING Display:** this display shows the gain reduction (in dBs) for each band and the master output level. When the Quad-Comp is used in STEREO mode the gain reduction will be shown for each single band independently. When the plug in is used in M/S (Mid Side) mode the gain reduction meter of each band will switch to a different visualization where two separate bar of LEDs will show Mid gain reduction (red LEDs) and Side gain reduction (grey LEDs).
- **MASTER OUTPUT Fader:** use this master fader to adjust the final output of the plug in in order to optimize level and avoid clipping (in case of too hot signals) or to bring back the signal to a more consistent level after compression has been applied.
- **ON/BYPASS Switch:** this control enables the plug in processing when in the "ON" status.
- **THRESHOLD:** this control sets the threshold level for the selected band at which the plug in will start compressing the incoming signal. Turning the knob counterclockwise will progressively lower the threshold resulting in more compression.

- **RATIO:** this control sets the amount of gain reduction applied to the incoming signal once it has passed above the threshold level. Lower values of this control (when turned fully counterclockwise) will result in none to light compression of the selected frequency band, while progressively turning the knob clockwise will increase gain reduction. The extreme clockwise position with a ratio of 30:1 will result in very high compression.
- **ATTACK:** this control sets the time it takes the compressor to react and apply gain reduction once the incoming signal has passed above the set threshold. The extreme counterclockwise values will result in a very fast attack (0.1 ms) which will emphasize transients on percussive sources.
- **RELEASE:** this control determines the amount of time it takes the compressor to bring the signal back to its uncompressed state after it has fallen below the threshold. Short release times will result in a more audible compression effect while longer ones (towards the extreme clockwise position) will result in a more natural compression and in increased sustain of several sources.
- **GAIN:** this control sets the output level of the selected band. It can be used as a makeup gain to compensate for level loss after compression or to adjust the balance among the four bands. Adjustments of this control will be reflected by the “gain dot” on selected band and the shape of the horizontal white gain line.
- **SOLO:** this button puts the selected band in solo mode to listen to its frequency range and gain reduction applied.
- **BYPASS:** this control bypasses the compression for the selected band.
- **BANDS:** this control lets you select the number of active bands the frequency spectrum will be divided into.
- **SLOPE:** This control determines the steepness of the filters which will be used to split the frequency spectrum in bands. At its minimum (6dB/Oct) the slope of the filter is quite gentle and allows for more overlapping among the bands. At its maximum value (48 dB/Oct) the filter is more selective and separation among bands becomes sharper.
- **LINK BANDS:** this button links the controls for all bands so that the values set for a selected band will be mirrored on all of the other bands. This can be handy when all control must be set identical for all bands. The only exception is the gain control which can be still independently modified for each band by using the Gain Dot on the frequency display.
- **STEREO:** this control links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **MID/SIDE:** this option uses a Mid-Side matrix to independently compress the Center or the Sides of your stereo program. Click MID to set the compression for the Center (Mid) of your stereo image or SIDE to set compression for the Sides.
- **RESET Switch:** clicking on this control will reset all of the control to their initial default state. Clicking again after the reset will bring them back to the last setting they were left, in case you accidentally hit the reset button, or changed your mind.

4.14 - Quad Lim



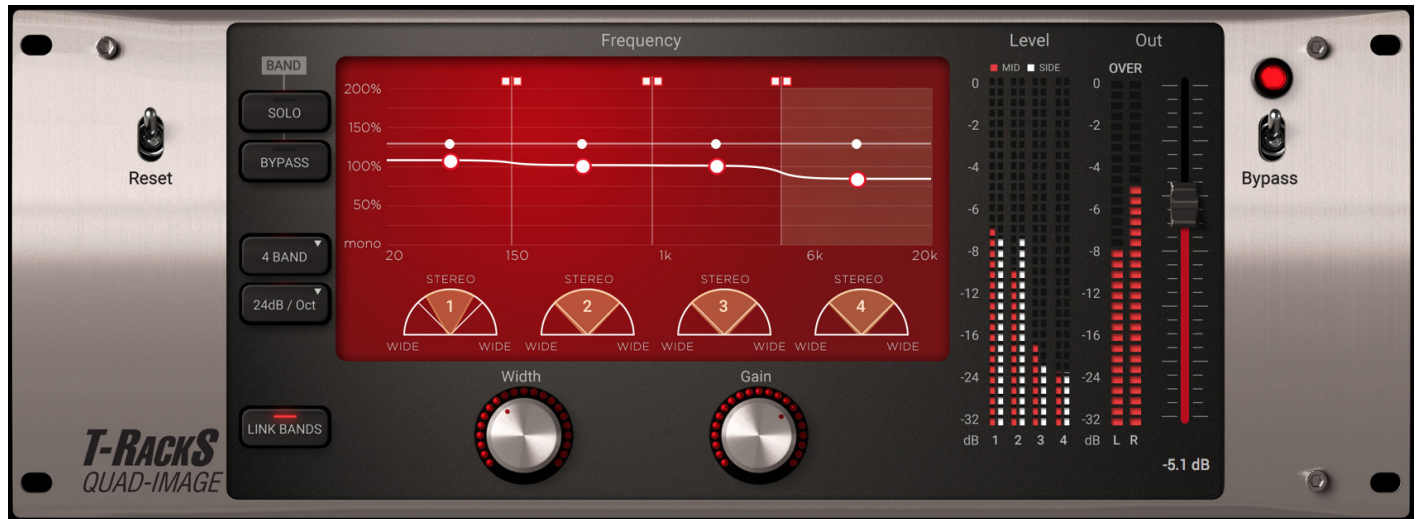
The Quad Limiter is one of the four plug-ins in the Multiband series for T-RackS. Multiband processing at the core of this plug-in splits the incoming signal into up to four frequency bands and treats each one separately with a dedicated set of controls. The width of each band can be finely adjusted so that the processing can be very precise (in case of a narrow band) or spread across a wider range. Each band is loaded with a peak limiter which controls dynamics with extreme precision and accuracy across the whole spectrum, with the added flexibility of selectable frequency zones. In addition to the four limiters the Quad Lim is loaded with a master brickwall limiter located post processing so that no overs can escape and clipping is avoided. This will let you produce very loud masters with a controlled frequency balance. Of course, be careful with limiting amount as too much will produce flat tracks without dynamics ... unless you want that!

Controls:

- **FREQUENCY Display:** this big display shows the whole frequency spectrum and how it's divided in the four bands by the vertical frequency bars. To modify the width of a band click and drag on the top square marker and adjust to the desired frequency (which will appear during movement). Clicking on a band's area will automatically select it and this will be reflected by the set of underlying controls which will change values accordingly. Click and drag on the central gain dot to adjust the band's gain (it will be reflected by the band's underlying gain control).
- **METERING Display:** this display shows the gain reduction (in dBs) for each band and the master output level. When the Quad-Lim is used in STEREO mode the gain reduction will be shown for each single band independently. When the plug in is used in M/S (Mid Side) mode the gain reduction meter of each band will switch to a different visualization where two separate bar of LEDs will show Mid gain reduction (red LEDs) and Side gain reduction (grey LEDs).
- **MASTER OUTPUT Fader:** use this master fader to adjust the final output of the plug in. This fader works pre Brickwall Limiter so it can be used to boost the overall level and amount of limiting after the settings on the four bands have been adjusted without the risk of clipping or overload.
- **ON/BYPASS Switch:** this control enables the plug in processing when in the "ON" status.

- **THRESHOLD:** this control sets the threshold level for the selected band at which the plug in will start limiting the incoming signal. Turning the knob counterclockwise will progressively lower the threshold resulting in more limiting.
- **ATTACK:** this control sets the time it takes the limiter to react and apply gain reduction once the incoming signal has passed above the set threshold. The extreme counterclockwise values will set a very fast attack (0.05 ms) which will result in a very fast response from the plug in catching the most instantaneous transient peaks.
- **RELEASE:** this control determines the amount of time (in ms) it takes the limiter to bring the signal back to its normal state after it has fallen below the threshold. As a general guideline for a correct peak limiting action, release time should be set as short as possible, in order to “clamp” transient peaks without touching the overall RMS level. Anyway, too short release times will cause audible distortion and artifacts (especially at low frequencies), so take the time to adjust this parameter carefully.
- **GAIN:** this control sets the output level of the selected band. Working in conjunction with the threshold parameter it can be used to raise the gain of the selected band effectively making it louder as transient peaks are limited and RMS value of the track increased. Adjustments of this control will be reflected by the “gain dot” on the selected band and the shape of the horizontal white gain line.
- **SOLO:** this button puts the selected band in solo mode to listen to its frequency range and limiting applied.
- **BYPASS:** this control bypasses the limiting action for the selected band.
- **BANDS:** this control selects the number of active bands the frequency spectrum will be divided into.
- **SLOPE:** this control determines the steepness of the filters which will be used to split the frequency spectrum in bands. At its minimum (6dB/Oct) the slope of the filter is quite gentle and allows for more overlapping among the bands. At its maximum value (48 dB/Oct) the filter is more selective and separation among bands becomes sharper.
- **LINK BANDS:** this button links the controls for all bands so that the values set for a selected band will be mirrored on all of the other bands. This can be handy when all controls must be set identical for all bands. The only exception is the gain control which can be still independently modified for each band by using the “gain dot “ on the frequency display.
- **BRICKWALL:** this control activates the brickwall limiter placed at the back end of the four bands. Consider this additional limiter as an extreme protection to avoid any possible clipping or overloading of the program; its fixed parameters are carefully adjusted to result in maximum transparency and effective overload protection. Working in conjunction with the master output fader it can also be used to globally increase the level of the track after the limiters on all four bands have been set.
- **STEREO:** this control links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **MID/SIDE:** this option uses a Mid-Side matrix to independently limit the Center or the Sides of your stereo program. Click MID to set the limiting for the Center (Mid) of your stereo image or SIDE to set limiting for the Sides.
- **RESET Switch:** clicking on this control will reset all of the controls to their initial default state. Clicking again after the reset will bring them back to the last setting they were left in, in case you accidentally hit the reset button or changed your mind.

4.15 - Quad Image



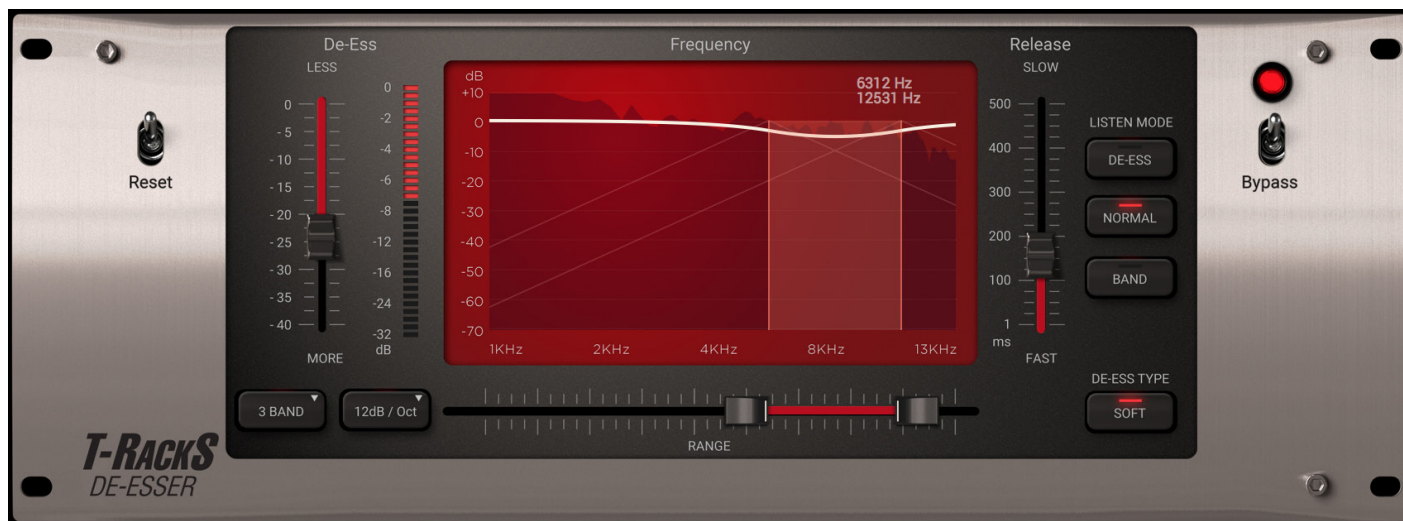
The Quad Image is one of the four modules in the Multiband series of IK Multimedia plug ins for T-RackS. Multiband processing at the core of this plug-in splits the incoming signal into up to four frequency bands and treat each one separately with a dedicated set of controls. The width of each band can be finely adjusted so that the processing can be very selective (in case of a narrow band) or spread across a wider range. Each band is loaded with an M/S matrix processor which modifies its stereo width.

Controls:

- **FREQUENCY Display:** this big display shows the whole frequency spectrum and how it's divided in the four bands by the vertical frequency bars. To modify the width of a band click and drag on the top square marker and adjust to the desired frequency (which will appear during movement). Clicking on a band's area will automatically select it, and this will be reflected by the set of underlying controls which will change values accordingly. Click and drag on the central gain dot to adjust the band's gain (it will be reflected by the band's underlying gain control). The horizontal white line will display the percentage of stereo widening applied to each band with 0% being mono (or all M signal) and 200% being wide open stereo (or all S signal). The four underlying "radar" displays will monitor the stereo width of each band.
- **METERING Display:** this set of led meters will display the M and S level (with different colors) for each band.
- **MASTER OUTPUT Fader:** this control will regulate the master output of the plug in which is shown on the adjacent led meter.
- **WIDTH:** this control regulates the amount of stereo spread for the selected band.
- **GAIN:** this control sets the amount of gain of the selected band.
- **SOLO:** this button puts the selected band in solo mode allowing to listen to its frequency range and stereo width applied.
- **BYPASS:** this control bypasses the stereo widening action for the selected band.
- **BANDS:** this control selects the number of active bands the frequency spectrum will be divided into.

- **SLOPE:** this control determines the steepness of the filters which split the frequency spectrum in bands. At its minimum (6dB/Oct) the slope of the filter is quite gentle and allows for more overlapping among the bands. At its maximum value (48 dB/Oct) the filter is more precise and separation among bands becomes sharper.
- **LINK BANDS:** this button links the controls for all bands so that the values set for a selected band will be mirrored on all of the other bands. This can be handy when all controls must be set identical for all bands. The only exception is the gain control which can be still independently modified for each band by using the “gain dot “ on the frequency display.
- **RESET Switch:** clicking on this control will reset all of the controls to their initial default state. Clicking again after the reset will bring them back to the last setting they were left in, in case you accidentally hit the reset button or changed your mind.
- **ON/BYPASS Switch:** this control enables the plug in processing when in the “ON” status.

4.16 - De-Esser



The De-Esser is one of the four processors in the Multiband series for T-RackS. With this processor, the de-essing action and excessive sibilance or high frequency control can be adjusted in a dual operating mode depending on the number of bands selected. A wealth of controls adjusts the de-essing action to suit any particular situation.

Controls:

- **FREQUENCY Display:** this big display shows the whole frequency spectrum and how it's divided with the vertical bars. The frequency range of this module is centered in a different area than the other processors in the Multiband Series (its range goes from 1 to 13KHz). This is because de-essing process is only used in this range of the spectrum. To modify the width of the bands, use the underlying slider control.
- **DE-ESS:** this control determines the amount of de-essing applied. The resulting reduction in HF content is displayed on the adjacent led meter. Be careful to adjust this control as too much de-essing action could result in unnatural sound, especially when used on vocal tracks.
- **RELEASE:** this control sets the release time it takes the De-Esser to bring the signal back to its normal state once the signal has fallen below the set de-essing action.
- **DE-ESS ACTION:** use this control to listen to the effective amount of de-essing applied to the track.
- **NORMAL:** this control sets the listening mode so that you'll hear the track while the de-essing action is applied.
- **DE-ESS BAND:** this control allows you to put the De-Ess band in solo mode.
- **SOFT:** this control sets the intensity of the de-essing action, when selected the operating mode is more gentle with a less pronounced and slower HF reduction, while when de-selected it's faster and more marked action.
- **BANDS:** this drop down menu allows to select the number of bands the frequency spectrum will be divided into: when set to 2 bands the de-essing action will be performed on the higher band, while when 3 bands is selected it will be possible to set the higher and lower margin so that the de-essing action will be performed on a selected area.

- **SLOPE:** this control sets the steepness of the filters used to split the frequency spectrum into bands. Higher values of this control will result in harder separation with less interaction of the de-essing action on adjacent bands.
- **RESET Switch:** clicking on this control will reset all of the controls to their initial default state. Clicking again after the reset will bring them back to the last setting they were left in, in case you accidentally hit the reset button or changed your mind.
- **ON/BYPASS Switch:** this control enables the plug in processing when in the “ON” status.

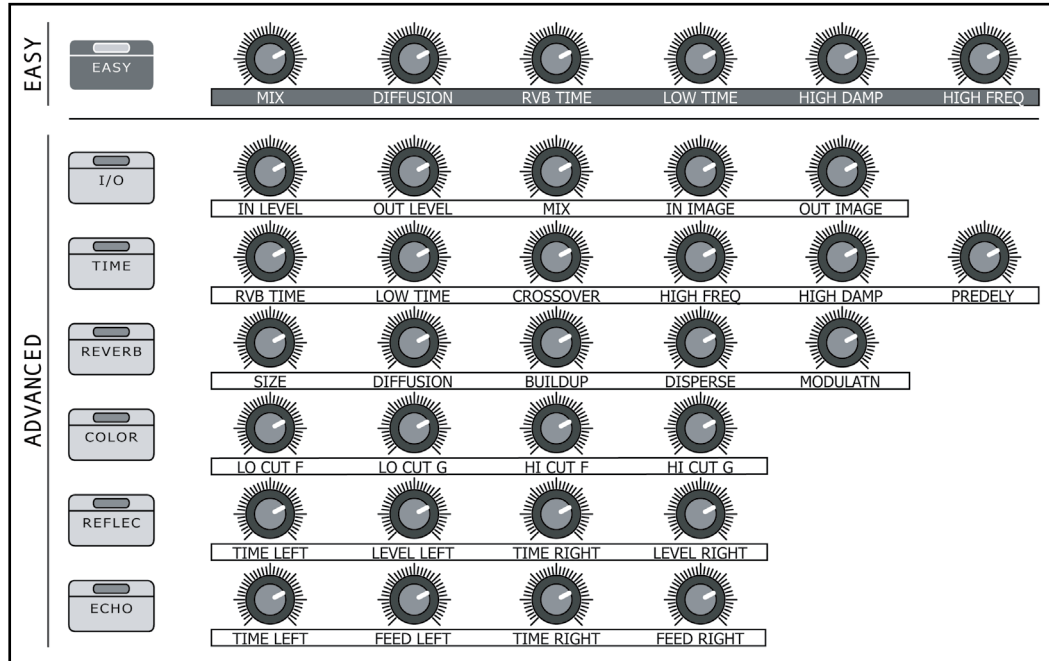
4.17 - CSR Plate Reverb



Plate reverbs have been a staple in pop music recording since the late fifties. They consisted of a large metal sheet, suspended from a heavy steel frame, with a small transducer (like a small speaker) transmitting the sound energy onto the plate. A pickup (or 2 if stereo) was used to capture the sound. This was all housed in a 4 x 8 foot box. When originally designed, this was considerably smaller than the forerunning “echo chamber”.

Since the creation of digital reverbs, to further exemplify its popularity, the plate reverb has been found on just about every rack-mounted reverb box. It is exceptional on vocals, having a bright, slightly metallic character, but warm enough to sit well in most mixes without muddying up the singer. It is also often used on drums and percussion. Its smooth attack and bright characteristic make it a great choice for adding depth to a snare drum without burying it in the reverberation.

There are two operating modes for this module: “Easy” for a basic control of the most commonly used parameters, and “Advanced” to tweak every detail and have deep control of the reverb shape and tone. The diagram below shows the controls breakdown of Classik Studio Reverb’s “Classik Plate”.



4.17.1 - Easy

- **MIX:** controls the ratio of reverb and original sound. 0% is all unprocessed sound. 100% is all reverb. Use this control to increase or decrease the amount of reverb in the sound. If you are using your reverb as a send effect, this control will probably be set to 100%.
- **DIFFUSION:** determines the rate that the echo density increases over time. A small diffusion setting simulates an acoustic space with smooth walls, like a bathroom. A larger diffusion setting creates a more open and dense sounding reverb.
- **RVB TIME (Reverb Time):** adjusts the length of the reverb. This is a commonly used control that shortens and lengthens the time that the reverb tail lasts.
- **LOW TIME:** controls the length of the reverb for low frequencies (determined by the Crossover control) of the RVB Time.
- **HIGH DAMP (High Dampening):** shortens the high frequency decay time of the reverb. This allows you to emulate the way different materials in a real room, such as carpet and furnishings, absorb sound.
- **HIGH FREQ (High Frequency):** changes the frequency of high frequency dampening.

4.17.2 - I/O Levels

- **IN LEVEL:** adjusts the input level of the reverb.
- **OUT LEVEL:** adjusts the output level from the reverb.

- **MIX:** controls the ratio of reverb and original sound. 0% is all unprocessed sound. 100% is all reverb. Use this control to increase or decrease the amount of reverb in the sound. If you are using your reverb as a send effect, this control will probably be set to 100%.
- **IN IMAGE:** controls the stereo image of the input signal before it is processed through the effect. This control allows you to change the signal from normal stereo to mono and manipulate the phase of the signal. For example a value of -45 swaps the left and right channels, 0 is mono, and 45 is normal stereo. With a setting of 90, all mono information will be excluded from the reverb sound.
- **OUT IMAGE:** controls the stereo image of the affected signal. With a value of -360 to +360, this control extends the stereo field of the processed sound. This control can be used to create mono, stereo, or surround type effects. It can also be changed in real-time or automated using the modulation or macro matrix.

4.17.3 - Time

- **RVB TIME (Reverb Time):** adjusts the length of the reverb. In other words, this shortens and lengthens the time that the reverb tail lasts.
- **LOW TIME:** controls the length of the reverb for low frequencies (determined by the Crossover parameter) of the RVB Time.
- **CROSSOVER:** changes the crossover frequency for the Low time control.
- **HIGH FREQ (High Frequency):** changes the frequency for the low-pass filter used for High Frequency damping.
- **HIGH DAMP (High Dampening):** dampens (attenuates) the high frequency being re-circulated into the reverberation. Using High frequency dampening often gives a more natural sounding reverb.
- **PREDELAY (Predelay):** determines the amount of time between the initial input signal and the reverberation. Increasing the predelay can create a sense of distance and, to some degree, change the size of the acoustic space.

4.17.4 - Reverb

- **SIZE:** controls the approximate size of the acoustic space from small to large. This control is measured in meters and is approximately equal to the longest dimension of the space.
- **DIFFUSION:** determines the rate that the echo density increases over time. A small diffusion setting simulates an acoustic space with smooth walls, like a bathroom. A larger diffusion setting creates a more open and dense sounding reverb.
- **BUILDUP:** changes the speed of the reverb's buildup. When this control is turned down, it causes a fast attack in the reverb. To create a slower buildup in the reverberation, turn up the buildup control. Increasing the Buildup will create the sense of a large room/hall, but the spatial information can be lost - this can be restored with the reflection parameters (delay and level).
- **DISPERSE:** disperse works in conjunction with the Buildup control to determine the initial envelope of the reverberation. You can use Disperse to vary the apparent size and character of the early reflection.
- **MODULATN (Modulation):** creates movement in the reverberation tail. Modulation continually changes the timbre of the reverberation. If used in smaller amounts, 10%-50%, it can result in a more natural sounding reverb. If used higher than that, it can cause acoustic instruments timbre to become unstable.

4.17.5 - Color

- **LO CUT F (Low Cut Frequency):** changes the frequency of the low cut filter in the reverberation path. This control ranges from 10Hz to 20 kHz.
- **LO CUT G (Low Cut Gain):** changes the gain of the low cut filter. This is adjustable from -24dB to 0dB.
- **HI CUT F (High Cut Frequency):** changes the frequency of the high cut filter in the reverberation path. This control ranges from 10Hz to 20 kHz.
- **HI CUT G (High Cut Gain):** changes the gain of the high cut filter. This is adjustable from -24dB to 0dB.

4.17.6 - Reflections

- **TIME LEFT:** changes the delay between the input signal and the early reflections in the left channel.
- **LEVEL LEFT:** changes the level of the reflections in the left channel.
- **TIME RIGHT:** changes the delay between the input signal and the early reflections in the right channel.
- **LEVEL RIGHT:** changes the level of the reflections in the right channel.

4.17.7 - Echo

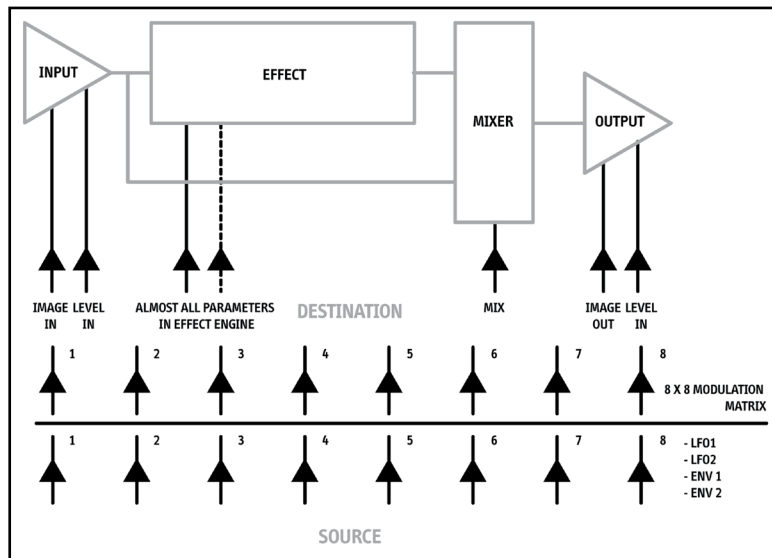
- **TIME LEFT:** increases and decreases the length of the echo in the left channel. This echo is routed to the reverb engine and the feedback path as well.
- **FEED LEFT:** controls the amount of delay that is being sent to the feedback paths. To hear significant changes in this control, you may have to change this level dramatically.
- **TIME RIGHT:** increases and decreases the length of the echo in the right channel. This echo is routed to the reverb engine and the feedback path as well.
- **FEED RIGHT:** controls the amount of delay that is being sent to the feedback paths. To hear significant changes in this control, you may have to change this level dramatically.

4.17.8 - Modulation Matrix

This section of controls is common to all the CSR modules.

The modulation matrix is a flexible system for adding continuous movement to the parameters and sound of Classik Studio Reverb modules. It allows you to route 4 modulation sources to a selection of parameters in the reverb engine. You can achieve subtle or extreme variations in the reverb by applying the modulation sources to the different parameters (destinations) in Classik Studio Reverb.

Here is a diagram of the modulation system and its sources and destinations.



4.17.8.1 - Modulation

The matrix window displays your modulations sources, destinations, minimum and maximum values, and curve controls.

SOURCE	DESTINATION	MIN	MAX	CURVE
LF01	Rvb - Diffusion	11%	100%	Lin
LF01	I/O - Image In	-247	360	Lin
LF01	Rvb - Buildup	6%	99%	Log
LF01	Rvb - Modulation	6%	90%	Exp
Env1	I/O - Mix	14%	100%	Lin
Env1	Rvb - Buildup	7%	86%	Log
LF02	I/O - Level In	-94.0dB	0.0dB	Exp
LF02	I/O - Level Out	-87.9dB	-4.1dB	Lin

4.17.8.2 - Sources (modulation sources)

LFO1 and LFO2

Low frequency oscillators generate Sine, Triangle, Sawtooth, Square Wave, and Noise all with adjustable rates. These signals can be used to modulate any of the available destinations in the matrix.



- **TYPE:** selects the waveform of the oscillator.
- **RATE:** selects the speed of the waveform.

ENV1 and ENV2

Envelope generators featuring attack and release. They can be triggered from Input L, R, or summed and Reverb L, R, or summed. These envelopes can be used to modulate any of the available destinations in the matrix.



- **INPUT:** selects the input trigger of the envelope generator.
- **GAIN:** adjusts the amount of gain required to trigger the envelope.
- **ATTACK:** changes the length of time that the envelope ramps up to full open.
- **RELEASE:** changes the amount of time that the envelope returns to off after it has been opened.

Any of the modulation sources can be routed multiple times up until all 8 inputs of the modulation matrix are full.

4.17.8.3 - Destinations

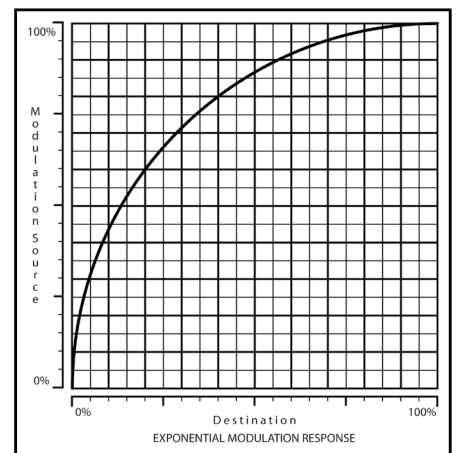
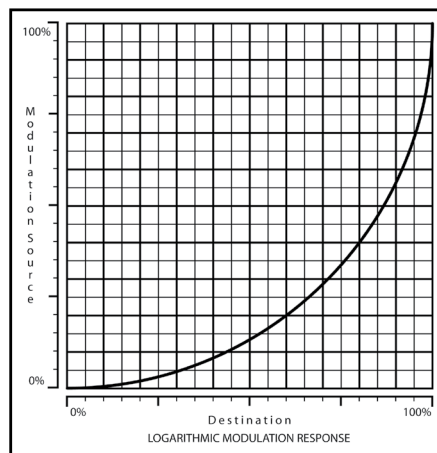
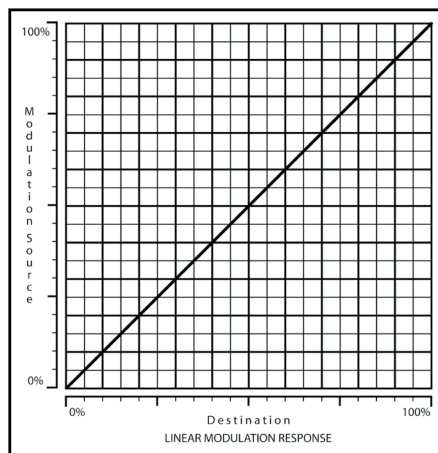
Destinations are the parameters in the reverb engine that can be modulated. Almost every parameter in the CSR reverb engine can be chosen.

SOURCE	DESTINATION	MIN	MAX	CURVE
LF01	Rvb - Diffusion	11%	100%	Lin
LF01	I/O - Image In	-247	360	Lin
LF01	Rvb - Buildup	6%	99%	Log
LF01	Rvb - Modulation	6%	90%	Exp
Env1	I/O - Mix	14%	100%	Lin
Env1	Rvb - Buildup	7%	86%	Log
LF02	I/O - Level In	-94.0dB	0.0dB	Exp
LF02	I/O - Level Out	-87.9dB	-4.1dB	Lin

- **MIN:** assigns the minimum value of modulation that can be applied to the destination. This value will change depending on the selected parameter.
- **MAX:** assigns the maximum value of modulation that can be applied to the destination. This value will change depending on the selected parameter.
- **CURVE:** changes the response curve between the source and the destination.

There are three available curves:

- **LIN (Linear):** provides a linear response between the source and the destination.
- **LOG (Logarithmic):** applies a logarithmic scale to the response between the source and destination.
- **EXP (Exponential):** provides an exponential scale of response between the source and the destination.



4.17.9 - MACROS

This section of controls is common to all the CSR modules.

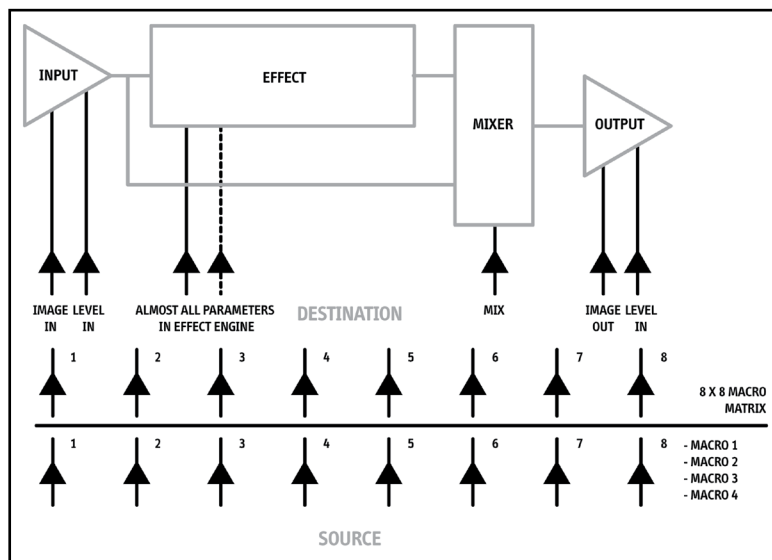
The MACRO controls are the 4 sliders on the far right hand side of the interface. These sliders can be assigned to multiple parameters. So you can add additional single controls to the front panel, giving you more easy control to a preset. You can also create complex morphing controls, making it possible to move from one type of reverb sound to a totally different reverb sound in a single slider movement.

4.17.9.1 - MATRIX

The MACROS are configured in much the same way as the modulation matrix.

SOURCE	DESTINATION	MIN	MAX	CURVE
Macro1	I/O - Level In	-83.3dB	-3.2dB	Lin
Macro2	I/O - Mix	11%	100%	Log
Macro3	I/O - Level Out	-96.0dB	0.0dB	Lin
Macro4	Rev - Buildup	22%	100%	Exp
Macro1	Color - Low Cut G	20Hz	20000Hz	Log
Macro2	Rev - Diffusion	10%	100%	Lin
Macro3	Time - PreDelay	8ms	250ms	Lin
Macro4	I/O - Image In	-360	360	Lin

There is an 8x8 matrix that allows you to route the 4 MACRO controls to multiple destinations. Here is a diagram of the MACRO controls matrix.



The MACRO matrix functions like the modulation matrix in that it can route any of the modulation sources multiple times up until all 8 inputs of the matrix are full. The controls of the MACRO matrix are the same as well.

4.17.9.2 - Sources (modulation sources)

MACRO 1-4: slider controls that can control multiple functions from the front panel, making it possible to create morphing effects using a single control.

4.17.9.3 - Destinations

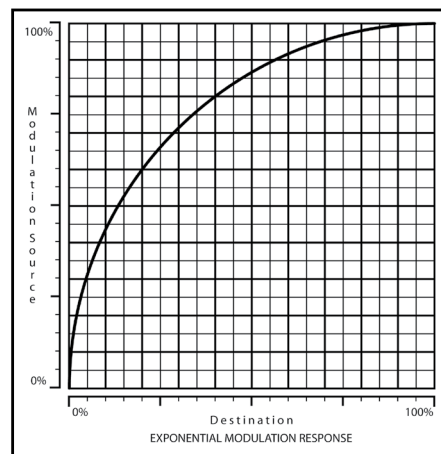
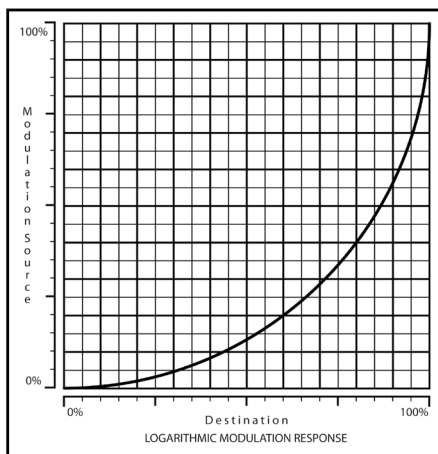
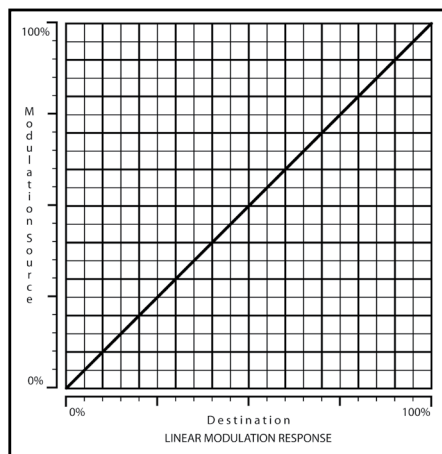
Destinations are the parameters in the reverb engine that can be controlled. You can choose almost every parameter as a destination source.

SOURCE	DESTINATION	MIN	MAX	CURVE
Macro1	I/O - Level In	-83.3dB	-3.2dB	Lin
Macro2	I/O - Mix	11%	100%	Log
Macro3	I/O - Level Out	-96.0dB	0.0dB	Lin
Macro4	Rev - Buildup	22%	100%	Exp
Macro1	Color - Low Cut G	20Hz	20000Hz	Log
Macro2	Rev - Diffusion	10%	100%	Lin
Macro3	Time - PreDelay	8ms	250ms	Lin
Macro4	I/O - Image In	-360	360	Lin

- **MIN:** assigns the minimum value the MACRO can apply to the destination. This value will change depending on the selected parameter. If the min value is set higher than the maximum value the macro control will move the parameter in the opposite direction.
- **MAX:** assigns the maximum value the MACRO can apply to the destination. This value will change depending on the selected parameter.
- **CURVE:** changes the response curve between the source and the destination.

There are three available curves:

- **LIN (Linear):** provides a linear response between the source and the destination.
- **LOG (Logarithmic):** applies a logarithmic scale to the response between the source and destination.
- **EXP (Exponential):** provides an exponential scale of response between the source and the destination.

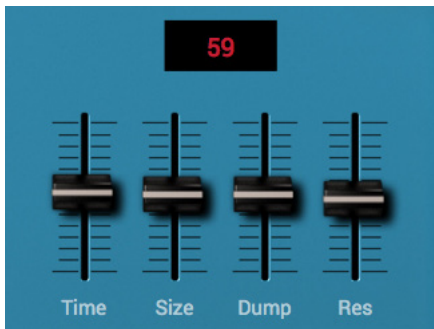


4.17.10 - MACRO NAME

In addition to assigning the MACRO controls in the matrix, you can assign a name to the MACRO control that will appear whenever the reverb patch is opened. This is done by clicking the MACROS button in the upper right of the MACRO matrix window .



This will open up the naming window. Now click on one of the boxes under “LABEL NAME”. This will open up a new dialog box. Type in a short name and click OK. Now this name will appear on the front interface of CSR after you save the preset.



4.18 - CSR Hall Reverb

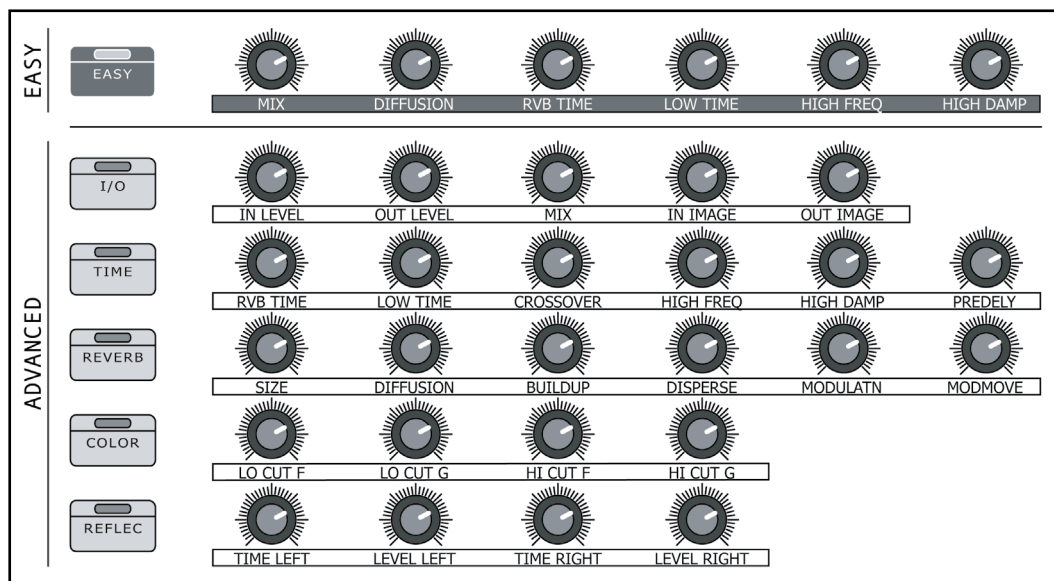


Hall reverb is a simulation of a large acoustic hall. This type of effect has been widely used to create large ambient spaces. Digital reverbs made this sound popular, simply because it was not until the introduction of digital technology that such complex artificial reverbs were possible, unless you had the keys to your local cathedral or concert hall...

Many people would describe this as a vocal reverb. This is useful on vocals if you are looking for a thick, full reverb. This is typically a very pronounced reverb effect. It often adds a "larger than life" characteristic to the affected sound. Hall reverbs are useful in creating a large space around any musical sound, as well as very useful for strange or moody film sound effects.

There are two operating modes for this module: "Easy" for a basic control of the most commonly used parameters, and "Advanced" to tweak every detail and have deep control of the reverb shape and tone.

The diagram below shows the controls breakdown of Classik Studio Reverb's "Classik Hall".



4.18.1 - Easy

- **MIX:** controls the ratio of reverb and original sound. 0% is all unprocessed sound. 100% is all reverb. Use this control to increase or decrease the amount of reverb in the sound. If you are using your reverb as a send effect, this control will probably be set to 100%.
- **DIFFUSION:** determines the rate that the echo density increases over time. A small diffusion setting simulates an acoustic space with smooth walls, like a bathroom. A larger diffusion setting creates a more open and dense sounding reverb.
- **RVB TIME (Reverb Time):** adjusts the length of the reverb. This is a commonly used control that shortens and lengthens the time that the reverb tail lasts.
- **LOW TIME:** controls the length of the reverb for low frequencies (determined by the Crossover parameter) of the RVB Time.
- **HIGH FREQ (High Frequency):** changes the frequency of high frequency dampening.
- **HIGH DAMP (High Dampening):** shortens the high frequency decay time of the reverb. This allows you to emulate the way different materials in a real room, such as carpet and furnishings, absorb sound.

4.18.2 - I/O Levels

- **IN LEVEL:** adjusts the input level of the reverb.
- **OUT LEVEL:** adjusts the output level from the reverb.
- **MIX:** controls the ratio of reverb and original sound. 0% is all unprocessed sound. 100% is all reverb. Use this control to increase or decrease the amount of reverb in the sound. If you are using your reverb as a send effect, this control will probably be set to 100%.
- **IN IMAGE:** controls the stereo image of the input signal before it is processed through the effect. This control allows you to change the signal from normal stereo to mono and manipulate the phase of the signal.

For example a value of -45 swaps the left and right channels, 0 is mono, and 45 is normal stereo. With a setting of 90, all mono information will be excluded from the reverb sound.

- **OUT IMAGE:** controls the stereo image of the effected signal. With a value of -360 to +360, this control extends the stereo field of the processed sound. This control can be used to create mono, stereo, or surround type effects. It can also be changed in real-time or automated using the modulation or macro matrix.

4.18.3 - Time

- **RVB TIME (Reverb Time):** adjusts the length of the reverb. In other words, this shortens and lengthens the time that the reverb tail lasts.
- **LOW TIME:** controls the length of the reverb for low frequencies (determined by the Crossover parameter) of the RVB Time.
- **CROSSOVER:** changes the crossover frequency for the Low time control.
- **HIGH FREQ (High Frequency):** changes the frequency for the low-pass filter used for High Frequency damping.
- **HIGH DAMP (High Dampening):** damps (attenuates) the high frequency being recirculated into the reverberation. Using High frequency damping often gives a more natural sounding reverb.
- **PREDELY (Predelay):** determines the amount of time between the initial input signal and the reverberation. Increasing the predelay can create a sense of distance and, to some degree, change the size of the acoustic space.

4.18.4 - Reverb

- **SIZE:** controls the approximate size of the acoustic space from small to large. This control is measured in meters and is approximately equal to the longest dimension of the space.
- **DIFFUSION:** determines the rate that the echo density increases over time. A small diffusion setting simulates an acoustic space with smooth walls, like a bathroom. A larger diffusion setting creates a more open and dense sounding reverb.
- **BUILDUP:** changes the speed of the reverb's buildup. When this control is turned down, it causes a fast attack in the reverb. To create a slower buildup in the reverberation, turn up the buildup control. Increasing the Buildup will create the sense of a large room/hall, but the spatial information can be lost - this can be restored with the reflection parameters (delay and level).
- **DISPERSE:** disperse works in conjunction with the Buildup control to determine the initial envelope of the reverberation. You can use Disperse to vary the apparent size and character of the early reflection.
- **MODULATN (Modulation):** creates movement in the reverberation tail. Modulation continually changes the timbre of the reverberation. If used in smaller amounts, 10%-50%, it can result in a more natural sounding reverb. If used higher than that, it can cause acoustic instruments timbre to become unstable.
- **MODMOVE (Modulation Movement):** changes the time that early reflection will move in the reverberation. This can be used to create more accurate spatialization of the reverb effect.

4.18.5 - Color

- **LO CUT F (Low Cut Frequency):** changes the frequency of the low cut filter in the reverberation path. This control ranges from 10Hz to 20 kHz.
- **LO CUT G (Low Cut Gain):** changes the gain of the low cut filter. This is adjustable from -24dB to 0dB.
- **HI CUT F (High Cut Frequency):** changes the frequency of the high cut filter in the reverberation path. This control ranges from 10Hz to 20 kHz.
- **HI CUT G (High Cut Gain):** changes the gain of the high cut filter. This is adjustable from -24dB to 0dB.

4.18.6 - Reflections

- **TIME LEFT:** changes the delay between the input signal and the early reflections in the left channel.
- **LEVEL LEFT:** changes the level of the reflections in the left channel.
- **TIME RIGHT:** changes the delay between the input signal and the early reflections in the right channel.
- **LEVEL RIGHT:** changes the level of the reflections in the right channel.

4.18.7 - Modulation Matrix, MARCOS and MACRO NAME

This section of controls is common to all the the CSR modules. Refer to paragraphs from 4.17.8 to 4.17.10 for detailed description.

4.19 - CSR Room Reverb

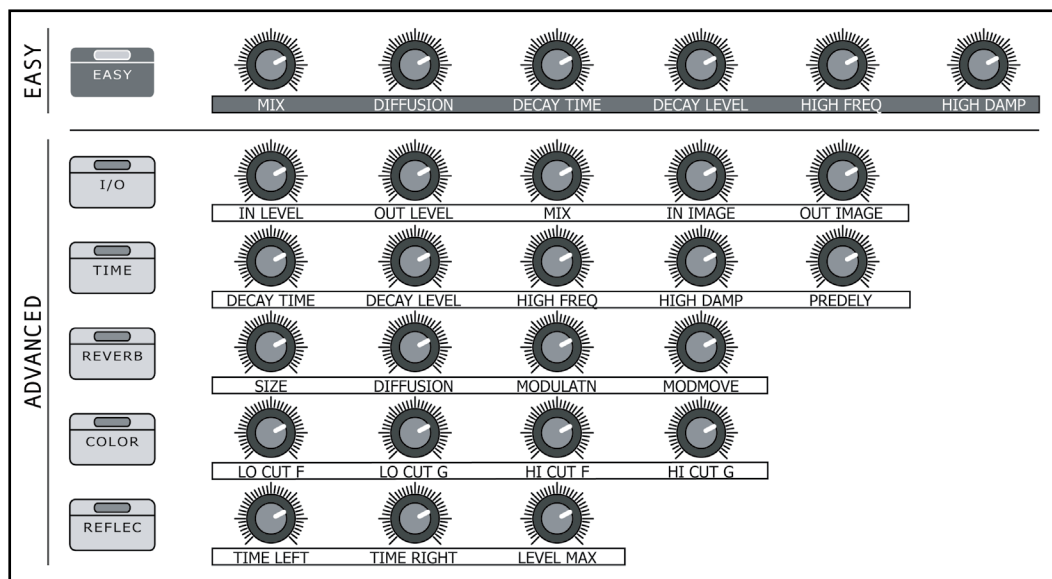


Room reverb is often used to create the ambience of a small to medium sized room. This type of reverb is very effective on any instrument where you want to add the illusion of recording the sound with room ambience. This might seem like a simple thing, but in reality most close-miked instruments lack very much ambience in the recorded sound. Adding a little bit of room reverb can make the sound sit nicer in the mix and give the illusion that the sound came from a nice sounding room, even if it was recorded in a small boxy closet!

This is useful on any sound needing a small to medium size acoustic space. Often used on drums, acoustic guitar, and even vocals, this is a very effective reverb sound on many different instruments.

There are two operating modes for this module: "Easy" for a basic control of the most commonly used parameters, and "Advanced" to tweak every detail and have deep control of the reverb shape and tone.

The diagram below shows the controls breakdown of Classik Studio Reverb's "Classik Room".



4.19.1 - Easy

- **MIX:** controls the ratio of reverb and original sound. 0% is all unprocessed sound. 100% is all reverb. Use this control to increase or decrease the amount of reverb in the sound. If you are using your reverb as a send effect, this control will probably be set to 100%.
- **DIFFUSION:** determines the rate that the echo density increases over time. A small diffusion setting simulates an acoustic space with smooth walls, like a bathroom. A larger diffusion setting creates a more open and dense sounding reverb.
- **DECAY TIME:** controls the length of the reverb tail.
- **DECAY LVL (Decay Level):** controls the level of the reverb tail. When turned off, only the early reflections are in the signal.
- **HIGH FREQ (High Frequency):** changes the frequency of high frequency damping.
- **HIGH DAMP (High Dampening):** shortens the high frequency decay time of the reverb. This allows you to emulate the way different materials in a real room, such as carpet and furnishings, absorb sound.

4.19.2 - I/O Levels

- **IN LEVEL:** adjusts the input level of the reverb.
- **OUT LEVEL:** adjusts the output level from the reverb.
- **MIX:** controls the ratio of reverb and original sound. 0% is all unprocessed sound. 100% is all reverb. Use this control to increase or decrease the amount of reverb in the sound. If you are using your reverb as a send effect, this control will probably be set to 100%.
- **IN IMAGE:** controls the stereo image of the input signal before it is processed through the effect. This control allows you to change the signal from normal stereo to mono and manipulate the phase of the signal. For example a value of -45 swaps the left and right channels, 0 is mono, and 45 is normal stereo. With a

setting of 90, all mono information will be excluded from the reverb sound.

- **OUT IMAGE:** controls the stereo image of the effected signal. With a value of -360 to +360, this control extends the stereo field of the processed sound. This control can be used to create mono, stereo, or surround type effects. It can also be changed in real-time or automated using the modulation or macro matrix.

4.19.3 - Time

- **DECAY TIME:** controls the length of the reverb tail.
- **DECAY LVL (Decay Level):** controls the level of the reverb tail. When turned off, only the early reflections are in the signal.
- **HIGH FREQ (High Frequency):** changes the frequency for the low-pass filter used for High Frequency damping.
- **HIGH DAMP (High Dampening):** dampens (attenuates) the high frequency being recirculated into the reverberation. Using High frequency damping often gives a more natural sounding reverb.
- **PREDELY (Predelay):** determines the amount of time between the initial input signal and the reverberation. Increasing the predelay can create a sense of distance and, to some degree, change the size of the acoustic space.

4.19.4 - Reverb

- **SIZE:** controls the approximate size of the acoustic space from small to large. This control is measured in meters and is approximately equal to the longest dimension of the space.
- **DIFFUSION:** determines the rate that the echo density increases over time. A small diffusion setting simulates an acoustic space with smooth walls, like a bathroom. A larger diffusion setting creates a more open and dense sounding reverb.
- **MODULATN (Modulation):** creates movement in the reverberation tail. Modulation continually changes the timbre of the reverberation. If used in smaller amounts, 10%-50%, it can result in a more natural sounding reverb. If used higher than that, it can cause acoustic instruments timbre to become unstable.
- **MODMOVE (Modulation Movement):** changes the time that early reflection will move in the reverberation. This can be used to create more accurate spatialization of the reverb effect.

4.19.5 - Color

- **LO CUT F (Low Cut Frequency):** changes the frequency of the low cut filter in the reverberation path. This control ranges from 10Hz to 20 kHz.
- **LO CUT G (Low Cut Gain):** changes the gain of the low cut filter. This is adjustable from -24dB to 0dB.
- **HI CUT F (High Cut Frequency):** changes the frequency of the high cut filter in the reverberation path. This control ranges from 10Hz to 20 kHz.
- **HI CUT G (High Cut Gain):** changes the gain of the high cut filter. This is adjustable from -24dB to 0dB.

4.19.6 - Reflections

- **TIME LEFT:** changes the delay between the input signal and the early reflections in the left channel.
- **LEVEL LEFT:** changes the level of the reflections in the left channel.
- **TIME RIGHT:** changes the delay between the input signal and the early reflections in the right channel.
- **LEVEL RIGHT:** changes the level of the reflections in the right channel.

4.19.7 - Modulation Matrix, MARCOS and MACRO NAME

This section of controls is common to all the the CSR modules. Refer to paragraphs from 4.17.8 to 4.17.10 for detailed description.

4.20 - CSR Inverse Reverb

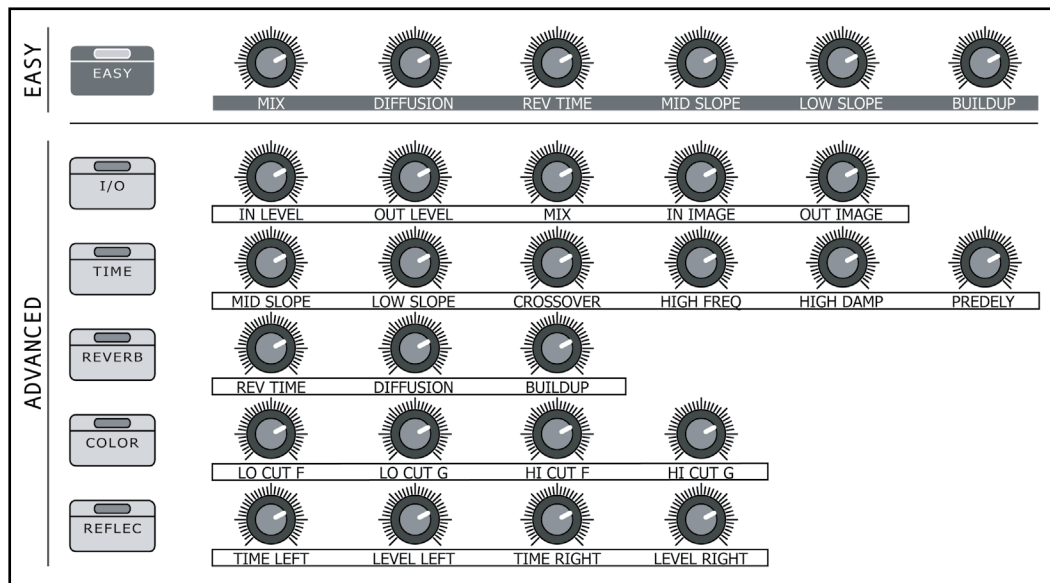


Reverse reverbs were popularized in the 1980's. This type of reverb can be heard on many rock snare drums from this time period. A Reverse reverb applies an envelope (slow attack, fast decay) to the early reflections. The main parameter that affects the sound of this reverb is time. Time can be adjusted for the buildup and cut off of the early reflections. This affects the sound as if it were being reversed in the reverberation, but in actuality it is simply an envelope effect.

This type of reverb has been heavily used on drums and percussion. It is also useful with vocals or any sound that needs an unnatural but interesting ambience. Try combining this reverb with another reverb effect to create a complex moving ambience around the sound.

There are two operating modes for this module: "Easy" for a basic control of the most commonly used parameters, and "Advanced" to tweak every detail and have deep control of the reverb shape and tone.

The diagram below shows the controls breakdown of Classik Studio Reverb's "Classik Inverse".



4.20.1 - Easy

- **MIX:** controls the ratio of reverb and original sound. 0% is all unprocessed sound. 100% is all reverb. Use this control to increase or decrease the amount of reverb in the sound. If you are using your reverb as a send effect, this control will probably be set to 100%.
- **DIFFUSION:** determines the rate that the echo density increases over time. A small diffusion setting simulates an acoustic space with smooth walls, like a bathroom. A larger diffusion setting creates a more open and dense sounding reverb.
- **REV TIME (Reverb Time):** adjusts the length of the reverb. This is a commonly used control that shortens and lengthens the time that the reverb tail lasts.
- **MID SLOPE:** controls the middle and high frequency envelopes. The actual frequencies affected are determined by the Crossover. Positive values will cause the middle to high frequencies to swell from soft to loud over time. The higher the value the more pronounced the swell. Negative values make the mid to high frequencies become quieter over time. More extreme settings will make this attenuation more prominent.
- **LOW SLOPE:** controls the shape of the low frequency envelope for the reverb. A value above 0 creates a smooth rise from soft to loud over time. A negative value causes the low frequencies in the reverb to become quieter over time before dropping off.
- **BUILDUP:** changes the speed of the reverb's buildup. When this control is turned down, it causes a fast attack in the reverb. To create a slower buildup in the reverberation, turn up the buildup control. Increasing the Buildup will create the sense of a large room/hall, but the spatial information can be lost - this can be restored with the reflection parameters (delay and level).

4.20.2 - I/O Levels (Advanced Mode)

- **IN LEVEL:** adjusts the input level of the reverb.
- **OUT LEVEL:** adjusts the output level from the reverb.
- **MIX:** controls the ratio of reverb and original sound. 0% is all unprocessed sound. 100% is all reverb. Use this control to increase or decrease the amount of reverb in the sound. If you are using your reverb as a send effect, this control will probably be set to 100%.
- **IN IMAGE:** controls the stereo image of the input signal before it is processed through the effect. This control allows you to change the signal from normal stereo to mono and manipulate the phase of the signal. For example a value of -45 swaps the left and right channels, 0 is mono, and 45 is normal stereo. With a setting of 90, all mono information will be excluded from the reverb sound.
- **OUT IMAGE:** controls the stereo image of the effected signal. With a value of -360 to +360, this control extends the stereo field of the processed sound. This control can be used to create mono, stereo, or surround type effects. It can also be changed in real-time or automated using the modulation or macro matrix.

4.20.3 - Time

- **MID SLOPE:** controls the middle and high frequency envelopes. The actual frequencies affected are determined by the Crossover. Positive values will cause the middle to high frequencies to swell from soft to loud over time. The higher the value the more pronounced the swell. Negative values make the mid to high frequencies become quieter over time. More extreme settings will make this attenuation more prominent.
- **LOW SLOPE:** controls the shape of the low frequency envelope for the reverb. A value above 0 creates a smooth rise from soft to loud over time. A negative value causes the low frequencies in the reverb to become quieter over time before dropping off.
- **CROSSOVER:** changes the crossover frequency between the LOW and MID DAMP controls.
- **HIGH FREQ:** changes the frequency for the low-pass filter used for High Frequency damping.
- **HIGH DAMP:** damps (attenuates) the high frequency being recirculated into the reverberation. Using High frequency damping often gives a more natural sounding reverb.
- **PREDELAY (Predelay):** determines the amount of time between the initial input signal and the reverberation. Increasing the predelay can create a sense of distance and, to some degree, change the size of the acoustic space.

4.20.4 - Reverb

- **REV TIME (Reverb Time):** adjusts the length of the reverb. In other words, this shortens and lengthens the time that the reverb tail lasts.
- **DIFFUSION:** determines the rate that the echo density increases over time. A small diffusion setting simulates an acoustic space with smooth walls, like a bathroom. A larger diffusion setting creates a more open and dense sounding reverb.
- **BUILDUP:** changes the speed of the reverb's buildup. When this control is turned down, it causes a fast attack in the reverb. To create a slower buildup in the reverberation, turn up the buildup control. Increasing the Buildup will create the sense of a large room/hall, but the spatial information can be lost - this can be restored with the reflection parameters (delay and level).

4.20.5 - Color

- **LO CUT F (Low Cut Frequency):** changes the frequency of the low cut filter in the reverberation path. This control ranges from 10Hz to 20 kHz.
- **LO CUT G (Low Cut Gain):** changes the gain of the low cut filter. This is adjustable from -24dB to 0dB.
- **HI CUT F (High Cut Frequency):** changes the frequency of the high cut filter in the reverberation path. This control ranges from 10Hz to 20 kHz.
- **HI CUT G (High Cut Gain):** changes the gain of the high cut filter. This is adjustable from -24dB to 0dB.

4.20.6 - Reflections

- **TIME LEFT:** changes the delay between the input signal and the early reflections in the left channel.
- **LEVEL LEFT:** changes the level of the reflections in the left channel.
- **TIME RIGHT:** changes the delay between the input signal and the early reflections in the right channel.
- **LEVEL RIGHT:** changes the level of the reflections in the right channel.

4.20.7 - Modulation Matrix, MARCOS and MACRO NAME

This section of controls is common to all the the CSR modules. Refer to paragraphs from 4.17.8 to 4.17.10 for detailed description.

4.21 - White Channel



The White Channel module is based on the EQ and dynamics sections of one of the most widely used, modern large format-British mixing consoles. It can be considered the evolution of the most popular 80's mixing consoles, of which IK has already released a killer module, the British Channel. By the late '90s, this console had become a defining tool for shaping countless hits, particularly in the Hip Hop/R & B music scene, due to its incredibly wide bandwidth and pristine sound. It also has all of the automation features that made it one of the most, if not the most flexible console for mixing jobs in top facilities the world over. IK's module faithfully recreates its very clean and transparent circuit design and the resulting sound and analog feel, with the flexibility of mode-switching EQ and a feature-rich dynamics section.

Controls:

EQ Section:

LOW Band:

- **Gain - dB:** determines the amount of boost or cut in decibels (dB) for this frequency band.
- **Freq - Hz:** sets the frequency in hertz (Hz) at which the boost or cut happens.
- **PEAK:** when this switch is pressed, the LOW band changes from low-shelving type to peaking type with a fixed Q (bandwidth).

LOW-MID Band:

- **Gain - dB:** determines the amount of boost or cut in decibels for this frequency band.
- **Freq - KHz:** sets the frequency in kilohertz (KHz) at which the boost or cut happens.
- **Q:** sets the width of the frequency band where extreme left results in a narrow and sharper band, while extreme right results in a wide and broader band.

HI-MID Band:

- **Gain - dB:** determines the amount of boost or cut in decibels for this frequency band.

- **Freq - KHz:** sets the frequency in kilohertz at which the boost or cut happens.
- **Q:** sets the width of the frequency band where extreme left results in a narrow and sharper band, while extreme right results in a wide and broader band.

HI Band:

- **Gain - dB:** determines the amount of boost or cut in decibels for this frequency band.
- **Freq - KHz:** sets the frequency in kilohertz at which the boost or cut happens.
- **PEAK:** when this switch is pressed, the HI band changes from high-shelving type to peaking type with fixed Q.

FILTERS Section (LOW-PASS/HI-PASS):

- **FLT IN:** this switch enables the filter section, which can be used independently from the rest of the EQ section.
- **Freq - Hz:** sets the frequency of the high-pass filter roll-off point. The filter has a 12 dB/octave cut (18 when in B mode). In the OUT position, the filter is inactive: this effectively allows the use of the two filters independently.
- **Freq - KHz:** sets the frequency of the low-pass filter roll-off point. The filter has a 12 dB/Octave cut. Just like the other, in the OUT position the filter is inactive.
- **FLT S/C:** this switch makes the filter section act as the detector for the sidechain circuit of the dynamics section. When engaged the HP and LP filters are not usable for their primary purpose.
- **IN:** this switch turns the whole EQ section (minus filters) on or off.
- **B:** this switch allows you to choose which EQ mode the White Channel will work in: when disengaged, the EQ behavior will be the one of the “E” EQ circuit in the original console. When engaged, the EQ mode will be the “G” EQ circuit with its distinctive frequency shifts and boost/cut/Q differences. These two different circuit designs offer sonically distinct results, and having them both available at the press of a switch is a very handy resource for ultimate tone tweaking.

DYNAMICS Section:

- **IN:** this switch enables the dynamics section.

GATE/EXP:

- **EQ PRE:** this switch modifies the sequence of EQ and dynamics sections. When engaged the compressor section is shifted to before the EQ section (when disengaged, it's after the EQ section), effectively changing how the whole module reacts to the incoming signal in relation to the parameter settings.
- **EQ S/C:** this switch enables the sidechain circuit, meaning that the EQ section is used as a detector for the dynamics section. This is a very effective way of compressing a particular frequency that is selected with the EQ section. When this switch is enabled the EQ section will not work the standard way and could not be used to EQ the incoming signal (you will have to instantiate another White Channel EQ/Dyn plug-in or use another EQ).
- **THRESHOLD:** this control sets the level (in dB) below which the incoming signal is attenuated by the gate/expander circuit.
- **RANGE:** this parameter controls the amount of gain reduction (in dB) applied to the incoming signal when it falls below the threshold level. The extreme counterclockwise position will have none-to-minimal gain reduction, while more effective gating (for more noisy or problematic tracks) is obtained by progressively

turning this control towards the 40 dB value (fully clockwise).

- **RELEASE - S:** this control determines the time (in seconds) it takes the expander/gate to release the signal to its normal level once it has gone above the threshold.
- **FAST ATT:** this control switches the attack time it takes to the gate/expander circuit to open, to 1ms; this can be useful to produce some very particular effects like snappy and percussive sounds with very marked attack transients.
- **EXP:** this control activates the expansion mode for the gate/expander section.
- **HOLD:** this control sets the amount of time (in ms) the gate will stay open after the signal has fallen below the threshold.

COMP:

- **THRESHOLD:** this control sets the level in dB that must be exceeded by the input signal coming into the White Channel module for the compression to start working. At the 0 position the module is not compressing. Turning the knob clockwise will lower the threshold and, depending on the input signal, start compressing. Turning the knob counterclockwise towards the +10 value will lift the threshold so that hotter signals can be easily managed. This control and its behavior can be affected by the Master section's I/O level controls. As an example, try driving the compressor's input level harder with the Master's section IN knob and set the threshold low for some serious squashing and pumping compression effects. The OUT knob will make calibrating the final output of the module easy in order to avoid clipping. A note on Auto Makeup Gain: this compressor circuit design is provided of an auto makeup gain control; this will progressively compensate the output level in relation to the threshold control avoiding a drop in overall output level as a result of gain reduction. The lower the threshold, the more you will hear it raise the output level, so pay attention about setting a correct gain staging among EQ, Compressor and Master I/O levels.
- **RATIO:** this control determines the gain reduction amount applied to the signal passing above the level set by the threshold control. Lower values of this control (1:1, 1.5:1) will result in no compression or very light compression, while turning the knob clockwise will progressively make for a more pronounced effect as you approach higher values. The extreme clockwise position will result in a limiting effect.
- **FAST ATT:** this switch shifts the attack time it takes the circuit to start compressing once the incoming signal has passed the set threshold to a faster 1 ms value. It can be quite handy to emphasize attack transients on instruments like snare or kick drums and other percussive sources.
- **RELEASE - S:** this control sets the time (in seconds) it takes the compressor to release the signal back to uncompressed state after it has gone below the threshold level. Shorter release times will make the compressor snappy to downright "pumping" when lots of compression is applied. Longer release times will result in a more natural sounding effect and increased sustain on particular instruments.
- **HARD:** this switch changes the detecting mode of the compressor from RMS to peak.

MASTER Section:

- **Ø (Phase):** this control shifts the phase of the incoming signal by 180°, useful to check phase compatibility of sources or to reverse phase in one of two coupled tracks (like snare drum top and bottom).
- **IN:** this control lets you trim the input level of the signal going into the White Channel module. It can be used to correct the level of the incoming signal when needed (in case of a particularly weak or hot track) since it has a range of +/- 15 dB. As described above (in the compressor's threshold parameter section) it can be used to drive the compressor section harder by boosting the input signal, thus producing some very particular compression effects, which can be further emphasized by ratio, attack and release time settings.

- **OUT:** this control lets you trim the master output level of the plug in. It can be used to avoid clipping and overloading of the output stage in case some very radical EQ boosting is applied, or to simply control the level of a track that needs to be adjusted.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific compression settings for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo compressor on your processing chain. This means that you will be able to independently compress the Center or the Sides of your stereo program. The L and R icons will be replaced by M and S, when in this mode. Click M to set the compression for the Center (Mid) of your stereo image, or S to set compression for the sides.

4.22 - Bus Compressor



The Bus Compressor is modeled after the stereo master-bus compressor built into one of the most popular British analog mixing consoles, which was a staple in the 80's and 90's pop and rock music scene (the same console our acclaimed British Channel is derived from). This compressor was so in-demand that it was also released as a separate, outboard rack unit. Due to its VCA circuit design and operation, it is known to impart a very defined sonic signature to the audio program, making all tracks in a mix “blend” together in a very musical way. As a result, its nickname among the audio engineering and recording studio insiders is the “glue compressor.” Most often used on the mix bus or on the drum sub-master, this is an essential piece of kit to use for imparting punch and boldness to any mix.

Controls:

- **THRESHOLD:** this control sets the minimum level above which the signal is compressed. Lower settings of this control result in a more compressed signal. This control has no fixed settings; it offers continuously variable values from +20 to -20.
- **RATIO:** this control sets the behavior of the compressor related to the signal hitting the threshold. A setting of 4:1 means that signal exceeding the set threshold will be reduced by four times (4 dB over the threshold will be reduced to 1dB).
- **ATTACK:** this control sets the time it takes the compressor to start its action once the signal has exceeded the threshold.
- **RELEASE:** this control sets the time it takes the compressor to release the signal back to its uncompressed state once its level falls below the set threshold.
- **MAKEUP:** this control is used to compensate for the loss of level in the program that can be produced by particularly heavy compression settings.
- **SC HI PASS:** this control sets the corner frequency (in Hz) for the hi-pass filter, which can be used before the detection circuit of the compressor to prevent too much bass-frequency content from modulating the compression. The hi-pass filter slope is a fixed 24 db/octave. This control is particularly useful when compressing mixes with heavy bass content. Setting it fairly high prevents elements like kick/bass from

making the compressor “pump”. At lower settings, “pumping” (which is a popular effect in dance and electro mixes) is more likely.

- **COMPRESSION IN:** this control turns the compression circuit ON or OFF.
- **GRIT:** this control engages second harmonic distortion when pressed. It is a very subtle effect but perceivable particularly with “cleaner” mix material. When engaged, the compressor behaves like an out-of-calibration hardware unit, imparting a gritty sound that has become popular among recording engineers.
- **RESET:** this switch brings the module back to its default state when pressed.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific compression setting for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo compressor on your processing chain. This means that you will be able to independently compress the center or the sides of your stereo program. The L and R icons will be replaced by M and S when in this mode. Click M to set the compression for the Center (mid) of your stereo image or S to set compression for the sides.

4.23 - Tape Echo (EchoFlex)



The Tape Echo (EchoFlex) is based on the classic hardware Echoplex EP-3 tape echo unit, which was one of the first echo devices used in audio applications and musicians' setups. Based on a standard 1/4" magnetic tape reel, just like the one used in standard studio multitrack recorders, it uses 3 tape heads, one of which can be moved with the linked "time bar," to set the length of the echo produced. These factors contribute to the classic "vintage" sound of the analog echo, which has a natural high-frequency damping as the repeats decay. The simple interface makes it a very intuitive and fast echo device, which can also be used for very creative effects when pushed to extreme settings. The Tape Echo has been modeled down to the tape behavior and anomalies like wow and flutter, tape wear, hum/noise and rec level (for tape saturation), putting these highly flexible and creative parameters at the user's disposal.

Controls:

- **INPUT LEVEL:** this control sets the input level of the signal going into the module. Boosting the level to +15 clips the input stage electronics creating analog saturation.
- **VOLUME A/B:** these controls regulate the level of the delayed signal independently for the two channels (dry/wet ratio). When A or B control is set to minimum the signal is completely dry, meaning that the tape effect is bypassed and the signal goes through preamp electronics only.
- **SUSTAIN A/B:** these controls regulate the decay /repeats of the delayed signal independently for the two channels. Setting decay to maximum sends the unit into feedback, which can be used to create some wild effects.
- **TIME Bar:** this control sets the delay time for the A and B channels. When the module is set for A/B independent operation, the bar will switch on the A or B setting according to what channel is selected by the MODE switch. When the BPM sync switch is set to ON, the Time Bar will switch from continuously variable to fixed steps that correspond to the time divisions described in the Echo Time display. Moving the Time Bar during playback creates a "glide" effect of tape speeding up and slowing down.
- **ECHO TIME Display:** this display shows the time set for each channel in milleseconds (ms), moving with the time bar. When the Mode switch is set to A or B the value shown will correspond to the one set for that

channel. When the BPM Sync switch is set to ON, the shown value will change from ms to time divisions. Possible values are: 1/1, 1/2d, 1/2t, 1/2, 1/4d, 1/4t, 1/4, 1/8d, 1/8t, 1/8, 1/16d, 1/16t, 1/16, 1/32.

- **MODE Switch:** this 3-position switch (A, B, STEREO) determines the behavior of the module with relation to the delay time. When set to STEREO, delay time for both channels is the same, and moving the time bar will change both. When set to A or B the delay times are independent for each channel. When set to A, the Time Bar will show the A-channel delay time; setting it to B will show the corresponding B-channel delay time.
- **BPM SYNC Switch:** this switch will make the delay times sync to the BPM set in the host DAW session.
- **ECHO Switch:** this switch is the master bypass of the module.
- **Meter Select Switch (IN/OUT):** this control will make the meter show the input or output level according to the position selected.
- **RESET Switch:** this control will reset the module back to its default state. Clicking a second time will bring the module to its last setting.
- **ECHOFLEX Logo:** clicking on this logo will open the hatch underneath showing the trim pots. Clicking on the border of the open hatch will close it.



- **REC LEVEL Trim Pot:** this control adjusts the level of the signal hitting the tape. At the highest settings the tape will start to saturate in a more evident manner, an effect which can be used to create interesting results.
- **TAPE WEAR Trim Pot:** this control adjusts the high frequency damping due to tape wear and use.
- **WOW/FLUTTER Trim Pot:** this control adjusts the global wow and flutter effect typical of an out aligned tape machine. At the most extreme setting the effect is particularly aggressive with pitch and speed changes (just like in the real unit), while at the minimum setting the effect is that of a perfectly aligned tape.
- **NOISE Trim Pot:** this control adjusts the amount of tape hiss and preamp noise in the signal. At the most extreme setting also AC line hum will become very audible; the hum's frequency can be selected to reflect either US 60Hz or EU 50Hz.

4.24 - Precision Comp/Limiter



The Precision Comp/Limiter is based on one of the legendary British compressors found in top-level studios worldwide. Originally produced in the '70's, it quickly became a much sought after unit for many audio applications, including music recording/mixing, mastering, post-production and broadcasting. Its sonic character is not as aggressive as that of many other popular units of the same era. Even when the unit is pushed to extreme settings and despite its fixed 1ms attack time, the compression is fairly gentle and retains a musical quality that doesn't "choke" or squash program material. Nevertheless, the compression effect is clearly distinguishable. When the limiter is kicked in, the gain reduction becomes more pronounced, especially with lower limiter thresholds. The compressor and limiter can operate independently from each other. The compression is achieved via a set of diodes, thus the original hardware's name "discreet". The overall effect is a nice "fat and thick" sound (a trademark of this British manufacturer), which doesn't lose focus or detail. These characteristics are also aided by the set of transformers used in the signal path.

Controls:

COMPRESSOR:

- **INPUT dB:** sets the input level in dB, and is useful for fine-tuning the level hitting the threshold. The minimum resolution for each step is 1.5 dB.
- **THRESHOLD dBU:** sets the level above which compression takes place; the lower the threshold the more compressed the signal.
- **RECOVERY ms:** sets the time it takes the compressor to release the signal back to its uncompressed state; available values are: 100ms, 400ms, 800ms, 1500ms, Auto 1 and Auto 2.
- **GAIN:** sets the amount of makeup gain applied to the compressor's circuit output. Use it to compensate for gain loss from the compression effect.
- **RATIO:** sets the amount of dBs the signal is reduced by once it gets past the set threshold. A ratio of 3:1 means that if the signal is exceeding the threshold by 3 dB it will be reduced to 1 dB.
- **COMP IN:** this switch engages the compression circuit.

LIMITER:

- **THRESHOLD dBU:** sets the level above which limiting takes place; the lower the threshold the more compressed the signal.
- **RECOVERY ms:** sets the time it takes the limiter to release the signal when its level falls below the set threshold; available values are: 50ms, 100ms, 200ms, 800ms, Auto 1 and Auto 2.
- **OUTPUT dB:** sets the output level of the limiter section. It's useful for raising the signal level after gain reduction.
- **ATTACK FAST/SLOW:** this switch allows you to select between 2 different attack times, a slower 2ms and a faster 1ms.
- **LIMIT IN:** this switch engages the limiter circuit.

MASTER Section:

- **POWER:** turns the module on or off, effectively bypassing it.
- **RESET:** resets all parameters to their default status once clicked.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific compression settings for L and R channels independently.
- **M/S:** this option uses a mid-side matrix to insert the stereo compressor on your processing chain. This means that you will be able to independently compress the center or the sides of your stereo program. The L and R icons will be replaced by M and S when in this mode. Click M to set the compression for the center (mid) of your stereo image or S to set compression for the sides.

4.25 - Master EQ 432



This module is based on what is considered to be the “golden reference” standard in mastering equalizers. This piece of hardware was the brainchild of two of the most ingenious minds in music recording technology, true pioneers who joined forces to create what was the first parametric EQ ever invented. The design quickly became the basis for subsequent parametric EQs from manufacturers the world over. On top of the fully parametric and totally independent bands with almost no interaction (something that was not available until this circuit topology invention), its success came from its no-compromise design based on discrete, high-performance, custom op-amps and a transformerless signal path yielding pristine, transparent audio. This was a key factor in defining it as the first-call EQ among mastering engineers which it still is today. This module is truly “as good as it gets,” the secret weapon to give mixes that master-quality final polish with unrivaled high end air, solid low end and extreme detail in the critical midrange spectrum.

Controls:

Shelving Bands:

- **HI SHELIVING:** determines the gain of the HI shelving band. The frequency of the band is fixed at 10 KHz.
- **LO SHELIVING:** determines the gain of the LOW shelving band. The frequency of the band is determined by the selector.
- **LO SHELIVING Frequency Selector:** sets the frequency of the low shelving band. The settings are 50Hz or 100Hz.

LO Parametric Band:

- **Level:** sets the boost or cut (in dB) of the low parametric band.
- **Frequency:** sets the frequency of the low parametric band.
- **Shape:** sets the Q of the LO parametric band. The settings are 5, 6, 9, 11, 15 dB/Octave.

MID Parametric Band:

- **Level:** sets the boost or cut (in dB) of the mid parametric band.
- **Frequency:** sets the frequency of the mid parametric band.
- **Shape:** sets the Q of the mid parametric band. The settings are 5, 6, 9, 11, 15 dB/Octave.

HI Parametric Band:

- **Level:** sets the boost or cut (in dB) of the high parametric band.
- **Frequency:** sets the frequency of the high parametric band.
- **Shape:** sets the Q of the high parametric band. The settings are 5, 6, 9, 11, 15 dB/Octave.

Other:

- **PUSH EQ:** works as the global ON/OFF switch for the module.
- **RANGE dB:** sets the range of the gain knobs for all frequency bands; possible values are +/- 6, +/- 9, +/- 12 dB. This allows to have a different resolutions per step of the gain knobs in case finer adjustments are needed.
- **OUT:** sets the boost or cut to the output level (in dB) of the module.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific EQ settings for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo EQ in your processing chain. This means that you can independently equalize the Center or the Sides of your stereo program. The L and R icons are replaced by M and S in this mode. Click M to set the EQ for the Center (Mid) of your stereo image or S to set EQ for the sides.

4.26 - EQ 73



The EQ 73 is based on what is probably acknowledged as the most universally used preamp in music recording history. This class A device has a sonic signature that is almost instantly recognizable to engineers and recording enthusiasts the world over. Thanks to this particular circuitry the signal passing through it becomes thick and bold making every instrument stand out with fullness. To bridge the classic sound with modern functionalities the control layout was improved over the original design, and a few features were added like the preamp circuit saturation.

Controls:

- **HI SHELving:** determines the gain of the HI shelving band. The frequency of the band is fixed at 10 KHz.
- **IN:** sets the input level of the modeled mic preamp section. Increasing gain will result in more level and progressively more saturation of the processed signal, up to a gritty level of distortion. The range is +/-20dB.
- **OUT:** sets the output level of the module. It can be used to compensate the variability of the input level in both positive or negative. The range is +/-20dB.
- **Hi-Pass Filter:** sets the frequency of the Hi-Pass filter. The slope of the filter is 12 dB/Oct. Available frequencies are: 50Hz, 80Hz, 160Hz, 300Hz. OFF position switches the band out of the signal path.
- **Low Shelf Band Freq:** sets the frequency of the Low Shelf Band. Possible frequencies are: 35Hz, 60Hz, 110Hz, 220Hz. OFF position switches the band out of the signal path.
- **Low Shelf Gain:** sets the gain of the Low Shelf Band. Range is +/- 16 dB.
- **Presence Band Freq:** sets the frequency of the Presence (mid) band. Possible frequencies are: 360Hz, 700Hz, 1.6KHz, 3.2KHz, 4.8KHz, 7.2KHz. OFF position switches the band out of the signal path.
- **Presence Band Gain:** sets the gain of the Presence (mid) band. Range is +/- 16 dB.
- **High Shelf Gain:** sets the gain of the HI Shelf band. Range is +/- 16 dB. Frequency is fixed to 12KHz.
- **Line/Mic:** select the operative level of the input section.
- **EQ:** this switch activates the EQ controls. When EQ is not active the signal is still passing through the

preamp section.

- **Phase:** inverts the phase of the signal being processed by the module.
- **ON:** turns on or off the whole module.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific EQ settings for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo EQ in your processing chain. This means that you can independently equalize the Center or the Sides of your stereo program. The L and R icons are replaced by M and S in this mode. Click M to set the EQ for the Center (Mid) of your stereo image or S to set EQ for the sides.

4.27 - EQ 81



This module is based on another classic piece of hardware whose character was instrumental in shaping the sound of many hit songs from iconic musicians and producers; the EQ 81 can be considered the evolution of its older brother, the EQ 73. There are a few twists which make the EQ 81 capable of delivering a voice of its own. The most evident difference here is in the versatility of the EQ section, sporting four independent bands (high / low shelving, plus two midrange peaking filters) rounded out by high and low pass filters; this module is particularly useful when a more precise intervention on a track is needed, still maintaining tone and organic, analog warmth throughout the signal path. As for the EQ 73, a special attention was given to the perfect matching of the EQ curves, but a step further was taken into the modelling of the analog preamp section in all of its colorful nuances.

Controls:

- **IN:** Sets the input level of the modeled mic preamp section. Increasing gain will result in more level and progressively more saturation of the processed signal, up to a gritty level of distortion. The range is +/-20dB.
- **OUT:** sets the output level of the module. It can be used to compensate the variability of the input level in both positive or negative. The range is +/-20dB.
- **Hi-Pass Filter:** sets the frequency of the Hi-Pass filter. The slope of the filter is 18 dB/Oct. Available frequencies are: 27Hz, 47Hz, 82Hz, 100Hz, 270Hz. OFF position switches the filter out of the signal path.
- **Low-Pass Filter:** sets the frequency of the Low-Pass Filter. The slope of the filter is 12 dB/Oct. Available frequencies are: 3.9KHz, 5.6KHz, 8.2KHz, 12KHz, 18KHz. OFF position switches the filter out of the signal path.
- **Low Band Freq:** sets the frequency of the Low Shelf Band. Possible values are: 33Hz, 56Hz, 100Hz, 180Hz, 330Hz. OFF position switches the band out of the signal path.
- **Low Band Shelf/Bell:** switches the Low Band from shelving to peaking mode with a constant 0.2 Q.
- **Low Band Gain:** sets the gain of the Low Band. Available Range is +/- 15 dB.

- **Low Mid Band Freq:** sets the frequency of the Low Mid Band. Possible values are: 220Hz, 270Hz, 330Hz, 390Hz, 470Hz, 560Hz, 680Hz, 820Hz, 1000Hz, 1200Hz. OFF position switches the band out of the signal path.
- **Low Mid band Gain:** sets the gain of the Low Mid Band. Available range is +/-10 dB.
- **Low Mid Band Hi-Q:** switches the bandwidth of the Low Mid Band from a constant 0.5 to a gain dependant 1.0 with a slightly more pronounced gain range.
- **High Mid Band Freq:** sets the frequency of the High Mid Band. Possible values are: 1.5KHz, 1.8KHz, 2.2KHz, 2.7KHz, 3.3KHz, 3.9KHz, 4.7KHz, 5.6KHz, 6.8KHz, 8.2KHz. OFF switches the band out of the signal chain.
- **High Mid Band Gain:** sets the gain of the High Mid Band. Available range is +/- 10 dB.
- **High Mid Band Hi-Q:** Switches the bandwidth of the High Mid Band from constant 0.3 to a gain dependent 1.0 with a slightly more pronounced gain range.
- **High Band Frequency:** sets the frequency of the High Band. Possible values are: 3.3KHz, 4.7KHz, 6.8KHz, 10KHz, 15KHz. OFF position switches the band out of the signal path.
- **High Band Gain:** sets the gain of the High Band. Available range is +/- 15 dB.
- **High Band Shelf/Bell:** switches the Low Band from shelving to peaking mode with a constant 0.2 Q.
- **Phase:** inverts the phase of the processed signal.
- **EQ:** turns the EQ section on or off; when EQ is off signal is still passing through the preamp section.
- **ON:** turns the whole module on or off.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific EQ settings for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo EQ in your processing chain. This means that you can independently equalize the Center or the Sides of your stereo program. The L and R icons are replaced by M and S in this mode. Click M to set the EQ for the Center (Mid) of your stereo image or S to set EQ for the sides.

4.28 - EQ PA



The EQ PA is based on one of the most famous classic American EQ circuit designs. This three band delivers a thick sound and excellent tone shaping capabilities thanks to it's proportional Q design and added preamp stage which will impart a distinctly “fat and punchy” sonic character.

Controls:

- **LF Gain:** controls the amount of gain of the Low band. Range is +/-12 dBs. The gain is stepped so that resolution is 1dB from 0 to +/-6 dBs and 1.5 dBs from +/-6 to +/-12 dBs. Intermediate steps are not shown on labels but available to the user.
- **LF Frequency:** sets the frequency of the Low band; available values are 30, 40, 50, 100, 200, 300, 400 Hz.
- **LF SHELF:** sets the Low band in peaking or shelving mode.
- **MF Gain:** controls the amount of gain of the Mid band. Range is +/-12 dBs. The gain is stepped so that resolution is 1dB from 0 to +/-6 dBs and 1.5 dBs from +/-6 to +/-12 dBs. Intermediate steps are not shown on labels but available to the user.
- **MF Frequency:** sets the frequency of the Low band; available values are 200, 400, 600, 800, 1500, 3000, 5000 Hz.
- **HF Gain:** controls the amount of gain of the Hi band. Range is +/-12 dBs. The gain is stepped so that resolution is 1dB from 0 to +/-6 dBs and 1.5 dBs from +/-6 to +/-12 dBs. Intermediate steps are not shown on labels but available to the user.
- **HF Frequency:** sets the frequency of the Hi band; available values are 2.5, 5, 7, 10, 12.5, 15, 20 KHz.
- **HF SHELF:** sets the Hi band in peaking or shelving mode.
- **PREAMP:** engages the preamp stage. Adjusting the gain when the preamp is engaged will add the harmonic saturation from this part of the circuitry, whose results can be either a subtle harmonic enhancement to the track or a very pronounced distortion. Use it to taste for many creative results.
- **GAIN:** controls the amount of preamp gain when the “Preamp” stage is engaged. Range is +/- 20 dB. When the Preamp stage is not engaged the Gain knob act as an input level control for the module with the

same +/- 20 dB of range.

- **PHASE:** inverts the phase of the signal going through the module by 180°.
- **OUT:** sets the overall output level of the module; range is +/- 20 dB.
- **BAND PASS:** engages the band-pass filter; the filter high-passes the signal with a broad 4th order high pass at 100 Hz, while the low pass filter starts the 4th order slope at 5 KHz.
- **EQ:** turns the EQ section of the module on or off. GAIN and OUT controls remain active and preamp stage can be engaged as well.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default position.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific EQ settings for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo EQ in your processing chain. This means that you can independently equalize the Center or the Sides of your stereo program. The L and R icons are replaced by M and S in this mode. Click M to set the EQ for the Center (Mid) of your stereo image or S to set EQ for the sides.

4.29 - EQ PB



Similar to its EQ PA brother, this module adds a fourth band for increased flexibility, particularly in the midrange part of the spectrum, where the two overlapping bands can create complex shapes, allowing deep sculpting of a track. Combined with the preamp stage this can be a toneful machine capable to deliver a vast array of tones.

Controls:

- **LF Gain:** controls the amount of gain of the Low band. Range is ± 12 dBs. The gain is stepped so that resolution is 1dB from 0 to ± 6 dBs and 1.5 dBs from ± 6 to ± 12 dBs. Intermediate steps are not shown on labels but available to the user.
- **LF Frequency:** sets the frequency of the Low band; available values are 30, 40, 50, 100, 200, 300, 400 Hz.
- **LF SHELF:** sets the Low band in peaking or shelving mode.
- **LMF Gain:** controls the amount of gain of the Low Mid band. Range is ± 12 dBs. The gain is stepped so that resolution is 1dB from 0 to ± 6 dBs and 1.5 dBs from ± 6 to ± 12 dBs. Intermediate steps are not shown on labels but available to the user.
- **LMF Frequency:** sets the frequency of the Low band; available values are 75, 150, 180, 240, 500, 700, 1000 Hz.
- **HMF Gain:** controls the amount of gain of the Hi Mid band. Range is ± 12 dBs. The gain is stepped so that resolution is 1dB from 0 to ± 6 dBs and 1.5 dBs from ± 6 to ± 12 dBs. Intermediate steps are not shown on labels but available to the user.
- **HMF Frequency:** sets the frequency of the Low band; available values are 800, 1500, 3000, 5000, 8000, 1000, 12500 Hz.
- **HF Gain:** controls the amount of gain of the Hi band. Range is ± 12 dBs. The gain is stepped so that resolution is 1dB from 0 to ± 6 dBs and 1.5 dBs from ± 6 to ± 12 dBs. Intermediate steps are not shown on labels but available to the user.
- **HF Frequency:** sets the frequency of the Hi band; available values are 2.5, 5, 7, 10, 12.5, 15, 20 KHz.
- **HF SHELF:** sets the Hi band in peaking or shelving mode

- **PREAMP:** engages the preamp stage. Adjusting the gain when the preamp is engaged will add the harmonic saturation from this part of the circuitry, whose results can be either a subtle harmonic enhancement to the track or a very pronounced distortion. Use it to taste for many creative results.
- **GAIN:** controls the amount of preamp gain when the “Preamp” stage is engaged. Range is +/- 20 dB. When the Preamp stage is not engaged the Gain knob act as an input level control for the module with the same +/- 20 dB of range.
- **PHASE:** inverts the phase of the signal going through the module by 180°.
- **OUT:** sets the overall output level of the module; range is +/- 20 dB.
- **EQ:** turns the EQ section of the module on or off. GAIN and OUT controls remain active and preamp stage can be engaged as well.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default position.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific EQ settings for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo EQ in your processing chain. This means that you can independently equalize the Center or the Sides of your stereo program. The L and R icons are replaced by M and S in this mode. Click M to set the EQ for the Center (Mid) of your stereo image or S to set EQ for the sides.

4.30 - EQ PG



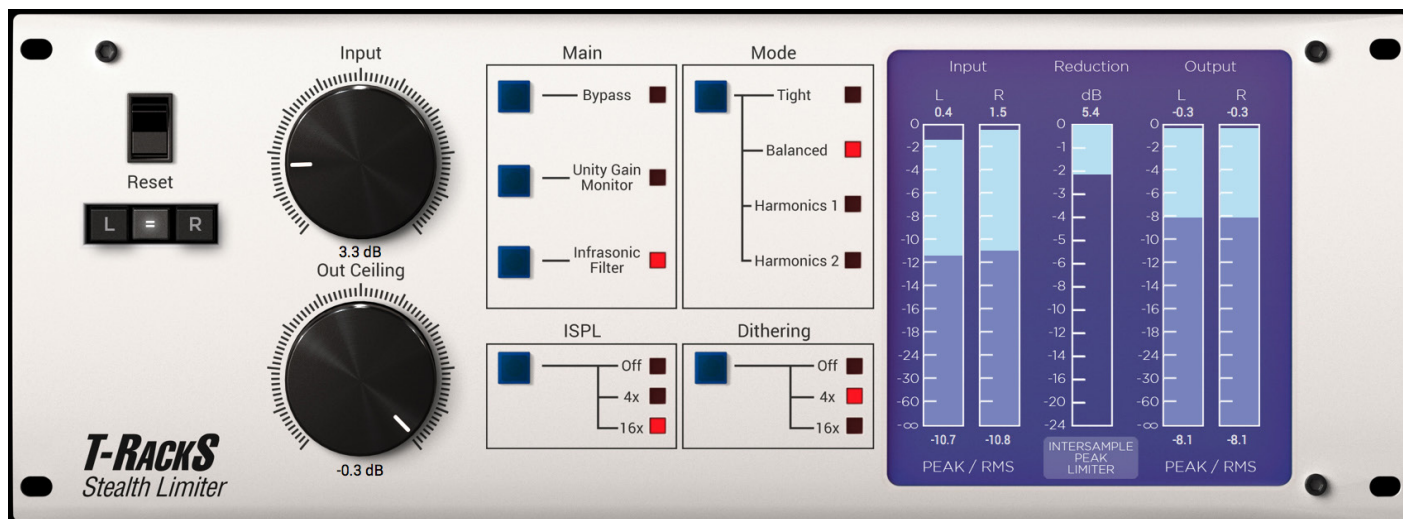
This module uses the same proportional Q approach and all discrete based signal path but in a ten bands graphic fashion instead of a parametric type. This makes for several differences in that the equalization can be applied on larger parts of the spectrum by acting on several bands at once. This, on top of the clear visual shape of the equalization applied, results in massive yet fast tonal shaping capabilities, which combined with the onboard preamp stage makes for a rather powerful tool, beyond simple equalization.

Controls:

- **EQ Bands:** these controls set the gain for the ten frequency bands. Range is +/- 12 dB. Available frequencies are: 30Hz, 63Hz, 125Hz, 250Hz, 500Hz, 1KHz, 2KHz, 4KHz, 8KHz, 16KHz.
- **PREAMP:** engages the preamp stage. Adjusting the gain when the preamp is engaged will add the harmonic saturation from this part of the circuitry, whose results can be either a subtle harmonic enhancement to the track or a very pronounced distortion. Use it to taste for many creative results.
- **GAIN:** controls the amount of preamp gain when the “Preamp” stage is engaged. Range is +/- 20 dB. When the Preamp stage is not engaged the Gain knob act as an input level control for the module with the same +/- 20 dB of range.
- **PHASE:** inverts the phase of the signal going through the module by 180°.
- **OUT:** sets the overall output level of the module; range is +/- 20 dB.
- **EQ:** turns the EQ section of the module on or off. GAIN and OUT controls remain active and preamp stage can be engaged as well.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default position.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific EQ settings for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo EQ in your processing chain. This means that you can independently equalize the Center or the Sides of your stereo program. The L and R icons are

replaced by M and S in this mode. Click M to set the EQ for the Center (Mid) of your stereo image or S to set EQ for the sides.

4.31 - Stealth Limiter



The Stealth Limiter is a mastering peak limiter whose main feature is the extreme transparency of the processing. The technology used under the hood makes it so clean and capable of raising the level of a track without introducing any side effects or distortion, even when very high amounts of gain reduction are applied and where other limiters start to fail.

Controls:

- **INPUT:** sets the input gain that will be added to the program material; basically, how loud you want to be.
- **OUT CEILING:** sets the maximum output level of the processed audio. The default setting for this control is -0.1 dB. We suggest that you never exceed -0.2 dB or -0.1 dB here to preserve the audio and avoid distortion by the audio playback system.

MAIN Section Controls:

- **Bypass:** bypasses the processing.
- **Unity Gain Monitor:** when engaged this control allows you to compare processed and unprocessed sounds at the same level; whatever input gain is added to the audio, it will be compensated in the opposite direction, thus avoiding the pitfall of “louder sounds better” and allowing for a more precise and objective evaluation of the processing applied to the audio.
- **Infrasonic Filter:** this control engages a 4th order high pass filter with a 22Hz cutoff frequency. This is very useful to get rid of infrasonic and very low frequency rumbles which could put excessive workload on the limiter.

METERING Section:

The Stealth Plug limiter includes a full featured level metering system to help evaluating the correct amount of limiting being applied, by just looking at the Limiter interface.

- **INPUT:** this meter shows both Peak and RMS level of the signal that is entering into the limiter. Peak level is shown with the brighter bar and RMS is shown on the darker bar. You can also read instantaneous numeric values for both Peak and RMS levels right above and at the bottom of the meters.
- **OUTPUT:** this meter shows both Peak and RMS level of the signal once boosted by the limiter. Peak level is shown with the brighter bar and RMS is shown on the darker bar. You can also read instantaneous numeric values for both Peak and RMS levels right above and at the bottom of the meters. As a very rough indication, loud masters will show RMS values of -9 / -8 dB RMS. Higher levels than this might result in distorted audio. Consider this as a very rough indication that only applies to commercial music.
- **REDUCTION:** this meter indicated the Peak reduction amount. The more the limiter works the more this meter will illuminate.
- **INTERSAMPLE PEAK LIMITER:** when this sign is displayed the limiter is working in ISPL mode.

MODE Section Controls:

The controls in this section allow you to set the operating mode of the limiter and the kind of processing going on under the hood.

- **Tight:** this is T-RackS Stealth Limiter's default mode, and its most neutral. It respects the source material's natural sonic dynamics the most. Even when set to the highest limiting setting, the music will never sound "compressed" or "constrained." With this setting, you can increase loudness while maintaining a neutral effect on sonic timbre and dynamics.
- **Balanced:** like its name suggests, this is a more balanced processing that can be used when a small amount of dynamic compression is acceptable (or desirable). The difference between this mode and TIGHT is so subtle that the two are only subtly distinguishable at high loudness levels and with extreme limiting.
- **Harmonics 1:** this mode adds the slight harmonic saturation of a Class A tube circuitry to the party, so that some color is added to the music program. Still, being the harmonic enhancement stage "pre" limiting, the transparency of the limiting process is maintained. The amount of harmonic enhancement is determined by the amount of signal that enters into the plug-in. There is no control over this on the limiter itself, so to experiment with more or less "coloration" just adjust the signal level before the Stealth Limiter plug-in.
- **Harmonics 2:** like the Harmonics 1 mode, this mode adds a nice solid state harmonic saturation for a different flavor and slight more "edge" that can be useful to impart a bit more character to a weaker mix. Also for Harmonics 2 mode, the amount of harmonic enhancement is determined by the amount of signal that enters into the plug-in. There is no control over this on the limiter itself, so to experiment with more or less "coloration" just adjust the signal level before the Stealth Limiter plug-in.

ISPL Section Controls (Inter Sample Peak Limiting)

When engaged, this control allows the module to run at an higher sampling rate and enables a special limiting mode that prevents unexpected possible distortion on playback devices even if the output samples level never exceed 0dBFS.

The ISPL limiter can work at two different rates, 4X and 16X.

Use 4X for an optimal balance between intersample peak limiting performance and CPU usage.

16X gives a slightly more safer intersample peaks protection, at the expense of a very high CPU load.

When ISPL is not engaged the limiter runs at the current session's sample rate and it is not guaranteed to protect the program against intersample peaks..

Note on Intersample Peak Limiting

With traditional mastering processes, it's normal to push the volume level towards a -0.1dB or even 0.0dB ceiling. Spikes above this ceiling are suppressed via the use of hard limiters or clippers. Yet these methods only ensure a stable digital level, they do not guarantee undistorted playback once the music is translated into an analog signal.

This is because the process of playback involves a number of conversion processes that, though the digital waveform may be below 0.0dB, can cause the analog waveform to exceed this threshold and introduce unwanted distortion and fuzz.

T-RackS Stealth Limiter's inter-sample peak limiting algorithm gets around this issue by predicting the converted analog waveform of the digital source. Through this process it's able to limit the sound in a way that ensures the final converted audio will never or very rarely exceed the set ceiling. This is useful in the modern era where music is often heard on different devices and via different services that all deliver their music via compressed formats, like MP3 or AAC.

DITHERING Section Controls:

The Stealth Limiter can add a dithering noise to the audio program to make the bit depth reduction that will eventually be applied at a later stage to sound best.

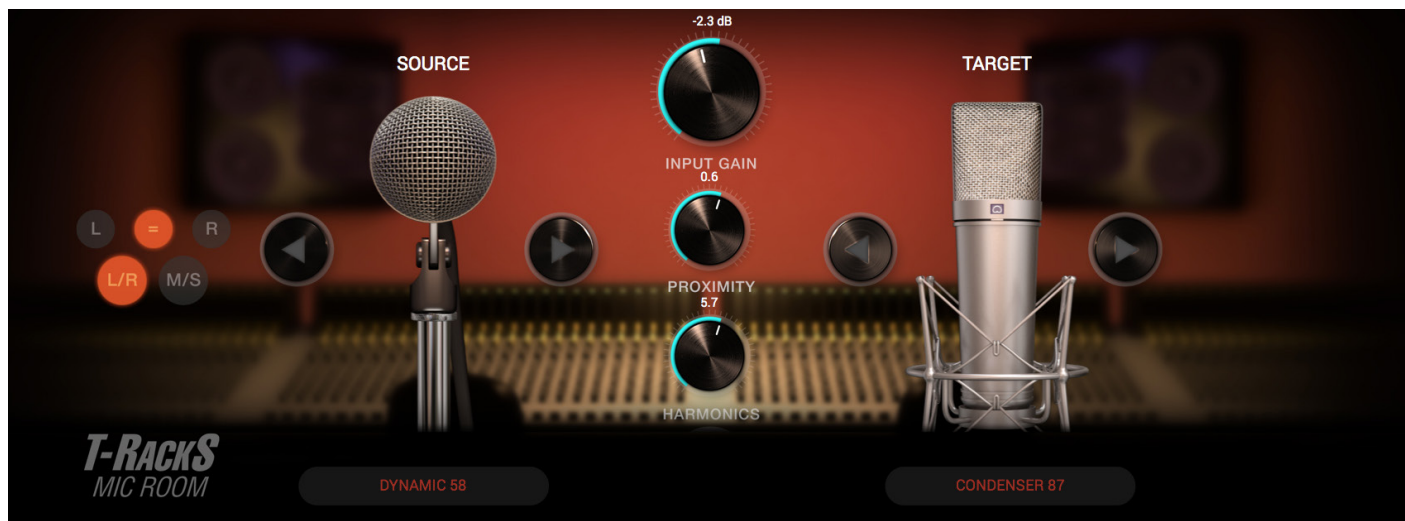
The controls in this section allow to set the level of the dithering noise depending on the bit depth the material will be reduced to (16 or 24 bits).

Please note: the Stealth Limiter **does not** apply any bit reduction or truncation, which has to be applied elsewhere later in the processing chain by the T-RackS standalone application or by the DAW of choice

General Controls:

- **RESET:** brings all controls to the default state.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image. Click on the L or R icons if you want to dial in specific settings for L and R channels independently.

4.32 - Mic Room



Mic Room is a microphone-modeling module based on IK's Multimedia patented technology. It gives you the power to process and “re-mic” your physical microphones to sound and perform like some of the most coveted mic models of all time. A vast selection of the most popular timeless classic and contemporary microphones used in recording studios and live stages around the world has been squeezed into this virtual “mic locker” and made ready for DAW use. A simple and effective approach allows you to have “any mic to any mic” in both source and model fields. This allows for tremendous flexibility when used in the DAW environment. Now, crafting the perfect vocal or instrument sound is possible, even when the original track has been recorded with a less than optimal dynamic, or vice versa, when you need to “grunge up” a too hi-fi track.

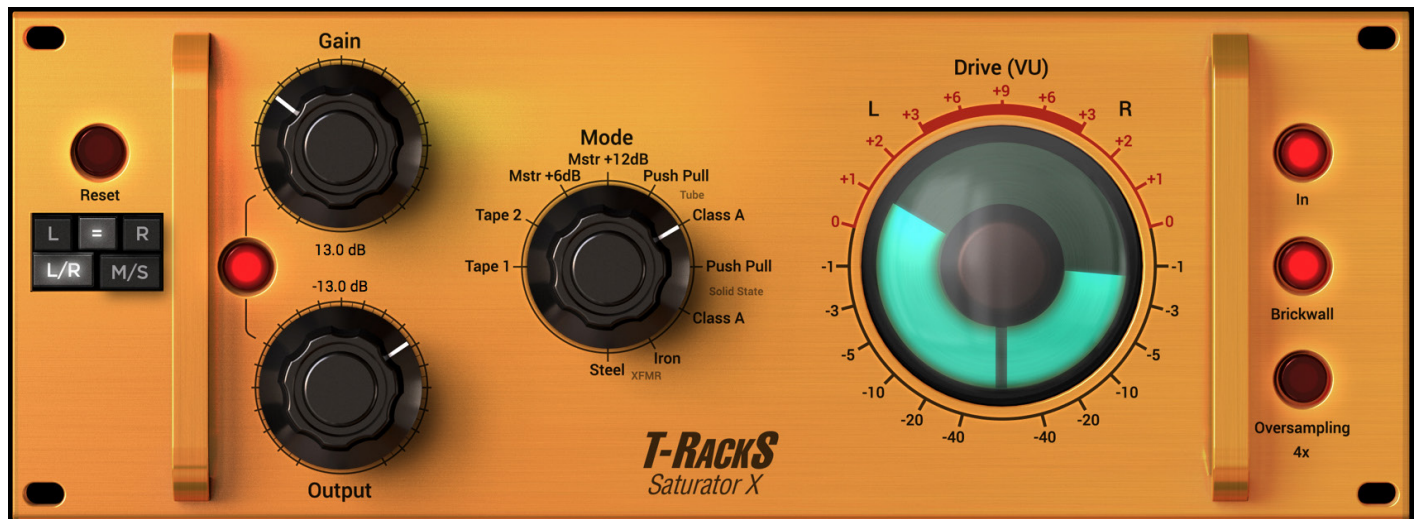
Controls:

- **SOURCE:** selects the mic that was used to record the track you are processing with Mic Room. If such a mic is not in the list, try using a similar kind of microphone (e.g. same diaphragm size, same mic type, condenser, ribbon or dynamic), for a compatible result. Selection can be performed via the microphone field and popup menu, or via the selection arrows.
- **TARGET:** use this control to select the microphone you want the source track to sound like. Selection can be performed via the microphone field and popup menu, or via the selection arrows.
- **INPUT GAIN:** this knob controls the gain applied to the input of the plug-in. Use this to adjust the level of the processed track so that the meter peaks to a ballpark between halfway to three quarters of its length. Range is +/- 24 dBs.
- **PROXIMITY:** this knob controls the amount of the proximity effect which is typical of directional microphones. The closer a source is to the microphone, the more the low end increases. Turn this control clockwise to augment the proximity effect. Counterclockwise to diminish it. This can be used to fine tune the bass amount in an previously recorded track without having to use an external EQ, and by naturally intervening on the selected mics frequency response curve. Range is +/- 5 dBs.
- **HARMONICS:** this knob controls the amount of harmonic non linearity which can be added to Target microphones. It can be used to add subtle character to a flat or dull track, or when exaggerated to add a

perceivable “grit”. Range is 0 to 10 dBs.

- **IN:** this button engages the module’s processing.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific mic type settings for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix on your processing chain. This means that you will be able to independently process the Center or the Sides of your stereo program. The L and R icons will be replaced by M and S when in this mode. Click M to set the mic types for the Center (Mid) of your stereo image or S to set mic types for the Sides.

4.33 - Saturator X



The new T-RackS Saturator X delivers that classic analog saturation and takes you on a trip back in time to the birth of “modern” recording before computers and opens your recordings to the mysterious, elusive and warm world of analog saturation. The effect of Saturation, or the introduction of harmonic distortion, on an audio track or mix can add that certain special “flavor” and “warmth” that only magnetic tape, tubes, transformers or transistors can deliver.

Each of the ten saturation modes in the Saturator X reproduce the behavior of the electronic hardware component and circuitry (or recording medium, in the case of tape modes) that has been modeled for its peculiar character and different harmonic patterns it produces when driven hard. This makes for a very versatile and comprehensive saturation device as the palette of achievable tones varies from the very subtle “color” of the transformers (which offer a super light change in frequency response and only a hint of breakup towards the very top end of the gain range) to more pronounced and downright “nasty” distortion as in tube or solid states circuit designs.

Thanx to this versatile structure it can be freely used as a creative tool to add dimension, depth and a “warm and fuzzy” analog feel to either individual tracks or on the master fader of the DAW. On single tracks, distortion amounts can eventually be pushed to extreme limits for artistic purposes, whereas on a full mix gain should be applied more judiciously, in order to impart the character of an analog stage, be it tube, solid state or a tape deck, unless the distortion effect is desired on the master 2 bus too.

Controls:

- **GAIN:** this control sets the level of the signal going into the module. The higher the level, the more pronounced the saturation effect will be as the electronics of the engaged mode will reach their limit and start to distort the signal. The available gain range is from 0 to 40dBs.
- **OUTPUT:** this control determines the output level of the module. It works in the opposite direction as the

Gain control so that it can be used to compensate the output volume increase as more level (and thus saturation) is generated. The available range is from 0dB (no attenuation) to -40 dB.

- **LINK:** this control links the Gain and Output controls in an inverse way: when active as Gain is raised the Output is lowered by the same amount in order to compensate the volume increase and keep the output level consistent. Knobs can be set independently even when linked: they will move together maintaining the relative difference.
- **MODE:** this control lets you select the electronics type whose saturation behavior is modeled.

Available modes are:

- **TAPE 1:** this mode reproduces the classic character and high frequency compression of an analog tape machine reaching saturation when pushed above normal recording levels. It can also provide a good amount of added RMS and loudness level to finished mixes to make them sound fuller and “more complete” if slightly overdriven. More pronounced gain can provide a beautiful overload effect on many kinds of individual tracks including drums and vocals, and it can also keep the high frequency content under control.
- **TAPE 2:** similar to Tape 1 in terms of typical uses, this mode reproduces a more “Lo-Fi” analog tape response with more pronounced distortion and high frequency roll-off. It can be very effective in creating bold sonic enhancements to individual tracks or stems.
- **MASTER +6dB:** this is a digitally-recreated soft saturation curve that can be very useful when mastering finished tracks to add RMS level and “body” to a somewhat weak-sounding mix. If used gently, it adds a nice shine to every mix. If pushed harder, it becomes more suitable for individual tracks to make them sound bigger.
- **MASTER + 12dB:** this is similar to Master +6dB but with a more pronounced soft-saturation curve to make the effect even more prominent.
- **TUBE - PUSH PULL:** this mode produces classic tube saturation with odd harmonics added by the symmetrical clipping of the Push/Pull electronics. This creates a punchy distortion that can make everything sound warmer if used lightly or with an aggressive edge when pushed.
- **TUBE - CLASS A:** this mode produces tube saturation with more even harmonics added by the asymmetry of the Class A design. When used gently on stems or a finished mix, it adds a beautiful “halo” to the audio that is processed, definitely making it sound more analog. When pushed, the asymmetrical characteristic of Class A produce an uncontrolled distortion that sometimes sounds more interesting than Push/Pull.
- **SOLID STATE - PUSH PULL:** same as the Tube mode, but with the more aggressive character of the solid state circuitry. This mode is best used when transients need to be preserved.
- **SOLID STATE - CLASS A:** same as the Tube mode but with the more aggressive character of the solid state circuitry. This mode adds the same kind of magic “halo” the Tube Class A provides but with a more defined transient response.
- **TRANSFORMER - IRON:** this mode reproduces the very subtle color of iron core transformer harmonic enhancement which features less even harmonics. This mode adds a very gentle and subtle saturation adding an slight amount of distortion. For this reason, this mode is not useful for creative distortion effects and is best used for adding just a touch of color to tracks, stems or finished mixes. Since this mode is not clipping the audio, it can also be used to get a very nice pumping effect from the brickwall limiter when pushed really hard.
- **TRANSFORMER - STEEL:** this mode reproduces steel core transformer behavior. The difference with the iron is that it provides almost equal amount of odd and even harmonic enhancement. This mode adds a very gentle saturation introducing only a slight amount of distortion. For this reason, this mode is not

useful for creative distortion effects and is best used for adding just a touch of color to tracks, stems or finished mixes. Since this mode is not clipping the audio, it can also be used to get a very nice pumping effect from the brick wall limiter when pushed really hard.

- **IN:** engages or bypasses the processing.
- **BRICKWALL:** this control engages a brickwall limiter post output control. It can be enabled to prevent overshoots of the module's output when the input gain is turned higher to achieve distortion effects. Limiter's ceiling is set to -0.2 dBs. The brick wall limiter included with Saturator-X is used to keep the output level under control, but it can also be used to create a specific sonic character. In fact, this limiter can have a nice, intentional "pumping" effect that is very desirable on rhythmic material. To drive the Brickwall limiter hard, always use the Transformer modes since they don't clip the audio, leaving all the compression duties to the limiter.
- **OVERSAMPLING:** this control sets the oversampling mode of the module. The values are 1X (no oversampling), 2X and 4X. Higher values of oversampling will allow higher fidelity of the harmonic pattern reproduction and an overall higher quality sound with less (if any) aliasing artifacts. The tradeoff is that here the module is run at double or four times the session's sample rate, so its CPU consumption will be higher. Since it can be set independently for each module in a session, you can decide which tracks need less or no oversampling versus which ones need to be run higher (like the master fader) according to the available power of your CPU.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default.
- **L=R:** links the two channels for stereo operation: all parameters affect the two channels in the same way to maintain a consistent stereo image.
- **L/R:** click on the L or R icons if you want to dial in specific settings for L and R channels independently.
- **M/S:** this option uses a Mid-Side matrix to insert the stereo saturator in your processing chain. This means that you can independently process the Center or the Sides of your stereo program. The L and R icons are replaced by M and S in this mode. Click M to adjust settings for the Center (Mid) of your stereo image or S for the sides.

4.34 - ONE



The new ONE mastering processor was designed as an “all-in-one” solution for powerful and easy to use adjustments in creating final masters. It includes an EQ, compressor, harmonic exciter, low-end enhancer and limiter that will let anyone achieve professional quality results quickly and easily.

The module has 9 controls:

- **AIR:** gently controls the boost or cut of the High Frequencies.
- **FOCUS:** let you adjust the intelligibility of the mids.
- **BODY:** lets you fine tune the amount of low frequencies without upsetting the overall frequency balance.
- **BASS PUNCH:** lets you control the low-end boost and tightness of the sound at the same time.
- **ANALOG:** adds in a touch of harmonic enhancement to help give the track dimension and color.
- **TRANSIENTS:** can enhance or smooth out the attacks, to achieve the ideal balance of impact for your mix.
- **WIDTH:** lets you fine tune the stereo image of your track.
- **PUSH:** is a macro that controls all the settings of the compressor from gentle to hard.
- **VOLUME:** the one knob that makes your music sound louder without worrying about artifacts.

The ONE module will let anyone achieve professional quality results in the blink of an eye. Just turn its knobs, experiment and hear the mojo.

4.35 - Dyna-Mu



The DYNA-MU is an astonishing emulation of a well-known tube compressor celebrated for its warm, creamy sound and vivid colored tone, which thanks to the highest level of expertise from IK Multimedia, the pioneer in virtual circuit modeling, is perfectly recreated and indistinguishable from the hardware.

This modern-classic work of art has the characteristic of gluing together your mix with incredible quality, while it shines on individual tracks and is perfect as part of a professional mastering chain. Additionally, the saturation it produces when pushed to its limits can be used in many creative ways.

The DYNA MU, has 2 rows of controls that work for both L/R channels or Mid-Side, which are:

- **Input:** controls the amount of signal entering the processor.
- **Threshold:** sets the audio level for when the processor is engaged.
- **Attack:** the time in ms for the start processing after the threshold is exceeded.
- **Release:** how fast the processor reduces the compression ratio after the level falls below the threshold.
- **Output:** controls the level of signal coming out of the processor.

As you might notice, a common control not shown is the Ratio. The Ratio in this processor depends the amount of gain reduction required and on the mode you select. Normally the Dyna-Mu will have a somewhat softer ratio of around 1.5:1 but when the **HARD** switch is engaged a medium ratio of 4:1 is used. The **HARD** setting is perfect for single tracks while the normal setting is best suited for mixes.

The other 3 controls are GANG, LINK and Side Chain:

The GANG switch lets you control both rows at the same time while the LINK switch controls the sidechain detection mode for the two channels which can be either linked or independent.

The Side Chain switch inserts a gentle high pass filter so the processor is less sensitive to frequencies below 100Hz, ideal for heavy kick drums or deep basses in the mix.

The sound palette of the DYNA-MU is perfect to give cohesiveness and warmth to your mixes or tracks.

4.36 - EQual



The T-RackS EQual digital equalizer is a 10-band, ultra-clear, high-end parametric equalizer with an extremely transparent sound, ultra-precise editing and a vast array of filter shapes that are based on the typical curves of classic analog EQ's like Neve, SSL, API and other sought-after hardware.

This incredibly flexible “hybrid monster” gives you the best of both worlds - digital precision with transparent clarity and on-demand analog character that works perfectly for high end mastering or individual track work.

EQual opens with a flat white EQ curve and displays the spectrum of your audio source in the background and by clicking anywhere on the curve, the first band is created. From there you can select the shape and types of equalization based on your needs with up to 10 total bands.

The result is an ultra-high resolution processor that is the most transparent equalization tool digital audio technology allows today, offering the most wanted EQ curves and crystal clear processing for the most musical adjustment to any tracks or an entire mix.

For each band you will have different parameters, depending on the shape and type of filter applied, which are:

BELL:

- Standard
- UK Classic 1 (SSL 4000 modeling)
- UK Classic 2 (SSL 9000 modeling)
- UK Modern (Neve 1073/81)
- US Classic 1 (API 550A/B)
- US Classic 2 (API 560)

LOW SHELF:

- Standard
- UK Classic 1 (SSL 4000 modeling)
- UK Classic 2 (SSL 9000 modeling)

- UK Modern (Neve 1073/81)
- US Classic 1 (API 550A/B)

HI SHELF:

- Standard
- UK Classic 1 (SSL 4000 modeling)
- UK Classic 2 (SSL 9000 modeling)
- UK Modern (Neve 1073/81)
- US Classic 1 (API 550A/B)

HI PASS:

- Standard

LOW PASS:

- Standard

BAND PASS:

- Standard

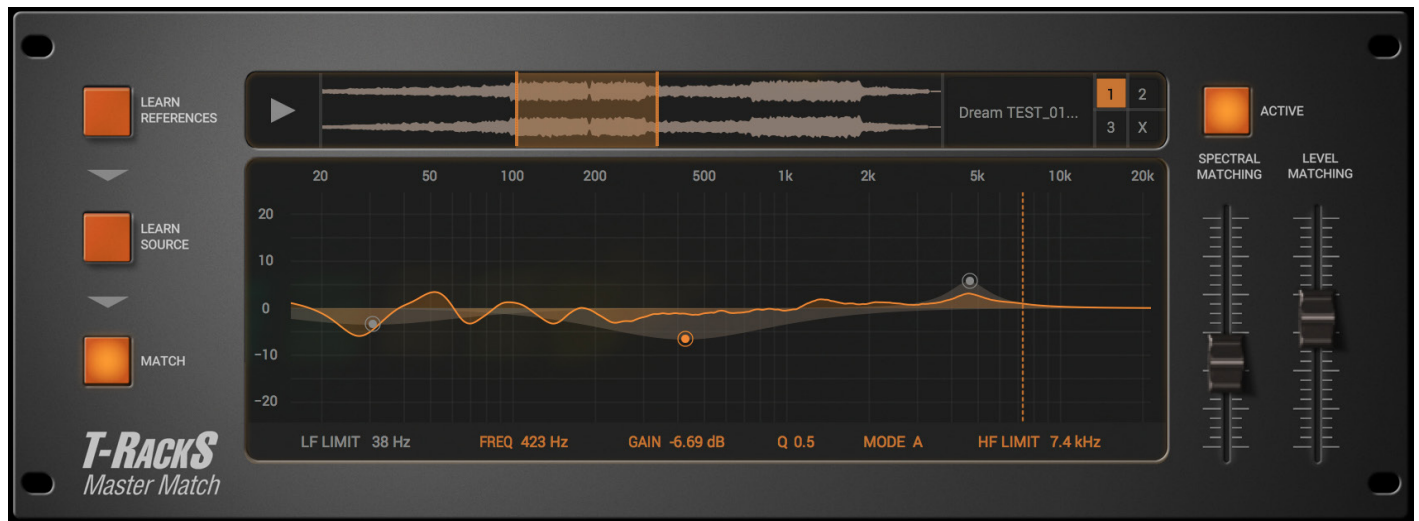
BAND STOP:

- Standard

TILT SHELF:

- Standard

4.37 - Master Match



The Master Match is a built in module of the new T-RackS 5, which, differently from the other modules, it has a fixed position in the chain: the last module.

This innovative module has the aim of easily give to your mix, the round and sonic characteristics of the references you had in mind already while recording and mixing, or at least the closest possible.

Also it gives you more sonic ideas, that you might never thought of.

The main point is find the perfect EQ that will give your track the sound of your reference, without interfere with your musical expression.

The technic is to analyze up to 3 references tracks equalization and RMS level, analyze your track and apply an EQ curve based on the results.

The Module is dividend in 4:

- Reference Upper Screen
- Curve Graphic Screen
- Matching Edit
- Matching Steps

To match your track with your ideas there are 3 simple steps to do:

Learn References, Learn Source and Match.

LEARN REFERENCES

To add a reference, drag and drop the track you choose in the upper screen or press the “PLUS” icon.

After insert at least 1 reference track, use the “PLAY” button to preview or the X to delete.

It is possible to analyze only a piece of the reference by selecting a range with the mouse; the selection turn

orange, and 2 handlers appears at the sides of the selection.

Press LEARN REFERENCE to analyze the track; the curve will appear on the Graphic and the LEARN SOURCE will blink to indicate the suggested next step.

LEARN SOURCE

The Learn Source start blinking suggesting to analyze your track.

Press LEARN SOURCE and the button will lights up, waiting you to play your track in your DAW or in the standalone.

Re-press the button to stop the analysis and your track curve will appear in the screen.

MATCHING

It is now time to match your track with your reference.

To do so, simply press MATCHING and the resulting curve applied will appear on the screen and the EQ MATCHING fader will automatically move to the suggested position for the best result.

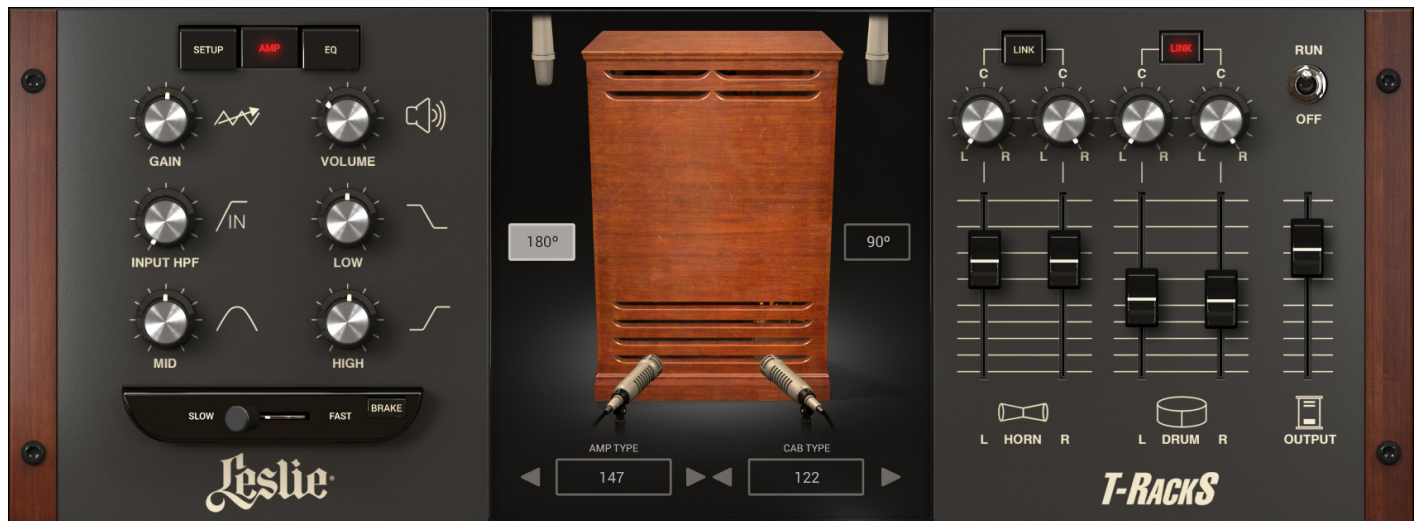
The best musical result is an EQ that gives you the best sounding song with the character of your references.

It is possible to create up to 8 Filters on the resulting curves now, simply clicking on the screen, each of them with individual FREQ, GAIN and Q, editable with the mouse or the values on the screen.

Also you can control other parameters to make your master sound perfect:

- **EQ MATCHING:** is the percentage of matching equalization resulted from the analysis. It goes from no matching at 0% to exaggerate matching at 100%.
- **LEVEL MATCHING:** is the percentage go matching RMS level resulted from the analysis. It goes from no matching at 0% to full match at 100%.
- **LF LIMIT:** set the limit until the matching is applied in the low frequency.
- **HF LIMIT:** set the limit until the matching is applied in the high frequency.
- **MODE A/B:** this mode gives you the possibility to choose between a more sharp or a more smoothed matching.

4.38 - Leslie®



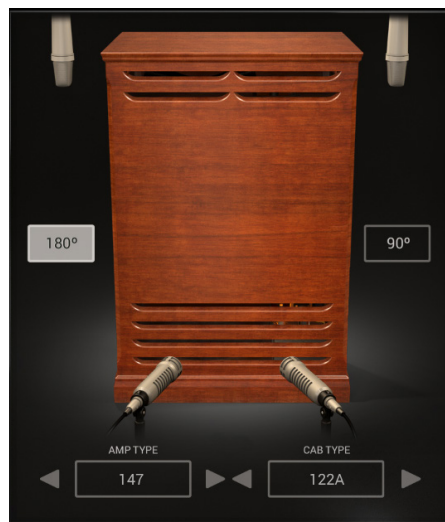
IK Multimedia and Hammond USA and Suzuki Music Corp. of Japan, teamed up to bring you the best sounding Leslie in the digital world, giving you **5 legendary Leslie® amps and 6 cabinets** in one powerful module to use inside the T-RackS 5 mixing and mastering workstation to further shape the sound of your tracks.

The amps were recreated using IK's innovative Dynamic Interaction Modeling Technology which can analyze and model the behaviour of every electronic component in the circuit as well as their mutual interactions to reproduce the exact behavior of the entire system in a non-linear and dynamic way.

Officially Certified Leslie® Models



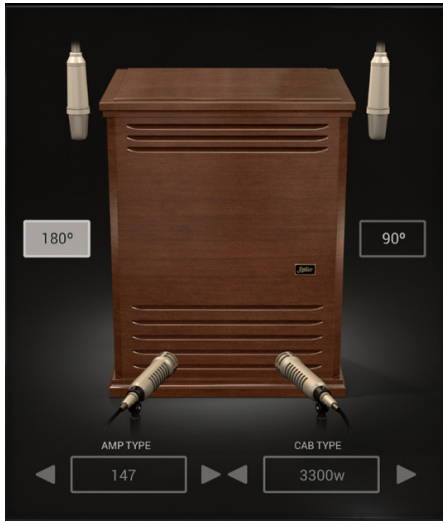
Leslie 122



Leslie 122A



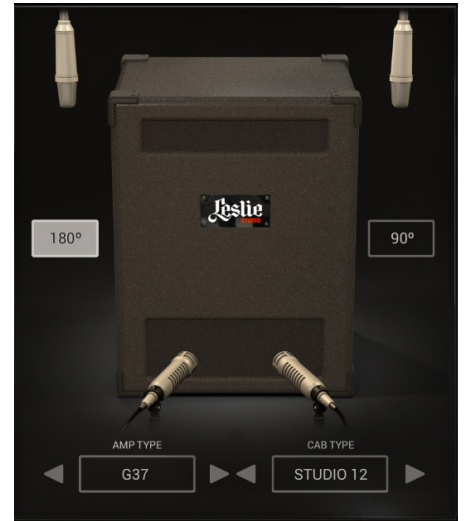
Leslie 147



Leslie 3300w



Leslie G37



Leslie Studio 12

The interface

The module is divided into 3 parts, one of which has 3 different “pages” to it.

The first part of the module is dedicated to the controls. This is where the majority of adjustments to amps and cabinets happens. It is divided into 3 pages: Setup, Amp and EQ. The Slow/Fast control is always visible at the bottom of the controls for easy access with the Brake as an external control.

Setup page



In the setup page, you have control over these essential parameters:

- **SPRING:** adds spring reverb to the signal before going in the amplifier. This simulates the on-board spring reverb tank found on many vintage Hammond® organs.

- **MIC DISTANCE:** you can decide how far from the cabinet you want the mic to be placed.
- **SLOW/FAST SPEED:** This sets the rotation speed of the horn and the drum.
- **ACCELERATION/DECELERATION:** sets the amount of time in the transition from fast rotation to slow and vice-versa.

Amp page



Leslie 122



Leslie 3300w



Leslie G37 / Studio 12

The “Amp” button will bring up the specific amplifier controls associated to each included model:

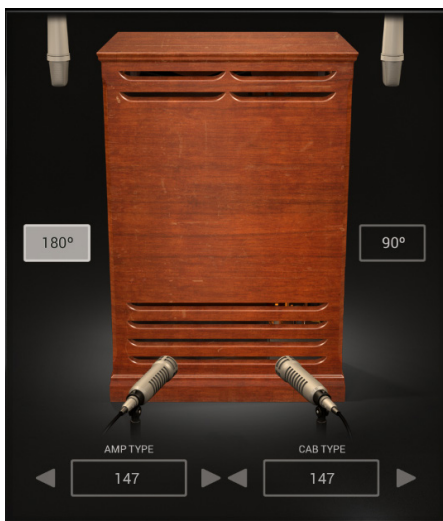
- **GAIN:** controls the amount of signal going to the amp and the level of saturation.
- **VOLUME:** controls the amount of signal going out of the amp into the cabinet.
- **INPUT HPF:** controls the frequency of the high pass filtering of the input signal from OFF to 200Hz.
- **TUBE DRIVE** (3300W amp only): Controls the amount of signal being “warmed up” by the tube section. When at minimum, the amp is in the clean channel. As soon as the knob is turned up, it automatically switches to the drive channel.
- **TUBE MODE** (3300W amp only): Controls the amount of tube saturation from Soft to Hard.
- **CONTOUR** (G37/Studio12 amp only): controls the amount of “warmth” created by the tube of the preamp.
- **EQ:** A guitar amp-style equalizer with controls for Low, Mid and High frequencies.

EQ page



The EQ button shows a full parametric EQ to let you further shape the sound with adjustable frequency and +/- 15dB of Gain control.

The center part of the module lets you select the amp and cabinet models and vary the mic positions which affects the perception of the rotation, or “doppler effect.” You can switch between a 90° position, which is the most common position that gives a non-symmetric perception of the rotation, to 180°, which is less common but still heard on a lot of famous recordings.



The third section of the module is for the mixer where you have full mix control of the four microphone channels used when capturing the Leslie speakers. Each channel has its own level and pan control so that you can set the Leslie mix as you prefer. You can, for example, set the drum to use just one channel panned center to have solid monaural low frequencies, and then use the horn in stereo. Or you can get a 70s sonic flavour by using just one mic on the top and one on the bottom panned just a bit apart, as another example.



4.39 - Tape Machine 24



A model of the MCI JH24. First produced in 1980, the JH24 was a staple in US studios during the 1980s. This completely transformerless, high-performance op-amp-based design delivers a pristine, phase-coherent and remarkably true-to-the-source audio performance. This results in a moderate and elegant polish to your music.

The interface controls are:

- **INPUT LEVEL:** this control sets the level of the signal going into the module.
- **INPUT/REPRO:** with INPUT the signal passes only through the input and output electronic stages of the recorders, bypassing the tape. Depending on how the recorder is designed the “Input” path can sound very very similar to the plug-in input, or slightly more colored. REPRO passes your signal through the entire analog recording system: input-> recording amplifier -> recording head -> tape -> playback head -> playback preamplifier -> output stage.
- **TRUE STEREO:** even a perfectly aligned tape machine will have a slight level, EQ and distortion differences between the left and right channels. This is good and part of the analog recording mojo! If a perfectly identical L and R processing is a must, simply disable this feature.
- **TAPE FORMULA:** selects between the 4 available tape formulas. Each tape formula has its own harmonic characteristic and the way it characterizes the sound will changes. This dynamic modeling is important in order to have an even broader palette of sounds:
 - **250:** modelled after the 3M/Scotch “250 Audio Recording Tape”. it delivers a warm tone with slightly more distortion and saturation than the other formulas included in the package.
 - **456:** modelled after the Ampex “456 High-Output Mastering Tape”. It offers a warm, round tone with a slight touch of saturation that greatly responds to the recording level.
 - **GP9:** modelled after the Quantegy “Quantegy GP9 Grand Master Platinum”. It’s very punchy and it’s perfect for high-quality analog recordings of nowadays.
 - **499:** modelled after the Ampex “499 Grand Master Gold Studio Mastering Audio Tape”. This tape is designed to handle a great amount of level with minimal distortion and compression. It also exhibits

a high frequency added definition that makes it perfect for printing “digital-like” recordings, while still maintaining an analog sound.

- **TAPE SPEED:** selects the speed in inches per second (IPS) of the tape transport between 15 ips and 30 ips. The higher speed gives you more fidelity, and the lower speed delivers a warmer rounder tone.
- **TRANSPORT MODELING:** precisely models the behavior of the mechanical transport. Small irregularities in the movement of the tape create various degrees of sonic alteration to the audio program, especially between the two channels. Keep it on for the magic to happen! In case a perfectly steady performance is needed, just disable the control.
- **AUTO CALIBRATION:** the Tape Machines are calibrated by default at their factory calibration and it possible edit the calibration parameters at your will. With the auto-calibration, it is possible to reset the edits and go back to the factory calibration.
- **RECORD BIAS:** this control adjusts the amount of bias in the recorded signal. Ideal bias voltage settings provide maximum sensitivity and low distortion. Intentionally overbiasing is a common technique to produce a warmer, gently saturated sound. Underbiasing can be used to nicely boost the high frequencies in a truly unique way and add distortion and other nonlinear effects.
- **RECORD LEVEL:** when in REPRO it adjusts the level of the recorded signal after the Input.
- **RECORD HF EQ:** adjusts the high-frequency shelving added to the recorded signal on the tape.
- **PLAY LEVEL:** when in REPRO it adjusts the level of the recorded signal before the output after the entire tape. It is used to calibrate the machine.
- **PLAY HF EQ:** adjusts the high-frequency shelving added to the playback signal from the tape.
- **PLAY LF EQ:** adjusts the low-frequency shelving added to the playback signal from the tape.
- **OUTPUT LEVEL:** this control determines the output level of the module.
- **ON:** activates or bypasses the processing.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default.

4.40 - Tape Machine 80



The legendary Studer A80 Mk II, in its various revisions, is an essential part of countless influential records. Engineered and manufactured in Switzerland from 1970 to 1988, it became the de facto standard for professional high-end multi-tracking. The A80's sonic signature is a perfect blend between transparency and subtle harmonic enhancement.

The interface controls are:

- **INPUT LEVEL:** this control sets the level of the signal going into the module.
- **INPUT/REPRO:** with INPUT the signal passes only through the input and output electronic stages of the recorders, bypassing the tape. Depending on how the recorder is designed the "Input" path can sound very very similar to the plug-in input, or slightly more colored. REPRO passes your signal through the entire analog recording system: input-> recording amplifier -> recording head -> tape -> playback head -> playback preamplifier -> output stage.
- **TRUE STEREO:** even a perfectly aligned tape machine will have a slight level, EQ and distortion differences between the left and right channels. This is good and part of the analog recording mojo! If a perfectly identical L and R processing is a must, simply disable this feature.
- **TAPE FORMULA:** selects between the 4 available tape formulas. Each tape formula has its own harmonic characteristic and the way it characterizes the sound will changes. This dynamic modeling is important in order to have an even broader palette of sounds:
 - **250:** modelled after the 3M/Scotch "250 Audio Recording Tape". it delivers a warm tone with slightly more distortion and saturation than the other formulas included in the package.
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 - **499:** modelled after the Ampex "499 Grand Master Gold Studio Mastering Audio Tape". This tape is

designed to handle a great amount of level with minimal distortion and compression. It also exhibits a high frequency added definition that makes it perfect for printing “digital-like” recordings, while still maintaining an analog sound.

- **TAPE SPEED:** selects the speed in inches per second (IPS) of the tape transport between 15 ips and 30 ips. The higher speed gives you more fidelity, and the lower speed delivers a warmer rounder tone.
- **TRANSPORT MODELING:** precisely models the behavior of the mechanical transport. Small irregularities in the movement of the tape create various degrees of sonic alteration to the audio program, especially between the two channels. Keep it on for the magic to happen! In case a perfectly steady performance is needed, just disable the control.
- **AUTO CALIBRATION:** the Tape Machines are calibrated by default at their factory calibration and it possible edit the calibration parameters at your will. With the auto-calibration, it is possible to reset the edits and go back to the factory calibration.
- **RECORD BIAS:** this control adjusts the amount of bias in the recorded signal. Ideal bias voltage settings provide maximum sensitivity and low distortion. Intentionally overbiasing is a common technique to produce a warmer, gently saturated sound. Underbiasing can be used to nicely boost the high frequencies in a truly unique way and add distortion and other nonlinear effects.
- **RECORD LEVEL:** when in REPRO it adjusts the level of the recorded signal after the Input.
- **RECORD HF EQ:** adjusts the high-frequency shelving added to the recorded signal on the tape.
- **PLAY LEVEL:** when in REPRO it adjusts the level of the recorded signal before the output after the entire tape. It is used to calibrate the machine.
- **PLAY HF EQ:** adjusts the high-frequency shelving added to the playback signal from the tape.
- **PLAY LF EQ:** adjusts the low-frequency shelving added to the playback signal from the tape.
- **OUTPUT LEVEL:** this control determines the output level of the module.
- **ON:** activates or bypasses the processing.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default.

4.41 - Tape Machine 440



The Ampex 440B series dates from the late 1960s and delivers a typical, unmistakably soulful color that made this professional and historic mixing and mastering machine a legend. This superb blend of art and technology has the power to turn your mixes into finished tracks.

The interface controls are:

- **INPUT LEVEL:** this control sets the level of the signal going into the module.
- **INPUT/REPRO:** with INPUT the signal passes only through the input and output electronic stages of the recorders, bypassing the tape. Depending on how the recorder is designed the “Input” path can sound very very similar to the plug-in input, or slightly more colored. REPRO passes your signal through the entire analog recording system: input-> recording amplifier -> recording head -> tape -> playback head -> playback preamplifier -> output stage.
- **TRUE STEREO:** even a perfectly aligned tape machine will have a slight level, EQ and distortion differences between the left and right channels. This is good and part of the analog recording mojo! If a perfectly identical L and R processing is a must, simply disable this feature.
- **TAPE FORMULA:** selects between the 4 available tape formulas. Each tape formula has its own harmonic characteristic and the way it characterizes the sound will changes. This dynamic modeling is important in order to have an even broader palette of sounds:
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 - **499:** modelled after the Ampex “499 Grand Master Gold Studio Mastering Audio Tape”. This tape is designed to handle a great amount of level with minimal distortion and compression. It also exhibits

a high frequency added definition that makes it perfect for printing “digital-like” recordings, while still maintaining an analog sound.

- **TAPE SPEED:** selects the speed in inches per second (IPS) of the tape transport between 7.5 ips and 15 ips. The higher speed gives you more fidelity, and the lower speed delivers a warmer rounder tone.
- **TRANSPORT MODELING:** precisely models the behavior of the mechanical transport. Small irregularities in the movement of the tape create various degrees of sonic alteration to the audio program, especially between the two channels. Keep it on for the magic to happen! In case a perfectly steady performance is needed, just disable the control.
- **AUTO CALIBRATION:** the Tape Machines are calibrated by default at their factory calibration and it possible edit the calibration parameters at your will. With the auto-calibration, it is possible to reset the edits and go back to the factory calibration.
- **RECORD BIAS:** this control adjusts the amount of bias in the recorded signal. Ideal bias voltage settings provide maximum sensitivity and low distortion. Intentionally overbiasing is a common technique to produce a warmer, gently saturated sound. Underbiasing can be used to nicely boost the high frequencies in a truly unique way and add distortion and other nonlinear effects.
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- **PLAY LEVEL:** when in REPRO it adjusts the level of the recorded signal before the output after the entire tape. It is used to calibrate the machine.
- **PLAY HF EQ:** adjusts the high-frequency shelving added to the playback signal from the tape.
- **PLAY LF EQ:** adjusts the low-frequency shelving added to the playback signal from the tape.
- **OUTPUT LEVEL:** this control determines the output level of the module.
- **ON:** activates or bypasses the processing.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default.

4.42 - Tape Machine 99



The Revox PR99 Mk II is a stereo professional recorder produced by Studer in the 1980s. Engineered around a hybrid design that combines the best of discrete and op-amp topologies, this model offers a very smooth frequency response, especially on bass, with almost no trace of the typical “head bump” phenomenon. A standard for broadcast and classical music, this machine was appreciated by audiophiles for its exceptional sonic quality.

The interface controls are:

- **INPUT LEVEL:** this control sets the level of the signal going into the module.
- **INPUT/REPRO:** with INPUT the signal passes only through the input and output electronic stages of the recorders, bypassing the tape. Depending on how the recorder is designed the “Input” path can sound very very similar to the plug-in input, or slightly more colored. REPRO passes your signal through the entire analog recording system: input-> recording amplifier -> recording head -> tape -> playback head -> playback preamplifier -> output stage.
- **TRUE STEREO:** even a perfectly aligned tape machine will have a slight level, EQ and distortion differences between the left and right channels. This is good and part of the analog recording mojo! If a perfectly identical L and R processing is a must, simply disable this feature.
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 - **GP9:** modelled after the Quantegy “Quantegy GP9 Grand Master Platinum”. It’s very punchy and it’s perfect for high-quality analog recordings of nowadays.

- **499:** modelled after the Ampex “499 Grand Master Gold Studio Mastering Audio Tape”. This tape is designed to handle a great amount of level with minimal distortion and compression. It also exhibits a high frequency added definition that makes it perfect for printing “digital-like” recordings, while still maintaining an analog sound.
- **TAPE SPEED:** selects the speed in inches per second (IPS) of the tape transport between 7.5 ips and 15 ips. The higher speed gives you more fidelity, and the lower speed delivers a warmer rounder tone.
- **TRANSPORT MODELING:** precisely models the behavior of the mechanical transport. Small irregularities in the movement of the tape create various degrees of sonic alteration to the audio program, especially between the two channels. Keep it on for the magic to happen! In case a perfectly steady performance is needed, just disable the control.
- **AUTO CALIBRATION:** the Tape Machines are calibrated by default at their factory calibration and it possible edit the calibration parameters at your will. With the auto-calibration, it is possible to reset the edits and go back to the factory calibration.
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- **RECORD HF EQ:** adjusts the high-frequency shelving added to the recorded signal on the tape.
- **PLAY LEVEL:** when in REPRO it adjusts the level of the recorded signal before the output after the entire tape. It is used to calibrate the machine.
- **PLAY HF EQ:** adjusts the high-frequency shelving added to the playback signal from the tape.
- **OUTPUT LEVEL:** this control determines the output level of the module.
- **ON:** activates or bypasses the processing.
- **RESET:** when clicked, this control brings the settings of all of the parameters back to their default.

4.43 - Sunset Sound Studio Reverb



Sunset Sound Studio Reverb is an all-new hybrid convolution/modelling engine made to deliver the highest quality reverbs. With the right amount of control, we captured each space in the studio using Sunset's own microphone collection and each studio's own console.

IK's award-winning physical modeling technology recreates each studio's preamps and console signal path to achieve a perfectly accurate amount of warmth and tone

With Sunset Sound Studio Reverb it is possible to use all the rooms, chambers, and mechanical reverbs of the physical studio in one single plug-in, giving to your track that magical vibe and ambiance it deserves.

The interface controls are:

- **POWER:** This control enables or disables the processor.
- **RESET:** This control resets the processor to its default state.
- **INPUT:** This control adjusts the amount of signal going into the processor. The signal can be lowered until reaching complete silence ($-\infty$) or boosted up to 6dB from its nominal level. The input state is saved in each preset.
- **MONO:** Since Sunset Sound Studio Reverb is a true stereo plugin, this controls sums the L and R channels of the input to create a MONO input for the wet signal.
- **PRE-DELAY:** This control determines the amount of time between the initial input signal and the reverberation. Values from the pre-delay range from 0ms to 1sec. All values are shown as ms (milliseconds).
- **DECAY:** This control adjusts the length of the reverb, to shorten or elongate the time that the reverb tail lasts. Because of the dynamic nature of the control, the displayed value is from 0% to 100%. To maintain the integrity of each space, the length of 100% will depend on the selected room, while the 0% will be a decay of 1 ms.
- **HPF (Hi pass filter):** This control sets the frequency of the high pass filter before the reverberation path. It is a 6dB/octave filter with a range from 10Hz to 20 kHz, with an OFF option set under 10Hz. The default setting is OFF. All values are shown in Hz and kHz.

- **LPF (Low pass filter):** This control set the frequency of the low pass filter before the reverberation path. It is a 6dB/octave filter with a range from 40Hz to 20 kHz, with an OFF option set above 20kHz. The default setting is OFF. All values are shown in Hz and kHz.
- **LOW (Low Shelf):** This control changes the gain of the low shelf filter after the reverberation path. The frequency is set at 200Hz with a Q of 0.4 and its range is from -12dB to 12dB. The default setting is 0dB. All values are showed in dB. This control can be used by clicking & dragging on the vertical axis or by typing in a numerical value in the space where the value is displayed.
- **HIGH (High shelf):** This control changes the gain of the high shelf filter after the reverberation path. The frequency is set at 5kHz with a Q of 0.4 and its range is from -12dB to 12dB. The default setting is 0dB.
- **MIC:** Selector control with 3 fixed positions. Set to “L” only the left microphone will be used, in “R” only the Right microphone will be used, and in STEREO both mics will be used.
- **WIDTH:** When STEREO is selected, this control lets you adjust width of the stereo image of the reverb.
- **DRY:** Controls the amount of original sound.
- **WET:** Controls the amount of processed sound.
- **SOLO:** Solos just the WET or DRY signal to let you hear only the processed signal or the original signal without needing to adjust the WET/DRY mix.
- **REVERB TYPE:** These controls select the reverb to be used.
The first row selects the room type:
 - Chamber
 - Live room
 - ISO booth
 - Plate/SpringThe second row selects which of the included reverbs to be used:
 - Chamber: Studio 1, Studio 2 or Studio 3
 - Live room: Studio 1, Studio 2 or Studio 3
 - Iso Booth: Studio 1, Studio 2 or Studio 3
 - PLATE: Plate 1, Plate 2 and Spring.

The screen default and main view are a 360-degree image which used to indicate the selected room. Each time a reverb is selected, the matching image is loaded. To navigate the screen simply click and drag the image. When a Plate or Spring is selected, the display will show a fixed photo of the Plate or Spring.

There are two controls that are placed at the bottom of the screen:

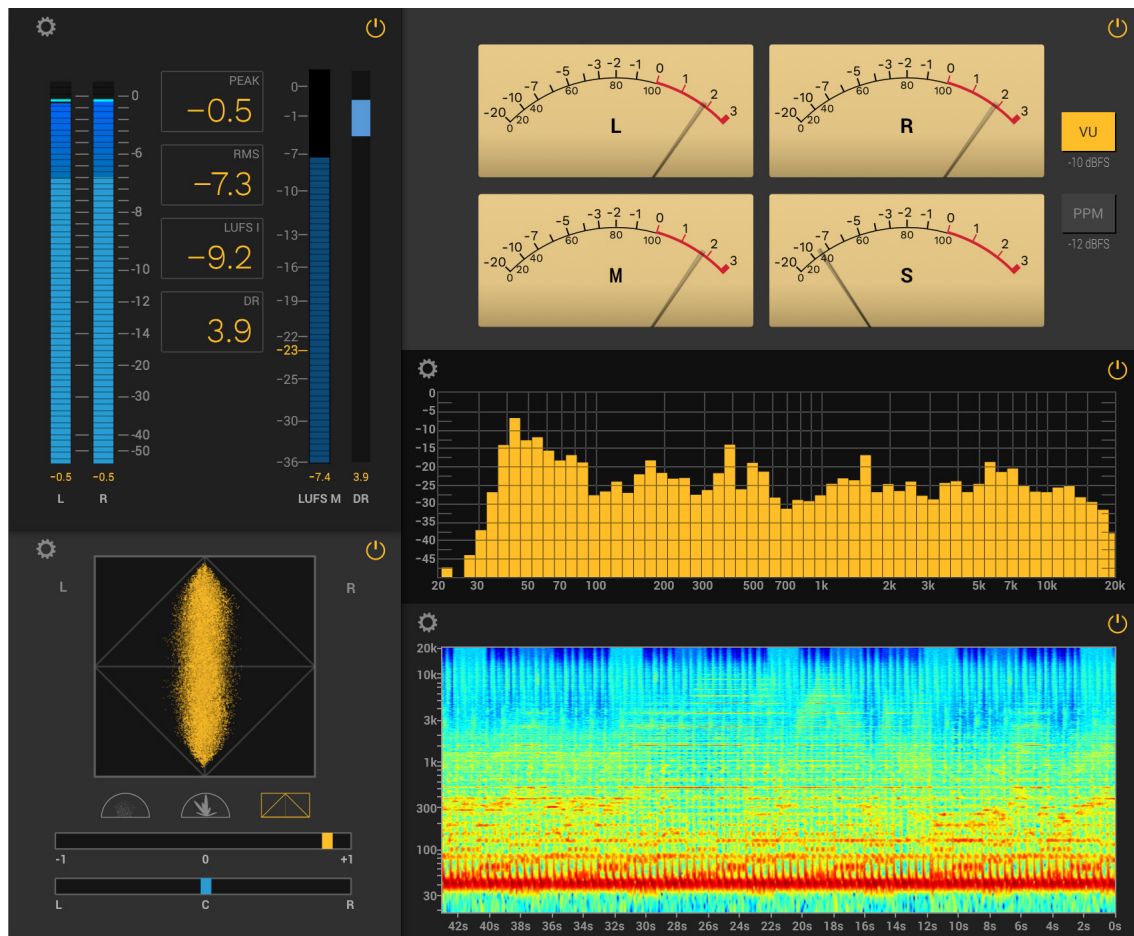
- **INFO:** Covers the image and displays the description and general info about the selected room or plate. If you want to know more about the Sunset Sound Recorders, visit the website <http://www.sunsetsound.com>
- **OPTION:** Some secondary controls for each room are found in this menu, adding interesting possibilities to shape the sound of the selected reverb.
The available options are:
 - **Chamber of Studio 1, Studio 2 and Studio 3:**

- Phase selection between 0° and 180°.
- **Live Room Studio 1:**
 - Dampening: None, Half and Full. This control alters the response of the room.
 - Phase selection between 0° and 180°.
- **Live Room Studio 2:**
 - Position: Position 1, Position 2 and Position 3. This controls lets users select which of the 3 main positions the room is recorded in.
 - Phase selection between 0° and 180°.
- **Live Room Studio 3:**
 - Dampening: None, Half and Full. This controls alters the response of the room.
 - Phase selection between 0° and 180°.
- **Iso Booth Studio 1, Studio 2 and Studio 3:**
 - Phase selection between 0° and 180°.
- **Plate 1:**
 - Decay: Low, Mid, High. Select between the shortest, intermediate and longest decays available on this vintage machine.
 - Phase selection between 0° and 180°.
- **Plate 2:**
 - Decay: select different length of decay between 300ms, 600ms, 1000ms, 1500ms, 1800ms, 2300ms, 4000ms, 5000ms and 6000ms.
 - Phase selection between 0° and 180°.
- **Spring:**
 - Decay: Low, Mid, High. Select between the shortest, intermediate and longest decays available on this vintage machine.
 - Phase selection between 0° and 180°.

Chapter 5 - Metering

The metering section is the most complete solution available and is fully equipped, with:

- A new precision loudness LUFS meter which is compliant with all leading professional standards and provides valuable information about the integrated, mid or short term loudness, to set the perfect levels for digital music delivery or professional broadcasting.
- The True Peak, RMS and Dynamic Range meters, give you useful information about the “crest factor” (the difference between RMS and Peak levels) and overall dynamic response of your mixes.
- A new spectrogram, a real-time analyzer and phase and correlation meters, will help you visually spot frequency imbalances and phase problems you might need to address before printing your masters.
- The VU and PPM meters provide another level reference in a classic visual format with different ballistics. Volume Units show an average level of loudness tied to the “crest factor” while Peak Program metering provides a faster response for monitoring transients.
- Mid-Side meters help you gather information about the relationship between the mono and the stereo content of a track in terms of its “overall energy”.



5.1 - Peak Meter

This is a precision sample-accurate peak meter that shows the instantaneous audio peak level, with peak hold and over indication. Numerical values for peak hold values are shown at the right of the barograph.

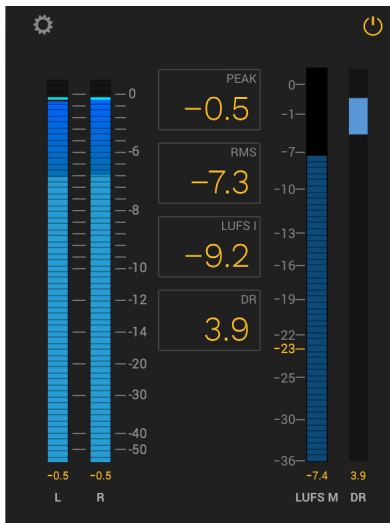
The peak meter can work on four scales, selectable by clicking on the Settings button and selecting the Peak Meters section.

- **-90 .. 0 dB:** this scale is very wide and is good for general purpose peak level representation.
- **-50 .. +5 dB:** this scale has reduced range on low levels and can be useful to monitor “higher than 0dB” signals, as it goes up to +5 dB.
- **-60 .. 0 dB:** this scale has a “zoom” at the high portion of the scale so that the upper 8 dB spans over the last half of the scale. This is very useful to keep the top of the available headroom under control.
- **-50 .. 0 dB:** standard scale for various purposes.

OVER indicators: these two RED light will turn on only when more than 3 consecutive 0 dB full-scale samples are present on the audio.

Use True Peak: A true peak meter displays the absolute peak of an audio’s waveform as it will be heard in the analogue realm through speakers. Using it, you can avoid inter-sample peak distortion.

Peak and RMS have also a numerical meter, to let you have a precise but smooth workflow.



5.2 - LUFS: Loudness meter

Loudness is definitely an important part of our engineer job, and we need to carefully measure and apply changes to our work.

This is why the new LUFS meter is compliant with ITU-R BS.1770-4, covering the majority of loudness standards worldwide.

This standard determines the reference level you must follow to be compliant for professional applications, and

selecting them on the settings button, the reference level tag will be highlighted.

The professional broadcast reference are:

- EBU R 128 (Europe) : -23 dBFS
- ATSC A/85 /(US): -24 dBFS
- OP-59 (Australia): - 24 dBFS
- TR-B32 (Japan): -24 dBFS

We wanted to add what we thought will be the new standard for music streaming, already in use but not official:

- Apple Music: - 16 dBFS
- Other (Spotify, YouTube, etc.): - 14dBFS

We also added Target loudness ranges, to help you fit in your music creation, where is maybe not standard related.

You can decide to visualise the LUFS meter as:

- Momentary
- Short term
- Integrated
- MAX Momentary
- Max Short Term

The numeric meter is not linked to the visual, so you can have the Short Term in the visual and Momentary in the numeric.

We also provide a Dynamic range meter, to visualize and control the ratio between the largest and smallest values that your signal can assume.

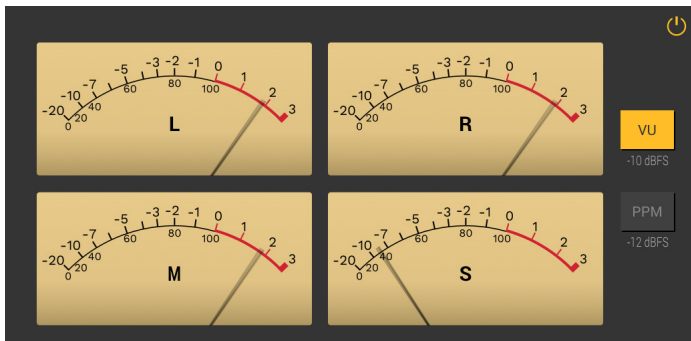
By pressing the Settings Icon, it is possible to select all those preferences and references in order to deeply customize your meter to fit your necessities.

5.3 - VU Meter and PPM

The VU-meter, or Volume Unit meter, displays the signal level of your track and intentionally “slows” measurement, averaging out peaks and troughs of short duration. It also reflects more of the perceived loudness of the audio material.

The PPM, or Peak Programme meter instead shows the peak level of the waveform, not matter how brief its duration.

It is possible to set the reference level in dB to fit your necessities.



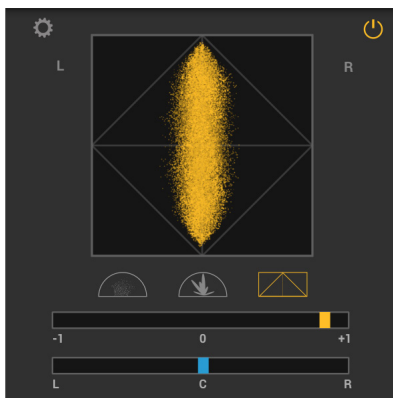
5.4 - Phase Scope

The phase scope provides essential information about the phase relationships between your left and right channels and reacts in real time to the incoming signal. The amount of out-of-phase material in your audio will be visualized under the X axis.

While the differences in phase between the L/R channels play a fundamental role in creating the sensation of wide stereophonic content, too much difference will also cause important mono incompatibility and an inconsistent listening experience.

When the stereo master is pushed very hard you will notice a sort “transformation” of the phase scope material that will start resembling a square, with sharper edges. Analyzing your preferred albums and referencing them against this characteristic will help you understand how hard you can push it before going overboard.

You can choose to view the entire goniometric meter, or as a half circle view in dots or lines.



By clicking on the Settings icon: you will be able to adjust the resolution, which means how detailed you want the image to be, and the persistence, which gives you information about the phase relations over a longer period of time.

By enabling the Normalize function, you will make the scope display signals with the same width at any level, even very low ones (default).

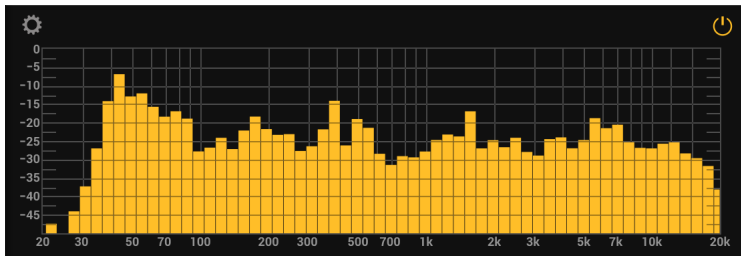
The phase correlator is a very simple meter that shows phase correlation between the two Left and Right channels on a scale from -1 to 1. A mono signal will show +1, a completely out of phase signal will show -1.

Normally, professionally mastered music will make the correlator move between +1 and 0. It is normal to have signals that occasionally fall in the negative area, but keep in mind that passages that stay constantly between 0 and -1 will be perceived significantly lower when played in mono, if not completely unheard.

5.5 - RTA

This is FFT RTA that can be configured to give various kind of information.

- **Settings:** click on Settings icon to have access to the RTA settings.
- **Filter Size:** the filter size can be 1, 1/3, 1/6 or 1/9 of octave.
- **Channel mode:** select between L, R and L+R.
- **Range:** the ranges of vertical scale of the RTA can be set in -130...+5, -90...0 or -60...0.
- **Release:** set the release time of the meter between 250ms, 500ms and 1sec.
- **Peak Hold time:** it is possible to visualise the peak in the spectrum and here you can select for how long.

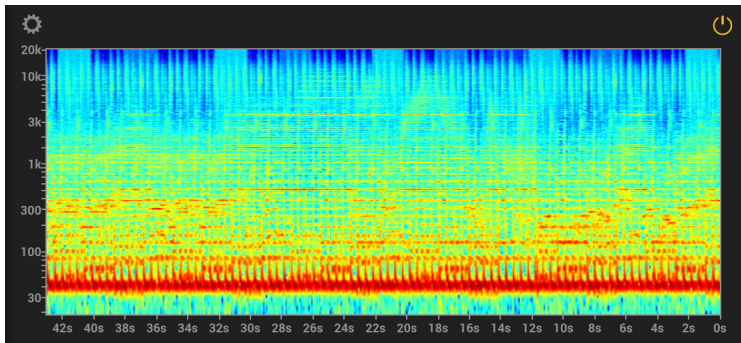


5.6 - Spectrogram

Spectrogram is a visual representation of the spectrum of frequencies of sound or other signal as they vary with time or some other variable.

In the horizontal axis represent the time, while the vertical axis represent the frequency.

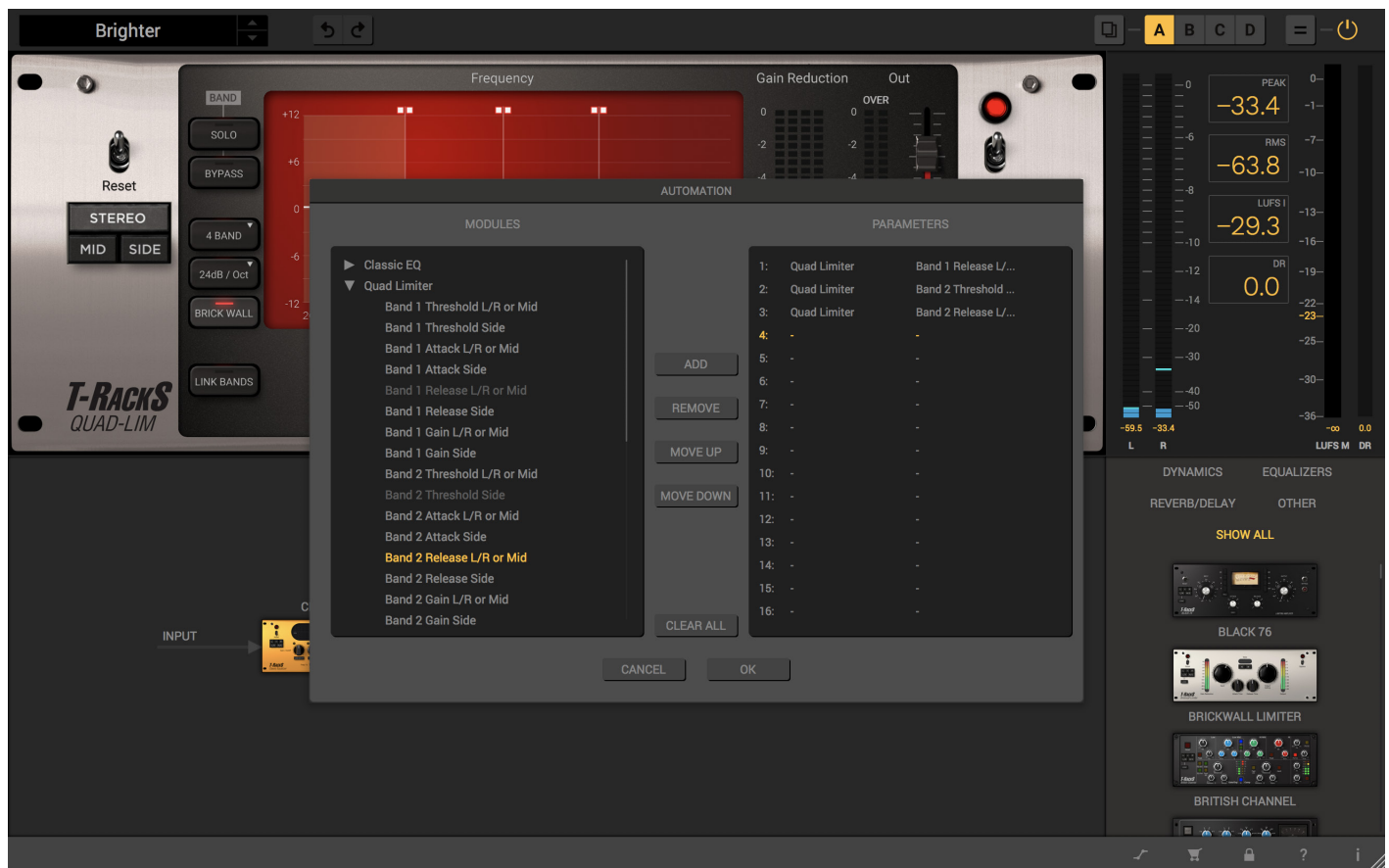
- **Settings:** click on Settings icon to have access to the Spectrogram settings.
- **FTT Size:** the filter size can be 2048, 4096, 8192, 16384 and 32764 samples.
- **FTT Overlap:** the overlap speed can be at 2x, 4x, 8x or 16x.
- **FTT Window:** the Window can be Hann, Hamming, Blackman-Harris, Flat Top style or no window.
- **Channel mode:** Select between L, R and L+R.
- **Reset Display:** click to clear the spectrogram.
- **Hold Display:** click to hold the spectrogram in a precise moment.



Chapter 6 - Automation

6.1 - Automating parameters

T-RackS Suite has lots of parameters, more than what typically audio sequencers (DAWs) can support. For this reason, T-RackS Suite has its own internal Automation Parameter assignment window that is launched by clicking on the AUTOMATION button.



The Automation window includes two panes. The Left Panel is called MODULES and the Right Pane is called PARAMETERS.

In MODULES, processors will appear dynamically, hence you will see only the loaded modules in the chain, showing the relative parameters.

The PARAMETER lists the 16 automation parameters that are declared to the DAW. These parameters names (Param 1 to Param 16) are fixed because most DAWs do not support automation parameter name changes while the plug-in is open.

To add a parameter to be automated, simply select it in the MODULES and press ADD, and it will be loaded in the selected PARAMETER slot.

You can remove single parameter assignment with the REMOVE button, or remove them all with the CLEAR ALL

button.

You can also swap DAW automation parameters with the MOVE UP, MOVE DOWN buttons.

Clicking OK will save and close the Automation window. The automation assignment map will be stored and restored to and from the session you're working on.

When working with multiple instances of T-RackS plug-in, each T-RackS instance is independent from the others and can have different Automation assignments setup. This means that saving your DAW session, will save each the automation setup for each T-RackS instance.

Chapter 7 - Mastering tips

7.1 - Introduction

Mastering is the final stage of processing made on a group of stereo mixes before going to press on audio CDs, movie soundtracks, multimedia soundtracks, MP3 files and so on. Its purpose is to obtain the maximum possible sound quality from the original mixes by “sweetening” them with equalization, compression and limiting with the aim of letting the music shine as much as possible in all its nuances and enhancing the original musical production goals.

Furthermore, mastering is essential to create a balanced, pleasing continuity from one track to the next of a CD, for example.

It is extremely important to work with the best possible speakers in a room that sounds good, because this is the final stage where you’ll have to make decisions on the overall balance between the songs, and it’s an extremely delicate process (see paragraph 7.7).

If you have to process the audio file with other effects like reverbs, stereo image corrections, phase coherency adjustments, spectral enhancers, etc., be sure to do this before mastering with T-RackS. **T-RackS should always be the last processing pass in the audio production workflow.**

7.2 - T-RackS 5 chain setup tips

There are no fundamental rules for the order of the processors in the T-RackS 5 chain. You can start analyzing the T-RackS factory presets to discover how typical mastering chains are done, and you can start experimenting with your own setups later on.

Something that should always be considered as a rule is **placing a “finalization” processor like the T-RackS CS Brickwall Limiter or the T-RackS Classic Multiband Limiter at the very end of the chain.** This will allow you to keep the final level of your master under control.

Typically, equalizers are placed before dynamics, but this is not an absolute rule.

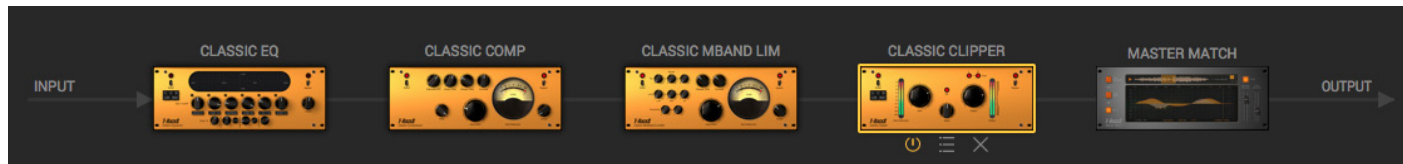
7.3 - Loudness

Nowadays everybody wants to make louder and louder masters! T-RackS CS allows for extremely loud, clean and detailed masters. But be aware that music needs to have its own dynamics respected as much as possible and that, even if good loud masters are possible, there is a limit that should never be exceeded.

T-RackS gives a good hint on these limits with Perceived Loudness and RMS meters, but it is fundamental to always remember that only your ears and experience will tell if the level is fine or not.

7.4 - Simple mastering chain

This is great starting chain to start with T-RackS CS:



It includes the Classic T-RackS EQ, Classic T-RackS Compressor, Classic T-RackS Limiter and at the end of the chain Classic T-RackS Clipper.

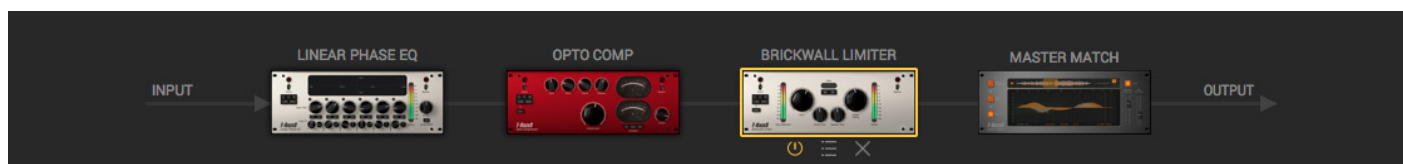
For Rock, Pop, etc., start with 2 dB of compression on the Compressor, another 1-2 on the Limiter and maybe gain a few dBs on the clipper as well.

For a more straight and close to the original sound you can remove the Multiband Limiter from this chain and just use the EQ, Compressor and Clipper.

7.5 - Advanced mastering chain

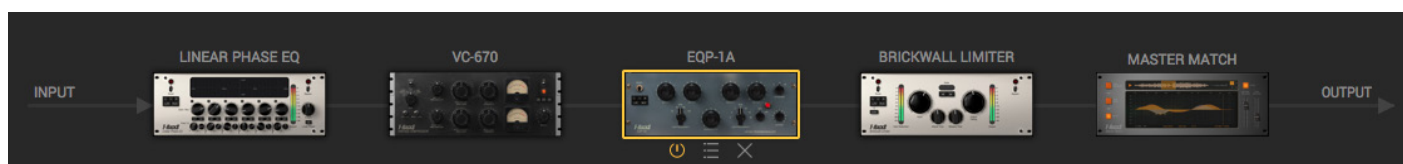
The high end processors included in T-RackS CS like Vintage Compressor Model 670, Linear Phase Equalizer, Brickwall Limiter, Opto Compressor, and the EQP-1A Tube Vintage Program EQ, allow the highest versatility during mastering. Think to these modules as tools in your gear arsenal to be chosen from depending on the music you're working on, on your taste, etc. Feel free to experiment with all combinations of processors on all kind of audio material, and you'll find very good results!

This is a very good starting chain for stereo Mastering with T-RackS CS processors if you want an extremely transparent and pure processing path:



It includes the Linear Phase Equalizer, Opto Compressor and Brickwall peak limiter. It's a very transparent chain that will add very little coloration and will respect the original nuances and timbral characteristic of the music as much as possible, even while giving you all the tools to master the music properly.

Experimenting with more vintage characterized sounds, add thickness and warmth, like with this chain .



This includes the Linear Phase Equalizer, Vintage Compressor model 670, Vintage Program EQP-1A and the Brickwall Limiter at the end, that could eventually be replaced by the Classic T-RackS Clipper.

Two Compressors: you can **use two compressors in the chain if you want to give more consistency to levels** without having to push too much one single compressor. To do that we suggest you use a Vintage Compressor model 670 with a very slow Time Constant, like 3 or 4 applying 2 dB of compression Max, followed by an Opto Compressor set to respond very quickly, with short attack and release times, applying 1 or 2 dB of compression max.

Parallel Compression: you may like the character and musical feel that a compressor can give when it's pushed heavily, but it can be too much for mastering, resulting in an overcompressed, somehow lifeless track. A good trick in this case is to insert in parallel with the compressor a non compressing device or another compressor set to attenuate way less, if not at all. Insert in parallel slots two compressors, one set to compress 5 or 6 dB, the other set to compress 0.5 dB max.

We suggest to always use linear phase equalization where you have to boost the low end in a mix that is already very dense, or when you need to tweak very sensitive midrange zones where lots of instruments are playing overlapping with vocals.

7.6 - T-RackS CS models tips

7.6.1 - Vintage Compressor Model 670



This compressor is magic. You will immediately notice that even with just 1 or 2 dB of compression, or even less, the music truly shines. Full digital recording and mixes will get enormous benefits from the warmth and thickness this unit will deliver.

There are various ways you can work with this device and get a large range of results from it. For example, since the modeling of the amplifiers stage in this compressor is very detailed, you can experiment with various

levels of Input Gain, compensating for more or less compression with the Threshold knobs, and for higher or lower output level with Output knob. Set the input level to -16 in order to get 2 - 3 dB of compression, and rise the Output level to get the level you want. This will result in a very transparent compression, with a bit of nice coloration, but not much.

Then you can try setting the Input gain at, say, -4 dB. The compression can then be set to get the same 2 - 3 dB of compression and the Output level again to match the level you need. This time, because of the higher input drive on the unit, the sound will be much more colored, with a fabulous thick, warm character.

Mixing single tracks

This compressor character is incredibly musical on single mix tracks as well (not just on mastering entire mixes), just try it on pretty much every type of material: Vocals, Guitars, Bass, Piano, Strings, etc., with various time constants and 5 - 6 dB of compression. If you want to enhance the articulation of a vocal track try compressing 5 dB with time constant 1. The resulting vocal won't sound heavily compressed, just more consistent and solid.

Keep DC threshold controls near the minimum when you want the most transparent and invisible compression, and move them toward maximum if you want the compressor to be more noticeable and more limiter-like.

Mixing drums

On drums this compressor can be used to process a Stereo submix with superb results, especially in Lat/Vert mode, where you will be able by compressing the sides independently from the center to get the sound of the room exactly the way you want. Since the attack time is very fast, drums compressed with this processor will never sound snappy, allowing for a compression that will sound more natural.

Typical Mastering Setting

If this is the only compressor used in your mastering chain you can start with a setting like the default one or very similar. This will deliver from 1 to 2.5 dB of compression with time constant set at 5. This setting will glue all mix elements together without having the sound to appear compressed. If you want to compress more just rise the Threshold controls and compensate the average gain loss by incrementing the Output level.

In stereo mastering this unit should be used Linked or Lat/Vert, but sometimes 1 - 2 dB of "unlinked" stereo compression can give a nice instability touch to the stereo image that can be very pleasant.

This model has its "own sound" when you just insert it, even without applying any compression (Threshold at zero). So an interesting experiment we invite you to do is just to insert this on the audio processing chain and listen to the very interesting sonic change that is applied by the 670 internal circuits and components.

7.6.2 - Opto Compressor



Think of this compressor as the purest analog modeled compressor in the T-RackS CS package. When you want the sound of analog compression but want to keep the music original sonic texture with the little possible coloration, this is the right choice.

You will notice that no more than a few dB of compression are allowed at low RATIO values. For example with a 2:1 ration you'll not be able to compress more than 3 - 4 dB. This is normal for this device and it's because of the optical device modeled in this compressor that cannot attenuate more than a certain amount. If you need more compression, just raise the RATIO value.

This behavior is also very useful when compressing vocals because the compressor will follow vocals dynamic but will at the same time avoid squeezing the highest passages.

7.6.3 - Linear Phase Equalizer



This is a top class equalizer that can be sonically compared only to the high end digital equalizers used every day in mastering studios. It is so good that in most cases, it surpasses even expensive analog high end equalizers in terms of purity and transparency of the equalization.

Use this equalizer when you need to apply EQ with the least possible negative side effects and when you absolutely don't want the any side effects from equalization.

We suggest using the Linear Phase Equalizer to set the EQ you want to apply on your audio, and then to experiment with the Linear Phase button to evaluate, on the material you're working on, which phase response is the best. Typically, linear phase equalization is more transparent, and material processed with this technique appears less equalized than conventional EQ, so it gives headroom for more pronounced equalization before sounding overprocessed.

Remember that LINEAR PHASE can only be activated if the preference "Oversampling + Linear Phase" is checked.

7.6.4 - Brickwall Limiter



This processor should normally be used at the end of the T-RackS processing chain. Its main purpose is to increase the volume by a few dB, typically from 1 to 3 with the maximum possible quality. You can normally start with CLEAN as this is the more neutral of all styles and then experiment with the various styles included.

If you want very high levels and a more hard character from the limiter when using the CLEAN style, simply lower the RELEASE time toward the very minimum. This will make the limiter work like a clipper, so fully preserving transient energy and impact, but with less drastic effect on the audio. The CLEAN style with very short release times (less than 1 ms) is particularly suitable for Pop, Rock, Hip Hop etc.

Keep slightly higher RELEASE times (around 10 ms) for music that does not need extremely intense levels. Longer RELEASE times, say more than 50 ms are especially useful for some “pumping” effects that are sometimes desirable.

Then you can start experimenting with the other Styles. ADVANCED styles uses a “mix” between warm and smooth saturation and digital limiting. They all are punchier and more colorful than CLEAN, and they can be effective on certain material because the actual limiting effect is less evident, at the expense of some saturation. Experiment with all advanced styles because the right kind of limiting could be a particular balance that only one of these styles provides.

HOW TO AVOID DIGITAL OVERS

Always keep an eye on the RED OVER indicators on the T-RackS CS peak meters. They should NEVER go RED during the entire song. To avoid this, you can simply use the Brickwall Limiter at the end of the chain and dial in a Ceiling value of -0.1 dB, OR enable the D/A Distortion Protection option in the Preferences.

7.6.5 - Vintage Program Equalizer EQP-1A



This is a classic. It provides colorful equalization that can give character and tonal shape with incredible musicality. This equalizer is on the opposite side, in terms of usage, than the Linear Phase EQ. It's intervention is not spot-on and you should think of it more as a “tone control” than a surgical equalizer.

Remember that both Low Frequency and High Frequency BOOST and ATTEN controls are not overlapping, so if you, for example, dial in a low BOOST and ATTEN, both at 5, the resulting response will not be flat. This is a typical characteristic of this device, and it's one of the reasons why it is so loved to this day.

The design is based on a passive EQ network followed by a make up amplifier. The passive EQ network gives a presence and a tonal shaping power that is hard to obtain with conventional “pure” equalizers. For example a very nice touch of presence is given by setting the High filter like this:

- BANDWIDTH: 5
- BOOST: 4 - 5
- HIGH FREQ: 4 - 5 kHz

Like we said for the Vintage Compressor Model 670, also this model has it's “own sound” when you just insert it, even without applying any equalization. So another experiment we invite you to do is just inserting this on the audio processing chain and listen to the very interesting sonic change.

7.7 - Master your room

When mastering with T-RackS, you're making very important decisions about audio processing of the overall balance of the music you're working on, and most importantly, you're making the very last audio intervention before public distribution. If you make a bad decision at this stage, there will not be the possibility to fix it after.

There is another key factor in a correct mastering and mixing process that is often underestimated: your room acoustics. Unfortunately the sonic signature of the room can make even the best monitors unreliable, because of the aberrations and irregularities of the room.

Even the most experienced engineer with the highest quality equipment can ruin a mix, or even more easily, a master, because of the unreliability of the listening environment. There are factors that cannot be compensated for by training or "learning the room."

For example, poor room acoustics can create very narrow peaks and dips in low-frequency that are very difficult to detect with the naked ear. In fact, bass notes for example, will never sound consistent, even if they are recorded evenly, so on certain notes you'll think you have too much bass, on other notes you'll find the bass disappearing.

For these reasons the room's acoustical quality should be considered as important as the quality of all other chain links in your audio production studio, from bedroom studios to high end facilities.

In order to improve sonic accuracy of a room you can apply acoustical treatment to the walls, floor, and ceiling. Doing so can have very high quality results, but at a high cost. These solutions typically include treatments done with various mixed elements such as absorbers, diffusers, bass traps, floating floors, and so on.

Another convenient solution is to use IK's **ARC System 2**, the acoustic correction system in a plug-in. In developing the ARC System, IK Multimedia has teamed up with the leading provider of sound equalization solutions, Audyssey, for the production of an innovative, low-cost and mobile solution to correct the distortion problems caused by room acoustics.

ARC System 2 delivers the most advanced solution to acoustical problems for any DAW-based studio. Combining a professionally calibrated microphone, standalone software that captures sound information and calculates proper room correction, and a multi-platform plug-in. This technology will improve how your studio sounds forever.

ARC System 2 features the revolutionary Audyssey MultEQ®XT 32 technology, which measures acoustical information throughout the listening area in your studio. It then combines this information to provide an accurate representation of the room's acoustical problems. This equalization solution then corrects for both time and frequency response problems, more effectively, and efficiently than any other room correction EQ on the market. The result is a clear and reliable representation of your mix, regardless of the acoustical issues in your studio. This means what you are recording, mixing, or mastering becomes immediately clear and reliable. Your studio sound will improve forever.

T-RackS 5 and ARC System 2 together form an ideal combination, allowing the T-RackS 5 user to trust what he or she is hearing, which is truly the most important factor.

For more information on ARC System 2:

www.ikmultimedia.com/arc

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Chapter 8 – Troubleshooting

Where can I find my IK product Serial Number?

The Serial Number is written on the Registration Card (included with your IK product) or in the email you received from DigitalDelivery@ikmultimedia.com (if you purchased the product as 'Downloadable only version').

IMPORTANT: the number zero can easily be identified in your Serial Number because it is crossed by a line.

Why is the Authorization Manager rejecting my Serial Number?

Probably because of a typo, here are some common errors:

- Typing a 0 ("zero" number) instead of an O ("o" letter).
- Typing 1 ("one" number) instead of an I ("i" letter).
- Typing 2 ("two" number) instead of a Z ("z" letter).
- Typing 5 ("five" number) instead of an S ("s" letter).
- Typing 8 ("eight" number) instead of a B ("b" letter).
- Typing a "." (point) instead of a "-" (minus).

Suggestions:

- If possible, please copy and paste the information.
- Cut off all the leading and trailing spaces.
- Please type all codes in UPPERCASE during the installation and registration process.
- Check that the Serial Number that you are entering is in correlation with the product installed.

How can I authorize my product on another computer?

To authorize your product on another computer just follow these instructions:

- Install the product and the Authorization Manager on that computer
- Run the Authorization Manager

I need to log in into the User Area but I forgot my User Name and Password. What should I do?

You can retrieve your User Area login details in two different ways:

- Run the Authorization Manager and follow the steps until you reach the login page. Here you just have to click on the 'Forgot password?' button and submit your email address. Your login details will be sent to your email account. In case you have checked the 'Remember me' checkbox before and are not able to access the login page, just click on the LOGOUT button.
- Go to www.ikmultimedia.com, click on 'I forgot my username and/or password' and submit your email address. Your login details will be sent to your email account.

I just bought an IK Multimedia software. What should I do to register and authorize my new product?

Just run the Authorization Manager and follow the instructions.

In case you don't have the Authorization Manager, you can download it for free from the Products page of our website at:

www.ikmultimedia.com/am

Chapter 9 – Support

For any questions you may have, please refer to the FAQ webpage at: www.ikmultimedia.com/faq

Here you will find answers to the most commonly asked questions.

To submit a Technical Support Form, go to: www.ikmultimedia.com/support

For other requests such as Product, Sales, or Web info, please go to: www.ikmultimedia.com/contact-us

9.1 – User Area

The User Area is a special section of our website specifically designed for our users.

Here you can easily edit your personal data, manage your product authorizations and licenses, and download the latest IK products updates.

Access your User Area to be able to:

- Edit your Personal data.
- View and download the latest product updates, free content, and Sound Libraries.
- Access any current Promotions.
- View exclusive offers and pricing.
- Manage your earned JamPoints.
- View your Orders.
- Share your AmpliTube presets.
- Download firmware and hardware drivers.
- Access the IK Forum and more.

To access the User Area go to:

www.ikmultimedia.com/userarea

To access your User Area, please login with your User Name and Password created upon registration and emailed to your registered email address. If you need to update your Password, you may do so at the User Area login.

IK Multimedia Production Srl

Via dell'Industria, 46,
41122 Modena
Italy

IK Multimedia US, LLC

590 Sawgrass Corporate Pkwy.
Sunrise, FL 33325
USA

IK Multimedia Asia

TB Tamachi Bldg. 1F, MBE #709
4-11-1 Shiba
Minato-ku, Tokyo 108-0014
Japan

www.ikmultimedia.com

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