



USER MANUAL

Supporting VST/VST3/AU/AAX Native and AAX DSP
Rev. Jun 12, 2020

intellijel



Solid State Logic
SOUND || VISION



ROLI



Valley People

TUBE-TECH
LYDKRAFT DENMARK



DOEPFER

DRAWMER

Buchla

Tonelux[®]



Mutronics Limited

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MS

Marshall
AMPLIFICATION

Softube User Manual

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Your rights to the software are governed by the accompanying software license agreement (End User License Agreement).

Disclaimer

Every effort has been made to ensure that the information in this manual is accurate. However, there are a chance that we have made mistakes, and we hope that you understand that we are only humans. Please let us know about the mistake, and we'll fix it in the mix (or in the next version of this manual).

Support

On the Softube website (www.softube.com) you will find answers to common questions (FAQ) and other topics that might interest you.

Support questions can be posted at <http://www.softube.com>, where we will help you as fast as we can!

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Contents

1	<i>Softube Central</i>	9	6	<i>Active Equalizer</i>	43
	User interface	9		User Interface	43
	Settings	10		Knobs	44
	Softube Central in use	11		Buying Recommendations	45
	System Requirements	11		Credits	45
	AAX DSP Supported Sample Rates	13			
	Plug-In Latencies	14			
2	<i>Preset Collection</i>	16	7	<i>American Class A</i>	46
	Preset Collection	20		User Interface	47
	User Interface	23		Input Section	49
3	<i>User Interface</i>	27		Shape Section	50
	Menu Row	27		Equalizer Section	50
	Key Commands	30		Compressor Section	51
				Output Section	52
				Credits	52
4	<i>Abbey Road Studios Brilliance Pack</i>	31	8	<i>Bass Amp Room</i>	53
	Foreword by Peter Cobbin	31		Introduction	53
	Introduction	33		User Interface	54
	RS127 Rack	33		The Amplifier	55
	RS127 Box	34		The Cabinets	56
	RS135	35		The Mix Panel	57
	Credits	35		Bypassing Amps or Cabs	58
				Block Diagram	59
				Buying Recommendations	60
				Credits	60
5	<i>Acoustic Feedback and Acoustic Feedback Deluxe</i>	36	9	<i>Console 1</i>	61
	Introduction	36		The SSL SL 4000 E Channel	62
	Getting Started	37		More Information	62
	Playing Techniques	37			
	User Interface	38			
	Using MIDI	40			
	FAQ	41			
	Credits	42			

10	<i>Drawmer 1973 Multi-Band Compressor</i>	63	15	<i>Fix Flanger</i>	95
	User Interface.	64		User Interface.	96
	In Use	68		Tweaking tips.	99
	External Sidechaining	69		Credits	100
	Parallel Compression	70	<hr/>		
	Keyboard Shortcuts	70	16	<i>Fix Phaser</i>	101
	Setup Options	70		Overview.	102
	Credits	70		User Interface.	102
<hr/>				Tweaking tips.	105
11	<i>Drawmer S73</i>	71		Credits	107
	User Interface.	73	<hr/>		
	In use	74	17	<i>Focusing Equalizer</i>	108
	Styles description	74		Work Flow	110
	Credits	75		Knobs	111
<hr/>				Meters	112
12	<i>Eden WT-800 Bass Amplifier</i>	76		Differences Between Passive and Active	113
	User Interface.	77		Finally, a Note on Modeling	113
	Cabinets and Microphones	80		Credits	113
	Meters	82	<hr/>		
	Mono and Stereo Operation	82	18	<i>Harmonics</i>	114
	Presets.	82		User interface.	115
	Credits	82		Parameters.	116
<hr/>				Meters	117
13	<i>FET Compressor</i>	83		In use	117
	Introduction	83		Credits	118
	User Interface.	84	<hr/>		
	Tweaking Tips.	88	19	<i>Metal Amp Room</i>	119
	Block Diagram	89		Introduction	119
	Credits	89		User Interface.	120
<hr/>				The Amplifier.	122
14	<i>Fix Doubler</i>	90		The Cabinets	123
	User Interface.	91		The Microphones	124
	Tweaking Tips.	93		The Balancing Stereo Preamps	124
	Credits	94		Mono and Stereo Operation	125
<hr/>				Bypassing Amps or Cabs	126
				Buying Recommendations	126
				Credits	126

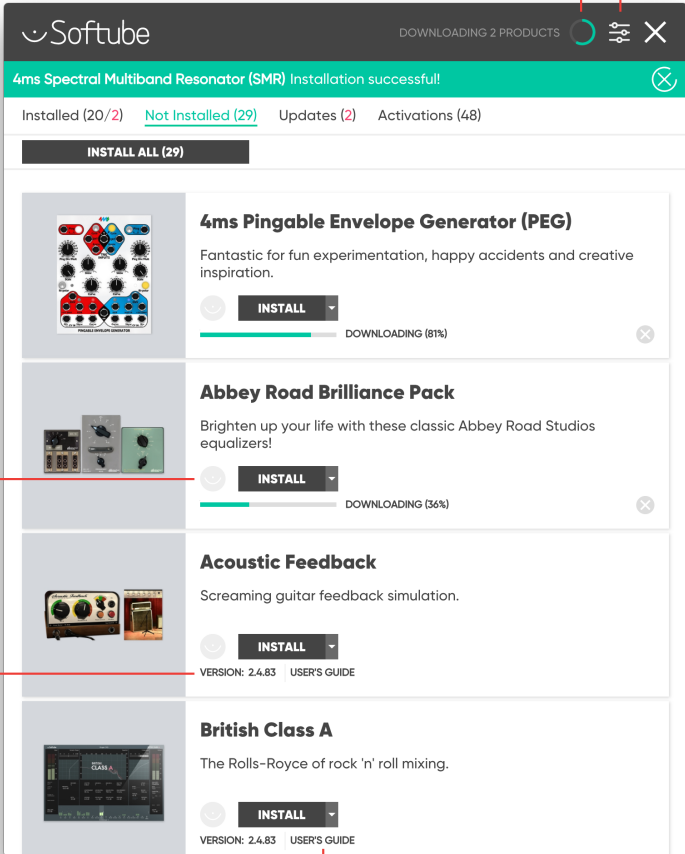
	User Interface at a Glance	190			
	Reverb Parameters	191			
	Presets.	197			
	TSAR-1R Reverb.	200			
	Automation	201			
	Mono and Stereo Operation	201			
	Credits	201			
32	<i>Tube Delay</i>	202			
	Introduction	202			
	User Interface.	203			
	Block Diagram	205			
33	<i>Tube-Tech CL 1B & CL 1B Mk II Compressors</i>	206			
	Foreword by John G. Petersen	206			
	About the CL 1B.	206			
	User Interface.	207			
	Suggested Applications	209			
	Mono and Stereo Operation	209			
	CL 1B Mk II	210			
	Credits	210			
	Credits – Mk II Version.	210			
34	<i>Tube-Tech Classic Channel & Classic Channel Mk II</i>	211			
	User Interface.	211			
	Gain Staging	213			
	Sidechain and the Equalizers	213			
	Tube-Tech Classic Channel Mk II.	214			
	Credits	215			
	Credits – MK II Version.	215			
35	<i>Tube-Tech ME 1B Midrange Equalizer</i>	216			
	Introduction	216			
	User Interface.	217			
	Filter Graphs	219			
	Buying Recommendations	219			
	Credits	219			
36	<i>Tube-Tech PE 1C “Pultec” Equalizer</i>	220			
	Foreword by John G. Petersen	220			
	Introduction	220			
	User Interface.	221			
	Buying Recommendations	223			
	Credits	223			
37	<i>Tube-Tech Equalizer Collection</i>	224			
	Credits – Mk II Version.	224			
38	<i>Valley People Dyna-mite</i>	225			
	User interface.	227			
	Basic Limiting	228			
	Basic Expanding	229			
	Switches	230			
	Knobs	232			
	Limiting Modes.	234			
	Expanding Modes.	237			
	Mono and Stereo Operation	240			
	Buying Recommendations	240			
	Credits	240			
39	<i>Valley People Dyna-mite Slam and Gate</i>	241			
	Dyna-mite Slam	243			
	User interface.	243			
	Parameters.	244			
	In use	245			
	Dyna-mite Gate.	246			
	User interface.	246			
	Parameters.	247			
	In use	248			
	Credits	248			
40	<i>Vintage Amp Room</i>	249			
	Introduction	249			
	User Interface.	250			
	White Amp.	252			

	Brown Amp	253
	Green Amp	255
	Bypassing Amps or Cabs	257
	Buying Recommendations	257
	Credits	257
41	<i>Other Platforms</i>	258
	Reason Rack Extensions	258
	Cakewalk Sonar ProChannel	258
	UAD-2 Powered Plug-Ins.	259
	TC Electronic PowerCore	259
42	<i>SOFTUBE AB'S END USER LICENSE AGREEMENT ("EULA").FOR PLUG-INS AND OTHER SOFTWARE PRODUCTS (ver 2019-06)</i>	<i>cclx</i>

1 Softube Central

Welcome to Softube Central! Here you'll be able to manage installations and activations of Softube plug-ins. Softube Central is configured by default to manage the installations and activations automatically. If you have several workstations and want to manage it manually, you can activate the option in the Settings.

User interface



The screenshot displays the Softube Central interface. At the top, there's a dark header with the Softube logo and a status bar indicating "DOWNLOADING 2 PRODUCTS". Below this, a green notification bar shows "4ms Spectral Multiband Resonator (SMR) Installation successful!". The main content area lists several plug-ins with their respective status icons, installation buttons, and progress bars. Red numbers 1 through 5 point to specific UI elements: 1 points to the status icon in the top right, 2 points to the settings icon, 3 points to the product status icon, 4 points to the version information, and 5 points to the user's guide link.

1. Status icon

2. Settings

3. Product status icon

4. Version

5. User's guide

Status icon

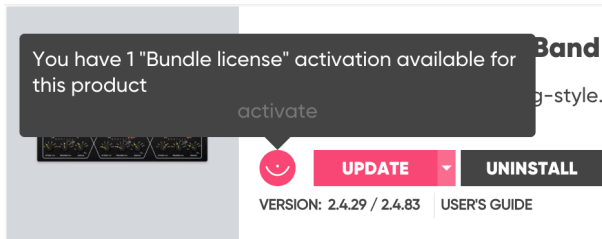
Status icon at the top right corner will inform you of the status of the Softube Central. In case it shows “Disconnected”, please check your internet connection or restart the daemon application from the Softube Central settings.

Settings

Settings can be entered by clicking on the  icon at the top right corner.

Product status icon

Hovering over the Product status icon will inform of the status of the available licenses.



Version

You can see the version of the plug-ins installed and the latest available version at the bottom of every product lane.

User's guide

You can access the user's guide from the product lane as well.

Note: for your convenience, Softube Central detects all available licenses on your iLok account automatically and will prompt you to uninstall products that you don't have licenses for.

Settings

Version	shows the version of the Softube Central installed.
Account details	section shows the registered email, user's name and the country.
PACE details	section shows the status of the backend connection, the version of the daemon installer and PACE version used.
Current locations	shows the iLok or machine based locations that are currently available.
Licenses	section includes a switch with Auto Activate mode. If set to Auto mode, the Softube Central will activate the licenses automatically after plug-in installation. If set to Manual mode, the Activations tab will appear where you'll be able to manage the license activations manually. The Activations tab will show the available activations and locations where the license was activated.

Note: in case the location is unavailable the Softube Central will grey out the location and mark it as unavailable. If it shows the iLok location as unavailable with iLok plugged in, please check if the latest ILOK drivers are installed.

Softube Central in use

After Softube Central installation please launch it from the Applications folder on your Mac computer. It will start automatically on PC.

Please follow the instructions on additional installations the Softube Central requires. When the installations are completed, go to Not Installed tab and click Install All button, Softube Central will install all purchased plug-ins and the plug-ins with active 20-day DEMO activations.

Softube Central will install and activate the plug-ins for you. Now you can close the Softube Central and make some music!

Note: it's helpful to run Softube Central once in a while to check if any updates are available.

Note: Softube Central public beta version doesn't have the update license functionality yet. We are working on it. Please handle the license update through the iLok manager.

System Requirements

These are the system requirements for all Softube plug-ins. **Please check the latest requirements on www.softube.com.**

All native versions

- Mac OS X 10.12 or newer
- Windows 64-bit, versions 7, 8 or 10
- Please Note: Softube plug-ins support 64-bit hosts only, and a 64-bit OS is required. Therefore, for example, Pro Tools 10.x is no longer supported (contact support for latest 32-bit compatible instalers).
- Intel Core 2 Duo, AMD Athlon 64 X2 or newer
- Screen resolution larger than 1280x800
- 8 GB RAM or more is recommended, as well as at least 8 GB hard disk space for installation (individual plug-ins take less space, while sample libraries may require additional disk space. Specific information available on each instrument's page).
- Softube account
- iLok account
- Internet access for downloading the plug-in (s) and managing the license(s)
- iLok generation 2 and up, if you intend to use an iLok dongle for authorizing your plug-ins (note that an iLok dongle is not a requirement)
- In theory, any 64-bit VST, VST3, AU, or AAX (11.0.2 or higher) compatible host application should work. However, due to plug-in host differences between DAWs - and our own rigorous testing standards - we only officially test our plug-ins and instruments in the most recent versions of Pro Tools, Logic Pro, Cubase, Live, Studio One, Digital Performer, Reaper and Sonar. Softube plug-ins are not

tested in the non-listed systems, but they most likely work as long as the system requirements are met. We cannot guarantee a solution for issues in unsupported systems.

- Supported sample rates: 44.1, 48, 88.2, 96, 176.4 and 192 kHz, in both mono and stereo.
- The most recent maintenance release of the DAW application is recommended.

Please Note: Pro Tools 10 is not officially qualified by Avid higher than OS X 10.8

AAX DSP version

For AAX DSP you need to meet the above requirements for the Native versions. You also need to have an Avid HDX processing card. For plug-ins available for AAX DSP, see [Which Softube plug-ins support AAX DSP?](#)

AAX DSP Supported Sample Rates

The following sample rates are currently supported for the AAX DSP format.

	44.1/48 kHz	88.2/96 kHz	176.4/192 kHz
Active Equalizer	√	√	√ (mono)
Acoustic Feedback	√ (mono)	√ (mono)	√ (mono)
Acoustic Feedback Deluxe			
Bass Amp Room	√	√	√ (mono)
FET Compressor	√	√	√
Focusing Equalizer	√	√	√
Metal Amp Room	√	√	
Passive Equalizer	√	√	√
Spring Reverb	√	√	√
Summit Audio Grand Channel	√	√	√
Summit Audio EQF-100	√	√	√
Summit Audio TLA-100A	√	√	√
Tonelux Tilt	√	√	√
Tonelux Tilt Live	√	√	√
Trident A-Range	√	√	√
TSAR-1 & TSAR-1R Reverb			
Tube Delay	√	√	√ (mono)
Tube-Tech Classic Channel	√	√	√
Tube-Tech CL 1B	√	√	√
Tube-Tech ME 1B	√	√	√
Tube-Tech PE 1C	√	√	√
Valley People Dyna-mite	√	√	√
Vintage Amp Room	√	√	√ (mono)
White Amp	√	√	√ (mono)

√ = sample rate supported in both mono and stereo.

√ (mono) = sample rate only supported in mono. Use “multi-mono” for stereo.

Plug-In Latencies

This table shows the plug-in latency in samples. These numbers are also reported to the DAW.

	44.1/48 kHz	88.2/96 kHz	176.4/192 kHz
Active Equalizer	8 samples	0 (no latency)	0 (no latency)
Acoustic Feedback	0	0	0
Acoustic Feedback Deluxe	0	0	0
Bass Amp Room	3	0	0
Drawmer 1973	4	0	0
Drawmer S73	4	0	0
FET Compressor	4	0	0
Fix Flanger	2650/2884	5292/5760	10584/11520
Fix Doubler	4	0	0
Focusing Equalizer	4	0	0
Metal Amp Room	3	0	0
Modular	2	2	0
Mutator	3	0	0
Passive Equalizer	8	0	0
Spring Reverb	0	0	0
Summit Audio Grand Channel	8	4	0
Summit Audio EQF-100	4	0	0
Summit Audio TLA-100A	4	4	0
Tonelux Tilt	0	0	0
Tonelux Tilt Live	0	0	0
Transient Shaper	8	8/9	16/17
Trident A-Range	4	0	0
TSAR-1 & TSAR-1R Reverb	0	0	0
Tube Delay	3	0	0

	44.1/48 kHz	88.2/96 kHz	176.4/192 kHz
Tube-Tech Classic Channel	4	4	0
Tube-Tech CL 1B	4	0	0
Tube-Tech ME 1B	12	8	0
Tube-Tech PE 1C	12	8	0
Spring Reverb	0	0	0
Summit Audio TLA-100A	4	4	0
Valley People Dyna-mite	4	0	0
Vintage Amp Room	3	0	0



FACTORY PRESET USER PRESET ACOUSTIC BASS ALL-ROUND BASS BRIGHTNESS CREAMY DARK DIST DRIVE DRUM KIT
 ELECTRIC BASS ELECTRIC GUITAR FAT FEMALE VOCALS GLUE HOWARD WILLING JOE CHICCARELLI KEYS KICK ASS PRESETS KICK DRUM
 LEAD SYNTH LOUD MALE VOCALS MASTERING MIX PIANO PRESENCE PUNCH RICH SATURATION SMOOTH SNARE DRUM SOFT
 SUBTLE SYNTH BASS TAPE WARM

Name	Rating	Description	Collection
bass 2 -2VU	★★★★★	More low-end body. Tips and Tweaks: Increase the input...	JOE CHICCARELLI
Bass Push THD3	★★★★★	Boost the lowest bottom end. Usage: Output level is increa...	HOWARD WILLING
Bear Master THD1	★★★★★	Lighten up your mix with slightly less bass and low mids. ...	HOWARD WILLING
Cassette Sat -1VU	★★★★★	Round off the top end with overloaded tape distortion. Tip...	JOE CHICCARELLI
Classic 70s 0VU	★★★★★	Warm loudness boost. Tips and Tweaks: Set Speed Stabili...	JOE CHICCARELLI
<input checked="" type="checkbox"/> Classic Analog -1VU	★★★★★	Smooth, creamy presence and subtle width to your mix. TL...	JOE CHICCARELLI
Clean Master THD1	★★★★★	Make the mix smooth and tight with slightly less midrange...	HOWARD WILLING
Darker Acoustics THD1	★★★★★	Make the mix round and soft in by attenuating the upper ...	HOWARD WILLING
Drum Buss Fwd THD1	★★★★★	Bring out the subtle presence and add length to room sou...	HOWARD WILLING
Drum Machine Analoged	★★★★★	Make your beats pop without nasty peaks. Tips and Tweak...	JOE CHICCARELLI
EGtr Transient Smooth	★★★★★	De-harsh your electric guitar with tape compression. Tips ...	JOE CHICCARELLI
Light Dist 1 0VU	★★★★★	Round off your tracks' high frequencies with subtle, shim...	JOE CHICCARELLI
Light Dist 2 0VU	★★★★★	Add compressed tape-treble to make the upper midrange ...	JOE CHICCARELLI
LoFi Grit 1 0VU	★★★★★	Soften the midrange and add silky tape-treble for smooth ...	JOE CHICCARELLI
LoFi Grit 2 0VU	★★★★★	Create flanger/chorus-sounding roundness. Tips and Twe...	JOE CHICCARELLI
LV Smoothing -3VU	★★★★★	Avoid harshness by rolling off upper midrange and top end...	HOWARD WILLING
Mix Buss Warmer 1 0VU	★★★★★	Warm up the track by attenuating the mids and very gentl...	JOE CHICCARELLI
Mix Buss Warmer 2 0VU	★★★★★	Boost the midrange to add presence and width to the mix...	JOE CHICCARELLI



JOE CHICCARELLI

Classic Analog - 1VU

★★★★★

Smooth, creamy presence and subtle width to your mix.


Tips and Tweaks: Set Speed Stability to Stable, Crosstalk to 25% and use Dry/Wet for subtle brightening of the mix.

Info: Subtle loudness curve/tape compression, Crosstalk and Speed Stability adding width and chorus.



2 Preset Collection

The Preset Collection is a tool to organize your presets in logic, simple, advanced or mysterious (?) ways (it's up to you!), or just a simple mechanism to save your favorite sounds and easily browse through artist's presets.

You can either use the simplified version in the menu bar of each Softube plug-in, or you can press the open window icon  to open the full Preset Collection.

Browsing presets

Use the ◀▶ buttons in the menu row at the top of the plug-in to step through presets. Click ▼ to open a menu to select presets. By default, presets are sorted by “collection”, usually by artist or theme.

Saving Presets

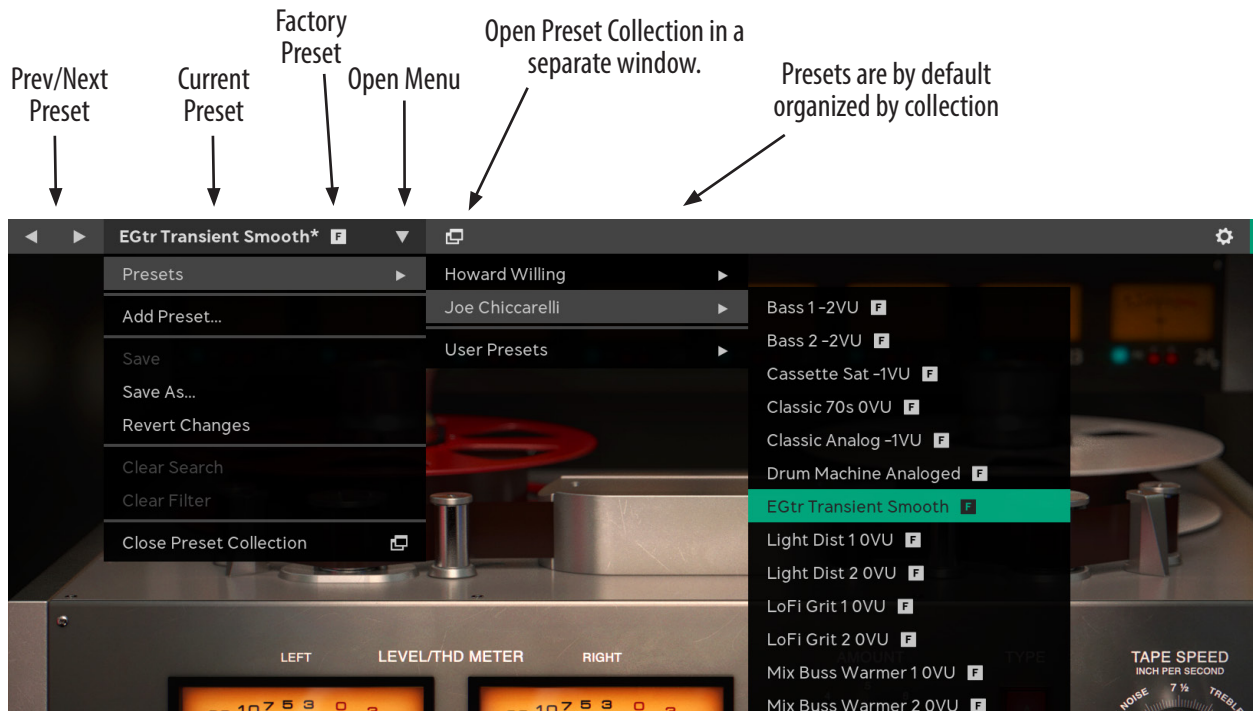
Press “ADD PRESET...” in the dropdown menu to save the current settings as a new preset. Type the name of the preset and press enter. If you’ve made changes to a current preset and wish to overwrite that preset, just press “SAVE”. Press “SAVE AS...” if you want to save it with a new name.

Searching for Presets

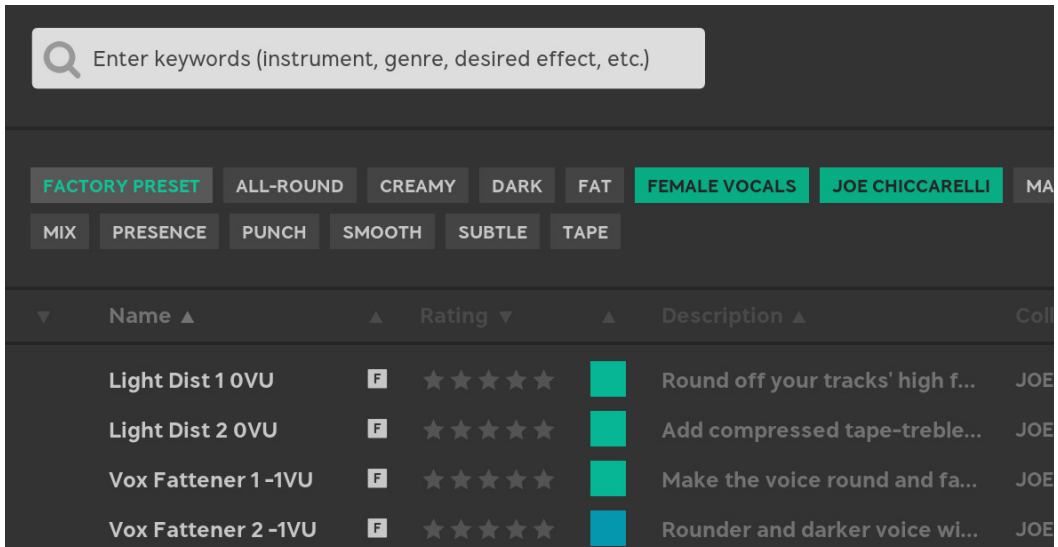
If you want to find a specific preset and have a lot of presets, there are many smart ways to search for preset, and all of them require that you open the Preset Collection by clicking the 📁 button.

Search by tag

All presets have a number of tags associated with them, in general they describe the function (“DISTORTION”, “EQ”, “COMPRESSION”), the use case (“FEMALE VOCALS”, “BASS”) and the character (“CREAMY”, “DARK”), etc. In plug-ins that use modules, such as MODULAR, the tags also include which modules are used by the preset (for example “SATURATION KNOB”)



Browse presets directly from the menu bar at the top of the plug-in.



Two tags (“Female Vocals” and “Joe Chiccarelli”) has been selected to show all Joe’s presets suitable for female vocals.

Search box

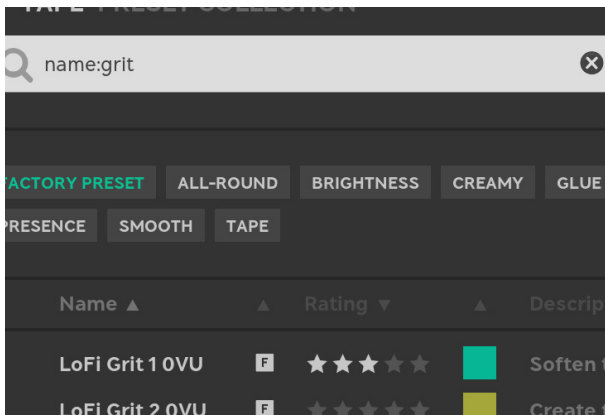
A search in the SEARCH BOX will search through all metadata (name, tags, description, etc). If you want to narrow your search, you can specify what you want to search for by using a qualifier, such as “name:” or “description:”.

Possible search qualifiers are *name*, *desc*, *description*, *tag*, *tags*, and *collection*.

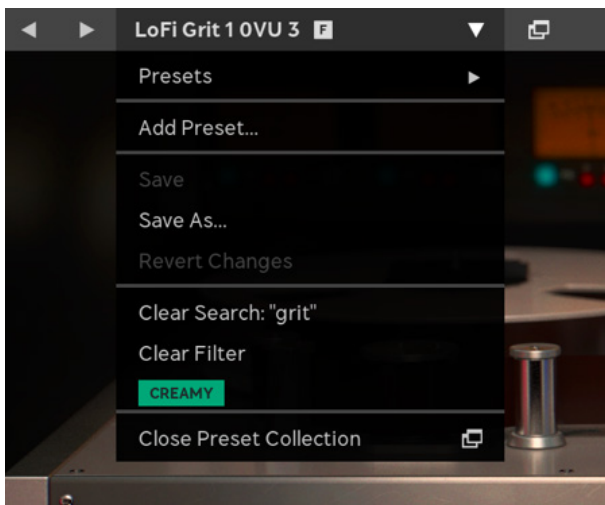
Search in the menu bar

When you have selected a subset of presets, for example by searching for “vocals” or using a tag, the search results are available directly from the plug-in’s menu bar. That means that you can easily step through the presets using the ◀▶ buttons.

Open the menu to see the search criteria or clear the search.



Type “name:” before the search criteria to limit the search to the name of the preset.



The menu shows the current search criteria, currently selected tags, and options to clear the filter and search criteria.

Plug-in Settings vs. “Metadata”

A Softube preset consists of two parts: the plug-in’s settings and the description of it, what we call “*metadata*”. Metadata is everything that’s not included in the plug-in’s settings, for example preset name, description, color, rating, tags.

When you make a new preset by clicking “ADD NEW PRESET...” the only metadata that you save is the preset name. You need to open Preset Collection to add other metadata, such as a description and tags. If you instead use “SAVE AS...” when you save a preset, the metadata (for example tags) in the currently selected preset will be carried over to the new preset.

Whenever you change the settings in the plug-in, the preset name will be marked with an asterisk * to indicate the current settings are different from the current preset. It also indicates that the preset has changes that aren’t saved. You need to click “SAVE” or “SAVE AS” to save those changes.




Asterisk (*) indicate that the settings of the plug-in is different from the saved preset.

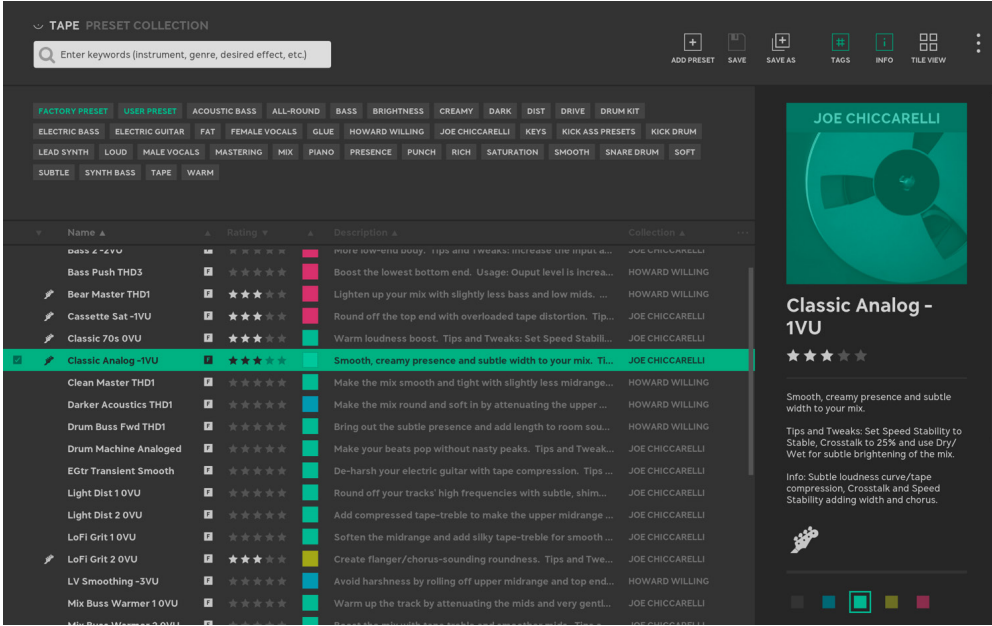
When you make changes to the metadata, you don’t have to save these. All metadata changes are saved immediately in the preset database.

*) An asterisk looks like this: *

*) *ibid.*

Preset Collection

Click  to open the Preset Collection. Here you can organize, colorize, tag, add icons and images and sort your presets.



The screenshot shows the 'TAPE' Preset Collection interface. At the top, there is a search field and a toolbar with icons for 'ADD PRESET', 'SAVE', 'SAVE AS', 'TAGS', 'INFO', and 'TILE VIEW'. Below the search field is a 'Tags pane' with various category buttons like 'FACTORY PRESET', 'USER PRESET', 'ACOUSTIC BASS', etc. The main area is a 'Preset pane' containing a table of presets. On the right, an 'Info pane' displays details for the selected 'Classic Analog - 1VU' preset, including a star rating and tips/tweaks.

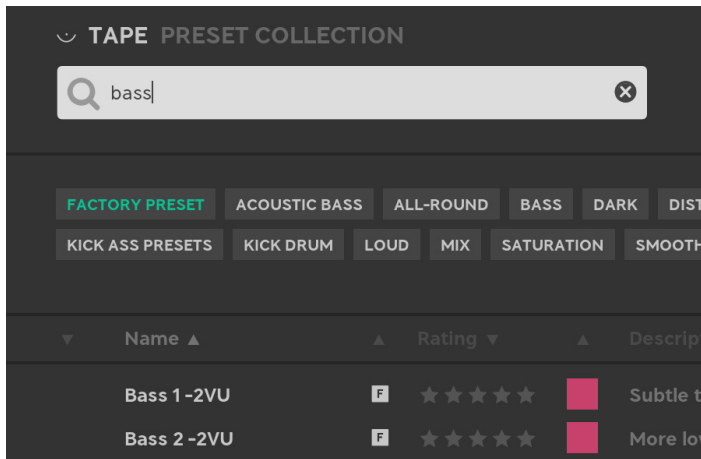
Labels and arrows pointing to the interface elements:

- Search field**: Points to the search input at the top left.
- Add or save presets**: Points to the 'ADD PRESET', 'SAVE', and 'SAVE AS' icons.
- Set display options**: Points to the 'TAGS', 'INFO', and 'TILE VIEW' icons.
- Menu**: Points to the vertical ellipsis icon on the far right of the toolbar.
- Tags pane**: Points to the row of category buttons.
- Info pane**: Points to the right-hand panel showing preset details.
- Preset pane**: Points to the main table of presets.
- Group edit**: Points to the edit icon in the table header.
- Category**: Points to the 'Name' column header.
- Name**: Points to the 'Rating' column header.
- Preset name**: Points to the 'Description' column header.
- Color**: Points to the 'Collection' column header.
- Factory preset indicator**: Points to the small square icon in the 'Name' column.
- Description**: Points to the 'Description' column header.
- Collection**: Points to the 'Collection' column header.

Name	Rating	Description	Collection
Bass 4-2VU	★★★★★	More low-mid energy. Tips and Tweaks: Increase the input...	JOE CHICCARELLI
Bass Push THD3	★★★★★	Boost the lowest bottom end. Usage: Output level is increa...	HOWARD WILLING
Bear Master THD1	★★★★★	Lighten up your mix with slightly less bass and low mids. ...	HOWARD WILLING
Cassette Sat -1VU	★★★★★	Round off the top end with overloaded tape distortion. Tip...	JOE CHICCARELLI
Classic 70s 0VU	★★★★★	Warm loudness boost. Tips and Tweaks: Set Speed Stabili...	JOE CHICCARELLI
Classic Analog -1VU	★★★★★	Smooth, creamy presence and subtle width to your mix. TI...	JOE CHICCARELLI
Clean Master THD1	★★★★★	Make the mix smooth and tight with slightly less midrange...	HOWARD WILLING
Darker Acoustics THD1	★★★★★	Make the mix round and soft in by attenuating the upper ...	HOWARD WILLING
Drum Buss Fwd THD1	★★★★★	Bring out the subtle presence and add length to room sou...	HOWARD WILLING
Drum Machine Analoged	★★★★★	Make your beats pop without nasty peaks. Tips and Tweak...	JOE CHICCARELLI
EGtr Transient Smooth	★★★★★	De-harsh your electric guitar with tape compression. Tips ...	JOE CHICCARELLI
Light Dist 1 0VU	★★★★★	Round off your tracks' high frequencies with subtle, shim...	JOE CHICCARELLI
Light Dist 2 0VU	★★★★★	Add compressed tape-treble to make the upper midrange ...	JOE CHICCARELLI
LoFi Grit 1 0VU	★★★★★	Soften the midrange and add silky tape-treble for smooth...	JOE CHICCARELLI
LoFi Grit 2 0VU	★★★★★	Create flanger/chorus-sounding roundness. Tips and Twe...	JOE CHICCARELLI
LV Smoothing -3VU	★★★★★	Avoid harshness by rolling off upper midrange and top end...	HOWARD WILLING
Mix Buss Warmer 1 0VU	★★★★★	Warm up the track by attenuating the mids and very gentl...	JOE CHICCARELLI

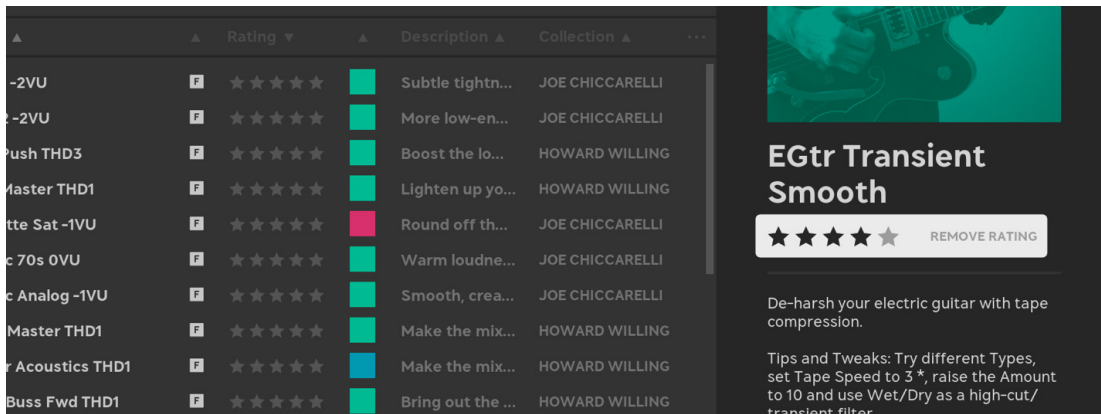
Workflow

For a plug-in with hundreds of presets, the easiest way to start is to type something in the search field, such as “bass”, or select an appropriate tag.



Type in the search field to find presets.

Use the keyboard up/down arrows to step through presets, and if you find something you like (or dislike), change the rating of that preset so that you can easily find it later.



Click on the stars in the Info pane to change the rating. It is also possible to right-click the preset and change rating from the context menu.

Philosophy

The main ideas behind Preset Collection are

1. A preset name doesn't give enough information about how to use a preset. Sometimes you need more info, for example what to listen for, how to tweak it, in which context etc.
2. Everyone wants to organize their presets in different ways.
3. Tags are a simple way to create “folder like” structures, but without being limited by placing the preset in a single folder. A tag can be a use case, a project name, or just about anything!

With the preset's description you'll be able to add info, for instance how many dBs of gain reduction you need for the drum bus to really glue together, and with names, tags, ratings, categories, and colors you can organize those presets any way you want. The tags become powerful if you want to organize presets after projects you're working on. Tag each preset you make/use with the project name, and you'll have an extra dimension to use when you browse presets.

Don't be afraid to add or change metadata in the factory presets. (And it's always possible to restore the original metadata later if you want to!)

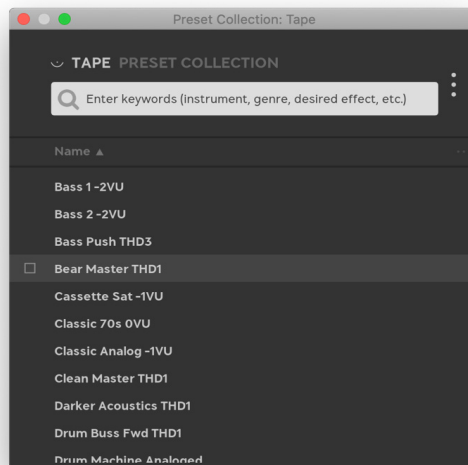
Organizing presets

All factory presets come with tags and color, but you can remove those colors and tags and organize them in other ways. You can mark your favorite presets with a high rating, a specific tag (“kick ass presets!”, “great for accordion”), a color, or a category, to make them easier to find later. You can select several presets at once to, for exam-

ple, reset their color, and use your own color scheme. All fields, except `COLLECTION`, are possible to change.

Customizing the Preset Collection

You can decide yourself how much or how little info you want to show in the Preset Collection, and these settings are stored globally.



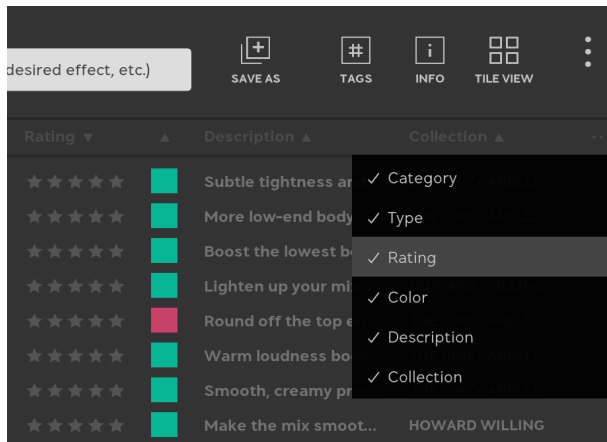
The minimal view of Preset Collection, only preset names and search field are visible.

Showing/hiding info panes

Click on `#` or `i` to show and hide the Tags and Info pane, respectively.

Showing/hiding columns

Click on the menu (`•••`) next to the columns header to turn on/off columns in the preset pane.





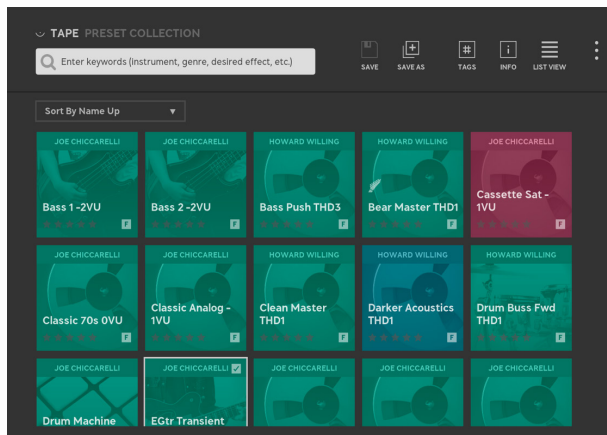
Select which columns are visible from the menu.

Sorting

Use the up/down arrows (▲▼) next the the column header to change preset sorting.

Tile view/List view

Click on Tile View  to show presets as tiles, or List View  to get back to the default mode.





The Tile View.


Resizing tags pane or window


You can easily resize the Preset Collection window or the size of the TAGS pane by clicking the edge and drag it.

User Interface

 **Search Row** Searches in name, description, tags, etc. To specify a particular field to search in, use one of these qualifiers: *name*, *desc*, *description*, *tag*, *tags*, and *collection*, for example “*collection:chiccarelli*”



 **Add Preset** Create a new, empty, preset from the current plug-in settings.

 **Save Preset** Overwrite the selected preset with the current plug-in settings . Only possible with user presets. If the current preset is a factory preset, use “SAVE AS” instead.

 **Save As** Save the current plug-in settings, together with the selected presets meta data (tags, description, etc) under a new name.

 **Tags** Show/hide the TAGS pane.

 **Info** Show/hide the INFO pane.

 **Tile view,**
 **List view** Switch between a list of presets (LIST VIEW) or tiled images (TILE VIEW)

⋮ Menu Opens the menu with some additional options. See below for Menu options.

Menu Options

Add Preset Add a new preset.

Save Overwrite the selected preset with the current plug-in settings.

Save As Copy the selected preset to a new preset with the current plug-in's settings.

Show/Hide Tags Show/hide the TAGS pane.

Show/Hide Info Show/hide the INFO pane.

Show Tile View Show the TILE VIEW instead of LIST VIEW.

Import Preset(s) Import presets from file, for instance if you downloaded a “.softubebundle” file from www.softube.com

Export Selected User

Preset(s) Exports the currently selected presets to a “.softubebundle” file, so that you can send them to a friend.

Enter Group Edit

Mode Let's you select several presets at once, which is useful if you want to

export a batch of presets, or change tags, description or other meta data in several presets at once.

Preferences Opens additional preferences, listed below.

Preferences

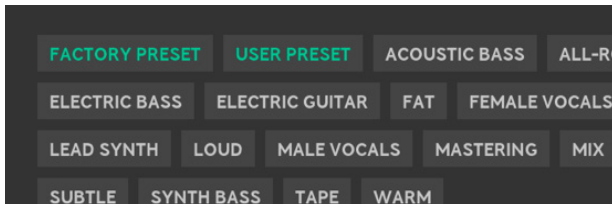
Warn When... Turn warnings on/off when for example deleting or overwriting presets.

Show Images In Info Panel Turn off to hide the image in the INFO PANEL.

Group presets by collection/group presets by category Change how presets are grouped in the presets menu in the plug-in menu bar.

Pro tip: select “Group Presets By Category” to get a flat list for all uncategorized presets in the menu bar.


Tags pane



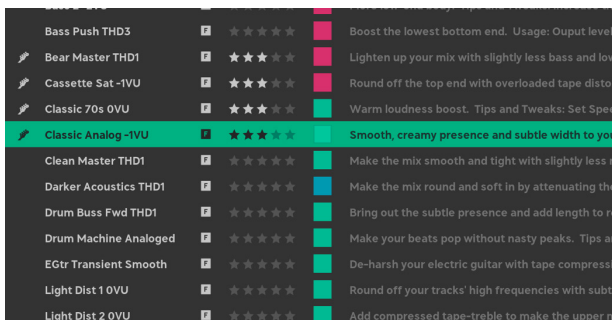
Use the Tags pane to filter presets by tag.

The TAGS PANE lets you filter presets by selecting one or several tags. Select for example the tag “MALE VOCALS” if you want to find all presets suitable for male vocals.

You can select several tags at once.

You can add tags to a preset by clicking “ADD TAG 


Presets pane



The Presets pane shows all presets in the current search or filter.

The PRESETS pane contains all presets in the current search. You can sort presets, select a preset, or edit a preset from the presets pane. The PRESETS pane can either be visualized as a list, or as a “TILE VIEW” with images for each preset.

Tip: Double click on the preset name to change it.

To sort presets in the LIST VIEW, click on  for a column to sort down or up for that column. In the TILE VIEW, open the “SORT BY...” drop down to select sorting.

Context Menu (Right Click on a Preset)

Right-click on a preset to bring up a context menu with the following options:

Save Save preset

Save As... Save as a preset with a new name.

Revert Changes Revert all changes made to the preset.

Rename... Rename the current preset

Export... Export the current preset

Delete Delete the current preset

Restore Factory

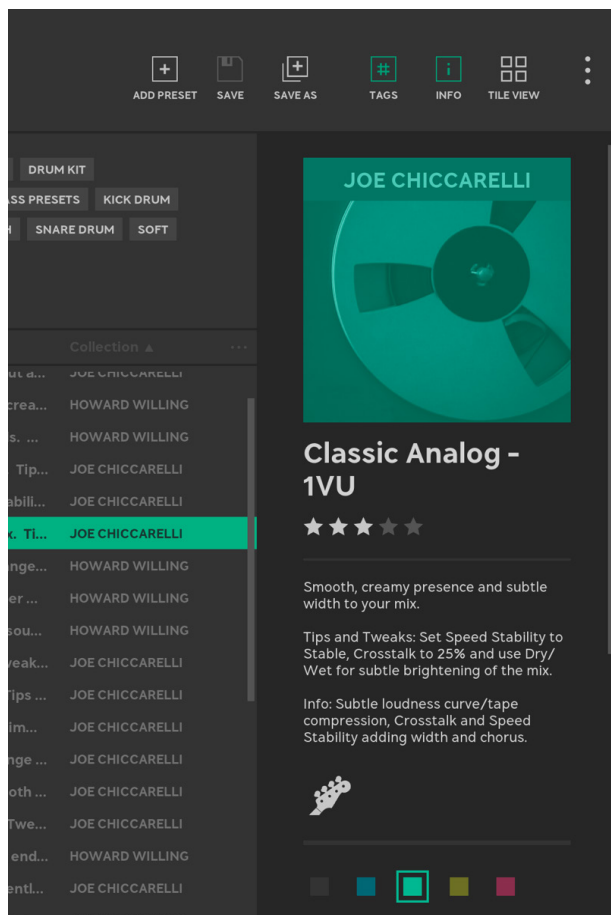
Metadata If metadata edits, for example color, tags or ratings were made to a factory preset, you can revert them here.

Category Sets the category of a preset. By default, no categories have been set.


Rating Sets/changes the rating (0-5 stars) of a preset.

Color Sets/changes the color of the preset.

Info pane



The Info pane contains all metadata for the currently selected preset and lets you edit these fields.

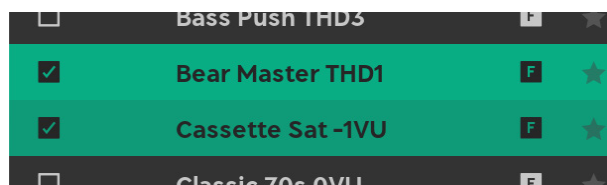
In the INFO pane you can edit the metadata of the selected preset. You can change image, name, rating, description, category, tags and color. Whenever it is possible to change a field, a pen icon  will appear. All changes made to the description of the plug-in are saved immediately.

Change Image

The image needs to be in PNG format, and will be converted to black-and-white on import.

Group Edit Mode

It's easy to edit several presets at once, for instance if you want to change rating or add a tag to multiple presets. Enter the GROUP EDIT mode by selecting several presets in the check box on the left. Once in GROUP EDIT mode, you can use SHIFT + click to select a range of presets.



Select several presets by clicking the check box to the left.

In GROUP EDIT mode it is possible to edit several presets at the same time:

- Change rating, color, and category
- Append text to the name or description
- Add tags, or remove tags common for all selected presets

3 User Interface

SOFTUBE PLUG-INS ARE “what you see is what you get” products. You should be able to intuitively learn the products within minutes, so that you can work fast and efficient with them. There are a couple of things that remain the same for all of our plug-ins, like keyboard commands and menu row. These will be explained in this chapter. For detailed information of a particular plug-in, please see its chapter.

Menu Row

In the bottom of the plug-in interface, you will see a thin black row with some buttons. We’ll use the Dyna-mite plug-in as example, but the same goes for all plug-ins.

About Box Open the “About” Box with version info.

Value Display Displays the knob value when the mouse is pointing at the knob.

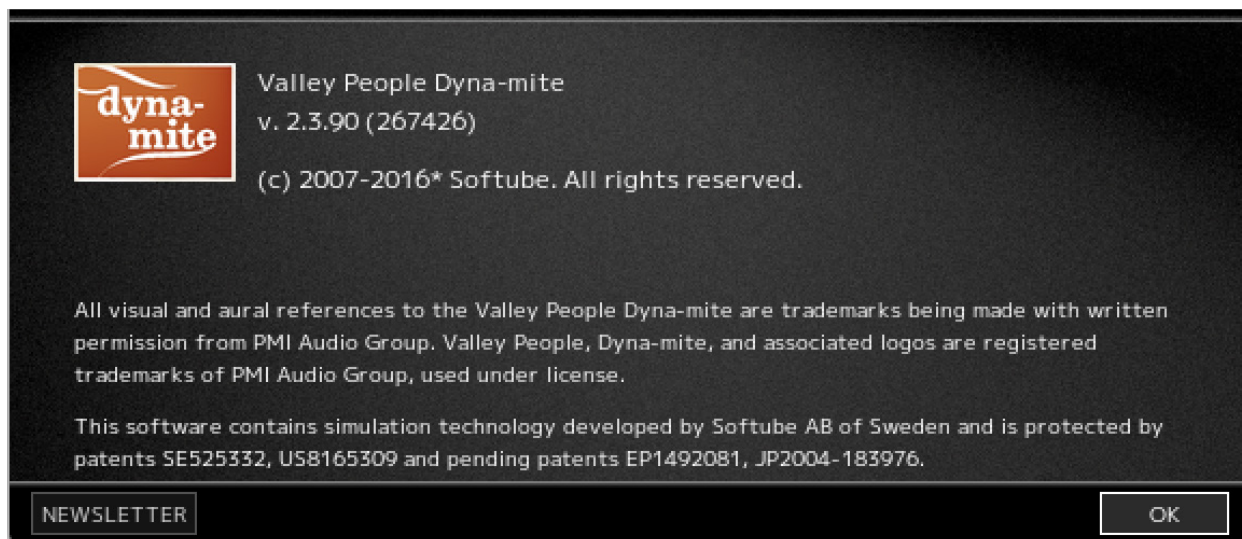
Setup Changes global options for all instances of that plug-in.

Quick Guide Opens the Quick Guide (if applicable) in a PDF reader.

Open Manual Opens the User Manual in a PDF reader.



1. “About” box with version information
2. Value display
3. Setup
4. Open quick guide (if applicable)
5. Open manual in PDF reader



About Box

Clicking on the Softube logo opens the “About” box, with information about the current version, trademarks and legal yada-yada. That’s not really interesting, but the cool stuff is the buttons on the bottom row.

Newsletter Opens the web browser and lets you sign up for our newsletter, so that you will be the first to know about new products, tips & tricks, updated versions, etc.

OK Closes the “About” box.

Setup

In the Setup window you can change settings that will affect all instances of that particular plug-in. If you for example de-select the “Show Value Display” option in the Valley People Dyna-mite plug-in the value display will be off for all Dyna-mites on your system until you select that option again.

The different options vary between Windows and Mac, and also different formats and plug-ins. The most common options are:

ENABLE EXTERNAL SIDECHAIN SUPPORT: If you want to use external sidechaining in VST2.

SHOW VALUE DISPLAY: Enables the parameter and value display in the bottom row of the plug-in.

PREFER COCOA GUI: 32-bit Mac Only. Tries to use a newer graphics system. Select this if you experience problems with the user interface.

You need to restart your host software (DAW) before the changes to fully take effect!

If you messed something up and manually need to set these options, you'll find them in text format in the following locations:

MAC OS: ~/Library/Application Support/Softube

WINDOWS: username\Application Data\



Key Commands

All numbers and labels in the plug-in are clickable. This allows you to easily select a setting by clicking on the wanted value. Hovering above a label will turn the mouse pointer into a pointing hand.

Mouse

Up/Down or
Mouse Wheel Change a parameter, such as a knob or a switch.

Keyboard

Fine Adjust ⌘ (Mac) or CTRL (Win), while changing the parameter value.

Reset to Default ALT, while clicking on the knob or fader.

Pro Tools Specific Commands

Automation Control
Window CTRL+ALT+WIN+click (Win) or CTRL+⌘+ALT+click (Mac)

Show Automation
Track CTRL+WIN+click (Win) or CTRL+⌘+click (Mac)

Plug-In Specific Key Commands

In many plug-ins, you can SHIFT-click on a knob or a switch to get some extra functionality. Below are some examples. See the chapter for each product for more information.

Metal Amp Room

SHIFT-click and drag a mic will move both mics simultaneously.

All Amp Rooms

SHIFT-click in the cabinet background will change cabinet (or amp) without any animations

Focusing Equalizer

SHIFT-click on the **Low** and **High Cut** faders will move both faders simultaneously.

Modular

In Modular, hold SHIFT and click to add several modules at once to the rack.

Mutronics Mutator

SHIFT-click on a parameter changes that parameter for both the left and right channels.

Drawmer 1973 Multi-Band Compressor

SHIFT-click on a parameter changes that parameter for all three bands. SHIFT-clicking the **Status** label will solo that band.



4 Abbey Road Studios Brilliance Pack

Foreword by Peter Cobbin

“Welcome to a little Abbey Road brilliance.

Throughout the ‘60s there were a number of small boxes dotted around Abbey Road Studios, which were simply known as ‘Brilliance’ or ‘Presence’ boxes. In this era, engineers were looking for ways of adding presence to their recordings, especially in popular music. Enter the Brilliance boxes: these simple passive equalizers were portable versions of the grey RS127s that were rack-mounted into the studio control room patchbays. They were designed to give Abbey Road Studios recording engineers additional frequencies that were not found on the EMI REDD studio mixing desks of the time.

Meet Lester

Lester Smith is a technical engineer at Abbey Road Studios and, amongst other things, is the custodian of our vintage equipment and microphone collection.

In recent years, while working on various Beatles and John Lennon-related mix projects, Lester introduced me to these little boxes. My assistants Mirek Stiles and Sam O’Kell had observed that, in various ‘60s setup sheets, the prevailing EQ was marked as “RS127”. This equalizer was the 127th item made in-house by EMI for the Recording Sector. Keen to try these, Lester blew 40 years of dust from some of the boxes and wired them into our patchbay via the old ‘Siemens’ connectors and presto – instant zing! The large cut and boost control made it very easy to hear an immediate presence. Put simply – they’re brilliant!

The Sixties were an adventurous period in our history. Rule books were put aside and considerable experimentation by engineers helped shape ways and means of producing sounds not heard before. Following this spirit of experimentation, our latest plug-in suite provides today’s engineer and musician with a bundle of little boxes straight from this era.

The RS127s

The green and grey RS127s have identical circuits and controls. The grey has a chrome rack handle that made it convenient to plug into the control room patch-bays. There were usually two of these in each room, but due to their popularity additional EQ was often needed, and so stand-alone boxes were made to achieve this. These were painted green. Because of their portability, they were often used throughout the studio complex; not just in the studios themselves but also in the mastering, transfer and post-production rooms.

The RS127 is well documented in the highly recommended “Recording The Beatles” book (Kevin Ryan & Brian Kehew, Curvember Publishing, 2006) which also refers to the change in line level and EMI standard impedance that has taken place since the early Sixties.

By happy accident, when comparing the green and grey units, we heard a dramatic difference when one of them was passed through an EMI interfacing transformer. The transformer “effect” exaggerated the EQ curves.

And so, in the spirit of the ‘60s, we have included this “not so precise” effect by providing two RS127 plug-ins: the grey rack version and the green standalone unit with the transformer effect. I have seen old setups where engineers used two RS127s in series for heavy EQ treatment, and I can highly recommend using these EQs on guitars, keyboards and vocals. If you find that +/-10 dB is not enough, do what they did back then – use two of them!

The RS135

The RS127 was primarily made to supplement the 5 kHz REDD desk EQ, and proved to be immensely useful and popular. However, there was still a need to boost frequencies between 5 kHz and 10 kHz.

EMI 8 kHz boost boxes were widely used but little was known about them. Only recently, when putting this project together, Lester removed the 8 kHz label (incidentally, made with a Dymo prototype) and found on the reverse that these boxes were in fact originally RS135s. Our researchers have shown that these boxes were designed for a 16.4 kHz boost, but modifications made at the time choked this frequency by half – enabling engineers to boost signals at 8 kHz. We have decided to add this to our ‘Brilliance’ suite, thus completing the range of presence EQs that were significantly used at Abbey Road Studios in the ‘60s.

Peter Cobbin, Director of Engineering, Abbey Road Studios April 2008

Introduction

The Brilliance Pack brings these classic EQs to modern audio workstations, providing authentic emulations of the hardware units.

The pack consists of three plug-in modules:

RS127 Rack

The RS127 Rack plug-in is a recreation of the original Brilliance Control rack modules which were installed in the studio control rooms.

RS127 Box

The RS127 Box plug-in recreates the standalone Brilliance Control and includes the exaggerated EQ curves caused when it is used with an EMI interfacing transformer.

RS135

The RS135 plug-in recreates the 8 kHz boost boxes, which were originally designed as 16.4 kHz boost units but later modified by Abbey Road engineers to work at 8 kHz.



RS127 Rack

The RS127 Rack plug-in recreates the original Brilliance Control. It boosts with a broad or 'blunt' bell-curve response and gives a 'medium blunt' bell-curve on the cut settings.

Frequency (kc/s) This control sets the centre frequency at which the plug-in will boost or cut in kilocycles per second - (kHz). The available settings are 2.7, 3.5 and 10 kc/s (kHz). The default setting is 3.5.

Gain (dBs) This control sets the amount of boost or cut which takes place at the selected frequency in decibels. It can be set to +/- 10 dB in 2 dB steps. The default setting is 0.



RS127 Box

The 'RS127 Box' plug-in recreates the standalone RS127 unit along with effects caused by interaction between components in the RS127 and an interfacing transformer.



Gain

Frequency

Frequency (kc/S) As on the RS127 Rack plug-in, this sets the centre frequency at which the plug-in will boost or cut in kilocycles per second (kHz). As on the RS127 Rack, the available settings are 2.7, 3.5 and 10 kc/s (kHz), but the interfacing transformer causes the centre frequencies to change and this effect has been replicated in the RS127 Box

plug-in. The actual centre frequencies are 2.9 kHz, 4.2 kHz and 11.5 kHz.

Gain (dB) This control sets the amount of boost or cut which takes place at the selected frequency in decibels. It can be set to +/- 10 dB in 2 dB steps. The default setting is 0.

The 'transformer effects' can be heard clearly at the higher boost settings, at which the plug-in gives a sharper 'Q' or bandwidth. At these settings, there is also significantly more boost than given by the same settings on the RS127 Rack plug-in.



RS135

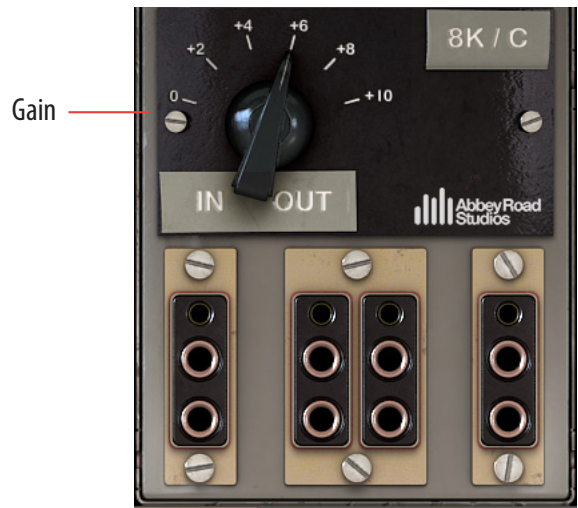
The RS135 plug-in provides an accurate model of the EMI 8 kHz boxes. It has a single control, which sets the amount of boost, and provides the same 'medium sharp' bell-shape characteristic as the original boxes.

Gain The Gain control sets the amount of boost in decibels. It provides up to 10 dB of boost in 2 dB steps. The default setting is 0.

Credits

Abbey Road Studios – product development and documentation, **Niklas Odelholm** – modeling, **Oscar Öberg** – DSP programming, **Torsten Gatu** – framework programming, **Arvid Rosén** – framework programming, **Ulf Ekelöf** – 3D rendering.

EMI AND ABBEY ROAD ARE TRADEMARKS OF EMI (IP) LIMITED. REDD, RSI27 AND RSI35 ARE TRADEMARKS OF EMI (IP) LIMITED.



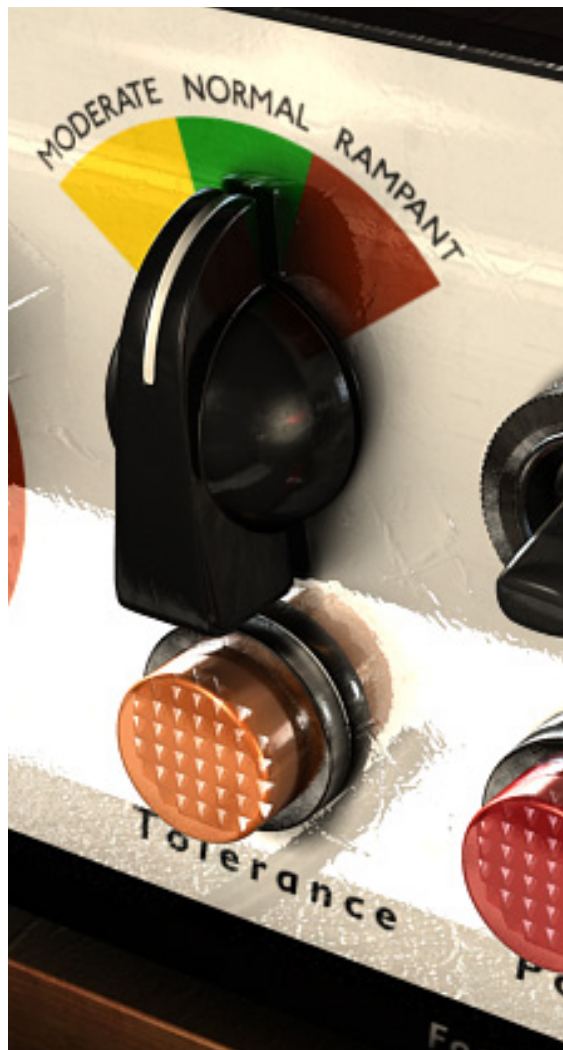
5 Acoustic Feedback and Acoustic Feedback Deluxe

Introduction

Real acoustic feedback appears when the sound waves from the cabinet hit the strings with enough energy (accomplished by playing loud enough) and with the right phase (accomplished by positioning the guitar at the correct distance from the cabinet). The difficult part with acoustic feedback on moderate volumes is therefore to get feedback on the desired notes—it is easy to get feedback, but if you for example need to get it on the high D, you have to find the exact distance so that the string vibrations and sound waves don't cancel each other. The rumour is that when Robert Fripp recorded the feedback parts on Bowie's *Heroes*, he made markings all over the floor in order to find the right position for different notes...

The Acoustic Feedback simulator is a mathematical model of real acoustic feedback. We have modeled how the sound waves hit the strings, how the strings interact with the amplifier and how the sound waves travel through the air back to the guitar again. But the only thing we couldn't model is the distance between the guitar player and the cabinet (computer?). So to solve this we made an “auto-positioner”, pretty much like a Brian Eno running around with the cabinet so that Robert Fripp didn't have to move between the different markings on the floor.

So with all these parts put together, we present to you the Acoustic Feedback plug-in!



Getting Started

1. Make sure that you have a good guitar signal in to your audio interface. Use a line box or DI with a high impedance input (more than 500 kilohms).
2. If the guitar signal is too low (with peaks lower than 12 dB), use a volume plug in or compressor to boost the signal.
3. Insert the Acoustic Feedback on your guitar track in your favourite host software, and set all controls to the 12 o'clock position.
4. Insert Vintage Amp Room or your favorite amp modeler after the Acoustic Feedback.
5. Try some single string notes, such as the E on the 9th fret on the G string. Strike the note and wait for the feedback to appear. If it takes too long time, increase the **Feedback** knob. If the feedback is too fast and aggressive—decrease it instead. Bend the note to see how that affects the feedback.

Playing Techniques

The basic techniques in order to get the most natural sound from the Acoustic Feedback simulator are:

- *Don't push it.* If there isn't any energy left in your strings (ie. the strings have nearly stopped moving) the plugin won't be able to feedback on the correct note. End your notes after a while by muting the strings or pick another note. Don't let them ring infinitely.
- *No hum or noise.* Noise interferes with the feedback and shortens the lifetime of it. A short sustain (ie. dead strings) will also shorten the feedback lifetime.
- *Play nice,* single string and always end your notes deliberately. Sloppy playing is not rewarded!
- Notes played on low-numbered frets usually have longer sustain than notes that are played high up (for example above 15th fret).
- *No chords!*

If you do have a sloppy playing style, you will have to increase the **Tolerance** control to make the effect trigger more easily. The downside is that it doesn't always end very naturally.

The feedback effect volume is independent of the input guitar volume! If you have a low guitar input volume, you will need to lower the Mix control in order to get a good balance between the clean guitar signal and the feedback effect.



1. Mix
2. Feedback
3. Tolerance and
Tolerance indicator

User Interface

Mix The mix between the DRY (original) and WET (feedback) guitar sound.

Feedback Adjusts how aggressive the feedback is

Tolerance Adjusts how easy it is to achieve feedback (and how tolerant it is with sloppy playing).

Tolerance Indicator When the light is on, you got full feedback. When it starts to fade it's time to end the note.

Mix

Controls the balance between the feedback and the original signal. Outputs only the clean guitar sound when set on **DRY**, and only the feedback effect when set to **WET**.

Set this control so that the balance between the guitar and the feedback sounds natural. This knob is usually set somewhere in the green area.

Try playing around with **Mix** set fully on **WET**. You can get some pretty interesting results from that...

Feedback

Controls how fast the feedback builds up. This corresponds in the real world to the volume of the real amplifier. Low settings will give subtle harmonics, and high settings will give an aggressive feedback.

Feedback = **SUBTLE** (Yellow area): The feedback will usually not buildup and the effect will be very subtle.

Feedback = **NATURAL** (Green area): The feedback will have a slow buildup time and act quite naturally, although it might be hard to get feedback on all notes.

Feedback = **WILD** (Red area): The feedback will have a faster buildup time and it will be quite easy to get the feedback effect on most notes, but sometimes it won't sound very natural.

How the **Feedback** control works depends greatly on your guitar, preamp, volume, what note you play etc.

Tolerance

Controls how easy it is to maintain the feedback while changing notes, bending, doing vibratos etc. This corresponds in the real world to a combination of the distance between the speaker and guitar, but also the

volume of the amp. When set at the lowest setting, it will only feedback on stable notes, and when set at the highest setting, it will feedback on most things, even noise or background hum.

Tolerance = **MODERATE** (Yellow): Feedback only on very stable notes.

Tolerance = **NORMAL** (Green): Feedback on vibratos, bends and slides. This usually gives the most natural sounding feedback.

Tolerance = **RAMPANT** (Red): Wild and unpredictable feedback. Sometimes.

If you find it too hard to get feedback, you can do one of three things:

1. Increase the **Feedback** control (to make the effect appear faster)
2. Increase the **Tolerance** control (to make the effect more tolerant towards note changes and different playing techniques.
3. Increase the input volume by increasing the gain of your preamp or using a volume plugin before Acoustic Feedback.

The timbre of the feedbacking tone will change slightly if you change the **Tolerance** control during a feedback. This can sound abrupt and unnatural, so it's better to only change the **Tolerance** when no note is feedbacking.

Tolerance Indicator

The **Tolerance Indicator** indicates when a feedback is on its way, and when the feedback is about to die. If the **Tolerance Indicator** starts to fade, make sure that you end your note deliberately, before any strange sounds appear. This is specially helpful when you have a high **Feedback** or **Tolerance** setting.

Harmonic Selector (Deluxe only)



Controls which harmonic that will dominate the feedback. In the real world, this can sometimes be controlled by changing the distance from your guitar to the amplifier. In a world built up by zeros and ones, this is done by pushing a button (or five).

16' = SUBOCTAVE. This knob will add a subtle harmonic one octave below the fundamental. If only this harmonic is selected, the plugin will never start a feedback, but if it's selected in conjunction with other harmonics you'll get a very growly feedback. Use this as a you would use the "nitro" button in your favourite car racing video game.

8' = FUNDAMENTAL. This will give you feedback on the fundamental, ie., the same note as you are playing.

5 1/3' = ONE FIFTH ABOVE FUNDAMENTAL. This will give you a feedback on the fifth above the fundamental. Works best in conjunction with the 16' button.

4' = ONE OCTAVE ABOVE fundamental

2 2/3' = ONE OCTAVE AND A FIFTH ABOVE fundamental.

Using MIDI

Although the Acoustic Feedback doesn't have any built-in MIDI support, it supports automation, and is therefore easy to control via MIDI in a number of hosts. The most effective (and fun) way to control the Acoustic Feedback is by using an expression control pedal.

Expression Pedal

It is very easy to control the feedback effect by assigning the **Feedback** control to an expression pedal (a MIDI volume pedal).

Connections

If you don't have an expression pedal connected to your system, you will need:

1. Expression pedal. The cheap ones cost about €30 and work perfectly fine.
2. Expression pedal input. These connectors are usually found on MIDI keyboards, so take a look at the back of your keyboard to see if you have a 1/4" stereo jack with "Expression" or "Foot Control" written on it.

Connect the expression pedal to the MIDI device, and use the MIDI control section in your host software to assign the control number and MIDI channel to the feedback effect.

Feedback control

1. Assign the expression pedal to the **Feedback** control.
2. Set the **Tolerance** control to 1 (NATURAL).
3. When ever you want to get the feedback started, push the pedal (hence increasing the **Feedback** control).
4. The feedback effect will decrease as soon as you lower the **Feedback** control.

MIDI Automation and Touch-Ups

To really get the sound you want without controlling it with an expression pedal, you might want to use the automation feature in your audio sequencer.

1. Start out by setting the Acoustic Feedback on a setting that you find works most of the time and record your guitar track.
2. For parts that you don't want any feedback on, set the **Feedback** control to 0%.
3. On parts where you desperately need feedback, but your current setting doesn't give it to you, set the **Tolerance** to **MAX**, and gradually increase the **Feedback** during the duration of the note you're playing.
4. Just before the note ends, do a quick fade out with the **Feedback** control, and if necessary set the **Tolerance** control back to its previous settings.

This way you can have “uncontrollable” settings, but by just using them on specific notes, you will avoid troublesome note endings and strange noises.

Sometimes it is easier to use the **Mix** control to fade the effect in and out, but the **Feedback** control will always give more realistic results.

FAQ

I don't get any feedback? What am I doing wrong?

It could depend on a lot of things, but you can start by setting the all controls fully clockwise; **Mix** = **WET**, **Feedback** = **WILD** and **Tolerance** = **RAMPANT**. Then you will only hear the feedback effect with the most aggressive settings on the feedback algorithm. Make sure that you have enough input gain on your guitar signal, and strike a single note, for instance the open B string, and listen for the feedback.

When you have learned to get the feedback, decrease the controls one by one until you get a natural sounding feedback.

The feedback is too aggressive and doesn't build up in a realistic way

If the feedback comes too fast, decrease the **Feedback** control. It can be a bit difficult to find that sweet spot since it depends on your guitar, but it's usually located somewhere in the green area.

The feedback doesn't blend in with the sound

This has probably to do with one of two things:

1. The **Mix** control is set too high so the feedback effect is too loud compared to the guitar signal. Decrease the **Mix** control.
2. The **Feedback** control is set too high, so that the feedback doesn't build up naturally. Decrease the control.

The feedback doesn't stop when I change notes

If the **Tolerance** switch is set at **RAMPANT**, lower it to **NATURAL** or **MODERATE**. The **Tolerance** controls how easy the feedback aborts when you change notes.

If it doesn't help, and you really want the feedback to abort, make sure that you strike the strings when you change notes. Hammer-ons or pull-offs are not always enough to abort the feedback effect.

The feedback ends in mysterious ways...

Decrease the **Tolerance** control and make sure that you end your notes deliberately. If you mute the strings using your left hand, make sure that you don't end up with a harmonic instead. If you for example play on the 13th fret, you might end up with a 12th fret harmonic when you mute the strings, which will cause the plugin to feedback on the 12th fret harmonic note.

How to feedback on chords?

You don't. Well, sometimes it works, sometimes it doesn't. If you need a feedback on a chord, you could record it by using two takes. On the first take you play the chord without the feedback effect, and on the second take you play a single string from the chord with the feedback effect. With some creative mixing you can make this sound very realistic.

How do I control what harmonic the effect feedbacks on?

There aren't any "intelligent" algorithms to decide harmonic, it's more of a "survival-of-the-fittest" kind of thing. The strongest harmonic content of the note from your guitar usually survives in the end. The basic rule is that lower notes usually have a high harmonic feedback, and high notes have a low harmonic (or the fundamental) feedback.

How do I get that big Woodstock sound?

The Acoustic Feedback together with the White Amp from Vintage Amp Room is a good start, you can make the amp sound a little larger by setting the mic in a far-field position, but to really get that arena feeling you will have to add some stereo slap-back delays and a bit of stadium reverb.

Credits

Ulf Kilberg – mathematical modeling and research.
Niklas Odelholm – modeling and sound design. **Oscar Öberg** – modeling. **Torsten Gatu** – interface and framework programming. **Arvid Rosén** – framework programming. **Per Connman** – prototype modeling. **Ulf Ekelöf** – 3D rendering and graphics.

6 Active Equalizer

THE ACTIVE EQUALIZER IS MODELED from a Swiss console equalizer, a Filtek Labo mk.5, with three adjustable bands, a low cut, and a high cut. It is a very solid construction and is very different from the Passive Equalizer. Where the knobs of the Passive Equalizer all influence each other (much like the tonestack of a guitar amp), the Filtek was built to minimize such effects. For instance, if you set the frequencies of the Low and Mid-filters so that they overlap, and cut both bands at -16dB , you won't get a doubled effect (-32dB) – you will still just have a cut at around 16dB . This is very different from how other (especially digital) equalizers usually work. And it is of course a big part of its sound.

For more info about this product, please see the chapter about the Focusing Equalizer.

User Interface

The Active Equalizer has three full parametric bands, a low frequency band (50-500 Hz), a mid band (380-3800 Hz) and a high frequency band (1.8-18 kHz) all controlled with a gain control (-16 to 16 dB), a frequency selector and a width/bypass switch. You also get two cut filters (18 dB/oct) and an output volume control.



- | | |
|--------------|------------------------|
| 1. Low band | 5. Gain control |
| 2. Mid band | 6. Frequency select |
| 3. High band | 7. Width/Bypass select |
| 4. Output | 8. Low cut |
| | 9. High cut |

Knobs

The three bands are equal, apart from their frequency ranges, and they all contain a gain control (labeled “+ dB –“), a frequency control (labeled “Hz” for the Low and Mid band, and “kHz” for the High band), and a width and bypass control (the switch beneath the knobs).

Gain Control Boost or cut in 2 dB steps (from –16 to +16 dB). Please note that all knobs are placed as if they are up side down. It isn’t our fault – the original looked that way!

Frequency Select Use this to set the center frequency of the bands. Unlike most of our stuff, where we model a real unit with all its quirks and the labels never match reality, this sturdy unit actually does what it says it does!

Width/Bypass Select The switch lets you select between a sharp curve (leftmost position), **BYPASS** (middle position) and a blunt curve (rightmost position).

Low Cut An 18dB/octave filter with selectable bypass, ranging from **BYPASS**, 80 Hz, 180, 240, 320 to 500 Hz.

High Cut An 18dB/octave filter with selectable bypass, ranging from 5 kHz, 8, 10, 12, 15 kHz and finally **BYPASS** in the 3 o’clock position.

Output Output volume. Goes from nothing to +12 dB. At the 12 o’clock position, it’s set at 0 dB, and going clock wise will increase the output volume by +2dB for each mark.

Please note that all gain controls range from –16 to +16dB, which is a quite big range, so please take it easy with the gain controls. In many cases, 6 or 9dB is the maximum gain you’ll need.

Buying Recommendations

If you like the sound of the Active Equalizer and would like to get your hands on it, you should look for the Filtek Labo mk.5 console equalizer. It's a remarkably small and versatile equalizer with a very distinct sound.

ALL PRODUCT NAMES USED ARE TRADEMARKS OF THEIR RESPECTIVE OWNERS, AND IN NO WAY CONSTITUTES AN ASSOCIATION OR AFFILIATION WITH SOFTUBE. FILTEK AND LABO TRADEMARKS ARE SOLELY USED TO IDENTIFY THE PRODUCTS WHOSE SOUND WAS STUDIED DURING THE SOUND MODELING DEVELOPMENT OF THE PASSIVE-ACTIVE PACK.

Credits

Oscar Öberg – modeling. **Arvid Rosén** – modeling. **Niklas Odelholm** – graphics design. **Torsten Gatu** – concept. **Ulf Ekelöf** – graphics rendering. Thanks to **Stefan Fandén** and the crew at Deluxe Music for letting us borrow the gear!



7 American Class A

WIDE-OPEN, AIRY, LIVING ELECTRONICS. If you will it, it is no dream with the American Class A. Based on the greatest classics of American mixing console hardware, this channel strip is fast and easy to work with, and brings a truck-load of character to the party. With a shape section featuring gate and transient designing, a stellar four-band EQ with proportional Q and range control, a fascinating compressor design which can be switched between feedback and feed forward operations, and a warm, crunchy output drive section, no modern studio is complete until this sonic option is at hand.

Design Philosophy

The American Class A follows a format very familiar to Console 1 users, and this plug-in is indeed also a Console 1 channel strip. While we wanted to stay true to the original's sound and character, we also wanted to expand on some functionality, like the Shape and Drive sections, and get a perfect combination of versatility and character. You may think that a stepped EQ with seven frequencies per band is an unnecessary limitation in today's digital world, but a lot of the character of this console comes from the way you use it, and limited options in the EQ force you to bring out that character!

The compressor topology is also a major part of the sound. It's punchy and firm and has a unique style. It simply sounds fantastic! By making the shape section from the same topology and circuits as the compressor, we made sure that the tone of the original is always

present, regardless of which part you use.

All in all, we're very proud of this. It features both extremely fine models of the hardware, as well as new concoctions, such as **SHAPE** and **Drive Character**, but they all breathe the same air and speak the same language. And on top of that, we added some nice visualizations. What's not to like?*

User Interface

For a Console 1 user, the user interface should feel very familiar. It consists of 5 sections; **INPUT**, **SHAPE**, **EQUALIZER**, **COMPRESSOR**, **OUTPUT**, and a large screen that visualizes the audio and settings.

On top of the screen you have the display section, which can be switched between **WAVE** and **CURVES**.



* Well, if you don't like the sound of the original console, we don't think you are going to like the sound of American Class A...

Input Meter Shows the peak level of the input signal (post **In Gain**). The top line is the peak hold level, white bar is peak level, and blue shows the RMS level.

Gate Curve Shows the knee of the **Gate**. The meter beneath displays the gain change of the entire Shape section.

- o 0 dB is in the middle, and if you increase **Sustain** or **Punch** you'll see that the white meter is showing a positive gain change (moves to the right), which means that you're increasing the audio level. Turn on the **Gate** and you see a negative gain change.

EQ Curve Shows the EQ settings as well as a frequency analyzer view of the output signal. For stereo signals, the analyzer displays the left signal in white and right in red/orange ("red is right", as always!)

Compressor Curve Shows the knee for the compressor, as well as the current gain reduction. 0 dB (no gain reduction) is all the way to the right.

Output Meter Works just like the input meter, but on the output. Output meter sits after the Volume.

View Wave and

View Curves By clicking on **View Curves** you'll get the above mentioned graphics, select **View Wave** and you'll get

a wave form view. The wave form displays four different waveforms:

1. **INPUT PEAK** (grey)
The Input Peak level shows your input level (post In Gain control). For stereo audio it shows the maximum of the two channels.
2. **OUTPUT PEAK** (blue)
Same as Input Peak, but for the output.
3. **RMS LEVEL** (white line)
The RMS of the output signal.
4. **GAIN CHANGE** (red)
The combination of the gain reduction from the compressor and the gain change from the Shape section.



Input Section

Aside from the normal controls, **Low Cut**, **High Cut**, **Phase Invert**, and **In Gain**, you can also turn on **Filt To Comp** to filter the audio that goes to the compressor's sidechain.

Parameters

LP Low pass filter, 500 to 20000 Hz.
Low pass filter is disabled in its maximum position.

HP High pass filter, 20 to 600 Hz.
High pass filter is disabled in its minimum position.

Filt To Comp Engage to filter the compressor's side chain. This means that the filters will no longer affect the main signal path, and will only be used to send to the compressor. If you also need the filters in the main signal path you can use the filters in the **EQUALIZER** section.

Phase Invert Engage to invert the phase of the input.

Shape Section

The shape section is not modeled from existing hardware. We have instead chosen to use the compressor hardware to build our own gate and transient shaper. That gives us a shaper that's extremely versatile but has the same unique DNA and sound as the compressor. Technically speaking, we've used the very special RMS detector, VCA and circuit solutions from the compressor, and built a completely new thing out of it!

Parameters

Gate Sets the gate's threshold. Disabled when set to its minimum position.

Gate Release Sets the release time for the gate, from 0.3 to 3 seconds.

Hard Gate Adjusts the ratio and knee of the gate. **Hard Gate** has higher ratio and sharper knee, and is more suitable for percussive sounds.

Sustain Increase to add more sustain to the audio.

Punch Strengthens or softens the sound's transient, the onset of the sound.

Ext S/C When engaged, the external side-chain is routed to the GATE part of the SHAPE section.

Shape On Engages or bypasses the SHAPE section.

Equalizer Section

The equalizer is based on a mastering grade 19" equalizer, instead of the console equalizer. With seven frequencies and 11 gain settings per band, the four bands offer great versatility, but at the same time force you to keep within the same limits as the original console, thereby giving you the same classic sound. For more fine tuning and mastering purposes, the original 19" unit has a **Gain Multiplier** which allows you to use finer steps than the 2-3 dB steps offered by the console version.

Parameters

Frequency Sets the frequency of the current band

Gain Sets the gain of the current band. The Gain has no effect if the LOW or HIGH BAND is set to "CUT".

Low/High Type Select between shelving, bell, or cut filters.

Gain Multiplier Sets the steps between different gain settings. At x1 the gain can be set to +/- 0, 2, 4, 6, 9, and 12 dB. At x0.5 it can be set to +/- 0, 1, 2, 3, 4.5, 6, and at x0.25 it can be set to +/- 0, 0.5, 1.0, 1.5, 2.25, and 3 dB.

Please note that the parameter value read-out always corresponds to the x1 setting.

Order Sets the order of the three main sections, **SHAPE**, **EQ**, and **COMPRESSOR**. **SHAPE** will always come before **COMPRESSOR**, but the position of the **EQ** can be shifted to before, in the middle of, and after the dynamics processors.

Equalizer On Engages or bypasses the **EQUALIZER** section.

Compressor Section

The compressor is faithfully modeled from the console compressor and has a couple of interesting features. The **New Type** switch allows you to switch between feed forward

(“NEW”) and feedback (“OLD”) types of compression. In feed forward mode you get a very predictable and firm compression, while the feedback mode has a lot of overshoots and slightly more unpredictable results. The feedback mode sound is punchier and more aggressive, while feed forward is more suited to bus compression duties, vocals, etc.

Parameters

Release Sets the release time of the compressor, from 0.3 to 3 seconds.

Ratio Sets the ratio of the compressor. In general, the actual ratio is often less than what the panel says it is.

Parallel Engages the parallel compression.

Threshold Sets the threshold of the compressor, from +10 to -20 dB.

Hard Knee Adjusts the knee of the threshold. A hard knee has a sharper transition between compressing and non-compressing states. A soft knee as a gradual transition where the ratio gradually increases around the threshold. For most applications, **Soft Knee** is where you want to set it.

New Type Switches between feed forward (“NEW”) and feedback (“OLD”) modes. Feed forward is generally smoother and more predictable, feedback is punchier and more aggressive.

Ext SC Routes the external sidechain to the compressor.

Compressor On Engages or bypasses the **COMPRESSOR** section.

Output Section

The output section features the distortion from the preamplifiers of the original unit, with an added **Drive Character** feature that allows you to control the tonality of the distortion. The original distortion can be a bit “fizzy”, and by increasing **Drive Character** you’ll get a more solid and fat tone.

Parameters

Drive Increase for more distortion. At “5” it will start to clip at 0 dB, pre output volume. Turn it off completely by setting it at 0.

Drive Character Increase for more high frequency distortion. At the minimum position (-10), you’ll get a thick low frequency distortion, which has the same sort of character that a really underspecified, worn-out, transformer has. At +10 it’s a “normal” full band distortion.

It is very effective to tweak the Drive Character according to the source material, if you for instance have a vocal track, you might want to set it so that the body of the vocals distort, but the “s”-sounds remain undistorted. You do that by starting from the minimum position and slowly turn up Drive Character. The sweet spot is usually somewhere between -3 and +3.

The Drive Character is not a part of the original hardware, and if you want it as “true to the original”

as possible, you should set it at its maximum value (+10).

Volume Adjust the output volume of the plug-in.

Credits

Kim Larsson – modeling. **Björn Rödseth** – modeling and graphics programming. **Patrik Holmström** – framework programming. **Niklas Odelholm** – sound and graphics design, modeling, and project management. **Ulf Ekelöf** – 3D modeling. **Pavel Shyrinskykh** – presets.

8

Bass Amp Room

Introduction

WHEN WE DESIGNED BASS AMP ROOM we had two goals in mind. The first goal was to give the user the best possible sound quality and state-of-the-art physical modeling. The other, equally important goal was to give the user a plugin that was intuitive and allowed for a really fast work flow. Time is money, but cutting down on time shouldn't have to mean that you need to accept poor results. At least not with Bass Amp Room. And having an amp model that behaves just like the real thing will definitely speed up the work process, since you probably already know how to use it.

In short, you need the same skills to master this software as you need when dealing with the real studio setup. You can get back to doing what you do best, because music production should be about using your ears, not your computer skills.

WE HAVE IMAGINED TWO TYPES OF USERS. The first wants the best possible amp modeling and is willing to spend time tweaking the sound of each bass track. By working the mics, amp and D.I. tone controls, and, perhaps most importantly, the blend between the D.I. and amp, this user gets full control over the sound without ever losing its authenticity.

The second user is one who, just like the previous user, wants the best possible amp modeling, but recognizes the fact that he/she cannot spend too much time tweaking it. That is why we have, for example, only a single tone control knob on the D.I. and a one-knob limiter. Don't be fooled by the simplicity, however; you still have the ability to create a unique signature sound. The strength lies in the very easy individual blocks (the amp, the D.I., the cabs and mics) and the ability to mix and match between them.



User Interface

Apart from a good sound, work flow is a crucial element of our design – the amps and mics should work just as they do in a real studio. We have tried to imitate this as much as it is possible in a computer environment, and if you have experience working in real studios, you will notice that Bass Amp Room handles and sounds just the way you expect it to.

Amp Panel (Top Area)

In the Amp Panel, you can alter the settings of the amp by clicking the knob and dragging the mouse up and down, or left to right. Switches can be switched either by clicking on them, or by click and dragging the mouse. In some hosts you can change the behavior of the mouse, but the default behavior of Bass Amp Room is the one described above.

Room View (Middle Area)

In the Room View you can do two things: select the other cabinet or change the mic's position.

Mic Position

When the mouse is located over a mic stand, the mouse pointer changes to an “updown” arrow and the mic gets a copper/goldish glow. Click-and-drag the mouse up or down to change the position of the mic stand. The microphone moves along a predetermined path, so you only need to move the mouse up or down, not to the left or right.

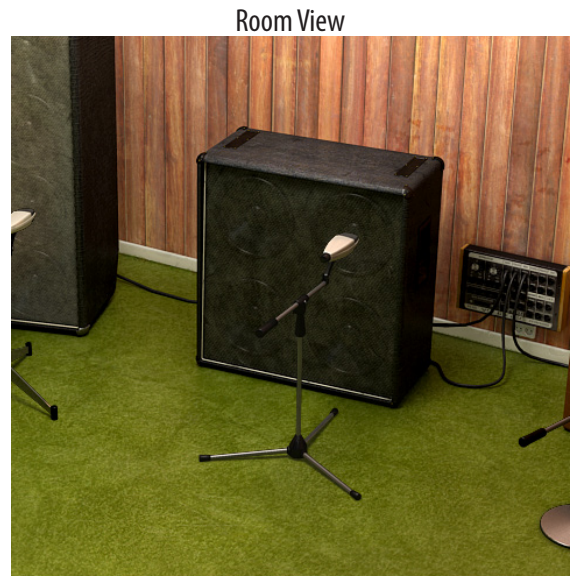
Cabinet Selection

You can change the cabinet by clicking on the background and dragging the mouse to the left or right. The mouse pointer becomes a “left-right” arrow when you are pointing at the background to indicate that it is possible to change cabinet.

Clicking on the background while holding the Shift-key toggles through the cabs without any sliding animations.

Mix Panel (Bottom Area)

In the mix panel, you can change the balance between the amplifier signal and the D.I. signal. The tonality of the D.I. signal can be changed with the three EQ knobs and the compressor/limiter. You can also change the phase of the amplifier signal and adjust the output volume.





1. High/Low
2. Normal/Volume
3. Bass
4. Middle
5. Treble
6. Master volume

The Amplifier

The amplifier that we chose to model is actually a guitar amplifier, but because of its very characteristic “non-fizzy” distortion and special-sounding tone stack, it’s been very popular among bass players during the last decades. When the **High/Low** switch is set to **HIGH**, the amplifier will work and act just like the real thing. The **LOW** mode is the same amplifier with a slightly different input stage, one which lowers the input gain a little bit while adding some warmth to the signal. The **LOW** mode gives you a fat bass sound without too much distortion.

Use the two “volume” controls as a means of dialing in the right amount of distortion, not to change the actual volume. More **Normal Volume** gives you greater preamp distortion, and more **Master Volume** gives you greater power amp distortion.

High/Low Use this switch to select either the attenuated channel (**LOW**) or the regular channel (**HIGH**).

Normal Volume This knob controls the amount of preamp distortion, and is usually called “Gain” on more modern amps. Use it to dial in the amount of distortion you want.

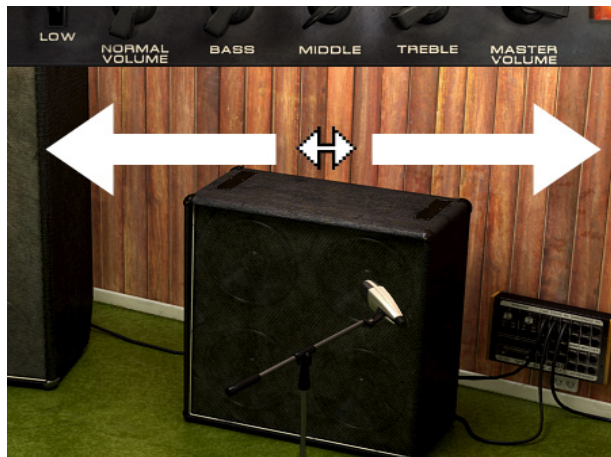
Bass, Middle, Treble These knobs are the tone controls of the amplifier. A common setting is **Bass** on full, **Middle** on min, and **Treble** somewhere in between.

Master Volume Adjusts the amount of power amp distortion. With a massive amount of **Master** and **Normal Volume**, the amp will start sound like it’s about to break. Unless you really want that kind of distortion you’ll find that a **Master Volume** at 12 o’clock will suit most of your needs.

Master Volume won’t distort much unless you increase the **Normal Volume**. Just like the real amp.

The Cabinets

Click and drag left/right on the background to change cabinets.



8x10

The industry standard 8x10” doesn’t need much presentation. With a fat sound and lots of low end, this is the goto cabinet for many bass players.

4x12

The regular 4x12” cabinet is preferred by many bass players over the industry standard 8x10”. It has a more focused sound than the 8x10”, and although it’s a bit thinner than the 8x10”, it is often easier to work with in a mix.

1x12

This is the odd bird among the cabinets. While trying out cabinets for Bass Amp Room, we felt that we needed to listen to something completely different just to clear our minds. We found this old openback 1x12”, plugged it in, and immediately fell in love with the sound. Some sort of ‘60s sound with lots of room. And you can dial in more low end if you need it just by moving the mic more to the off axis position.

Positioning the Mic

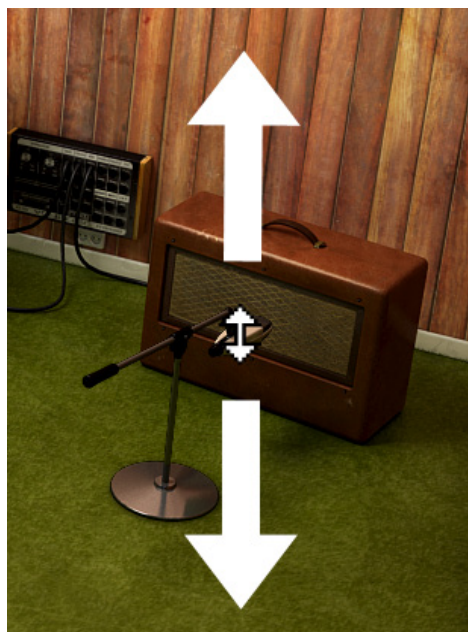
By changing the mic position, you can get lots of different sounds from a single cabinet. How the sound changes depends on the cabinet and the room, but there are some common features for all the cabs in Bass Amp Room:

CLOSE OFF AXIS: The position with most bass and the least amount of highs.

CLOSE ON AXIS: Moving the mic towards the onaxis position will gradually give you more highs and a more pronounced mid. Usually the most “focused” sound.

FAR FIELD: Moving away from the cabinet will give you more room ambiance and often (but not always) a thinner sound.

If you find that the selection of cabinets isn’t enough for you, you can always bypass the cabinet in Bass Amp Room and use the cabinets from one of our other Amp Room plugins (with the amp modeling bypassed).





1. Low Cut
2. Tone
3. High Cut
4. Limiter
5. DI/Amp balance
6. Output volume
7. Solo DI
8. Phase invert (Amp)
9. Solo Amp

Direct Inject (D.I.) Knobs

Low Cut A 12dB/octave low cut filter. This control is intended for filtering out *unwanted* frequencies, but can, of course, be used more creatively. Keep increasing the knob until you cannot hear any difference in the sound. When you start to lose the low end, stop and go back a couple of millimeters (yes, we are metric).

High Cut Same as the Low Cut, but for high frequencies.

Tone When this knob is set at 12 o'clock (**FLAT**), it doesn't change the sound at all. Turning it clockwise will give you a fatter and more scooped sound much like the tone stack in the amplifier. Turning it counter-clockwise will give you a sound with lots of mid.

The Mix Panel

SO, WHAT ELSE DO YOU NEED EXCEPT a good sounding amp? Since a lot of engineers and producers prefer to work with both the raw D.I. signal and the mic'ed signal (using the amp signal for character and the D.I. signal for focus or low end), we decided to incorporate that work flow into Bass Amp Room. But instead of having two tracks (one with the microphone signal and one with the direct bass line), you can easily blend the two signals directly in Bass Amp Room. And, to make life easier, we added a couple of very powerful tone controls and a stateoftheart singleknob compressor in the D.I. section. That way you will be able to get a really good bass sound just by using the D.I!

Changing the **Low Cut** and **High Cut** will drastically change the characteristics of the **Tone** knob. Technically speaking, the center frequencies of the **Tone** knob filters will change when you change the bandwidth (the settings of **Low** and **High Cut** combined).

Limiters Use the limiter to increase the amount of compression. This function can, for instance, be used to limit the peaks of the D.I. signal, and would then be set somewhere around 12 o'clock, depending on the level of the input signal. With a higher setting you will get a longer sustain on each note.

Limiters LED The LED on the top right of the **Limiters** knob indicates when the limiter is working (like a gain reduction meter).

Mix Knobs

D.I./Amp Balance This fader sets the balance between the D.I. signal and the amp signal. Use the **Solo** knobs to audition each channel (post fader).

Solo Click the Solo texts to solo either the Direct Inject channel or the Amplifier (and cabinet) channel. You can only solo one channel at a time, so by soloing one channel you automatically un-solo the other channel.

Phase Invert (Amp) Inverts the phase of the amp signal. Changing the relative phase between the D.I. signal and amp signal will often completely change the sound. If you have a hard time trying to find a good blend between the two signals, try flipping the **Phase Invert**. It usually makes a big difference to the sound.

Phase invert! If you have the balance fader somewhere in the middle, you should make it a habit to toggle the **Phase Invert** switch. It can make a huge difference to the sound.

Output Sets the output volume. It goes from 32 dB to +32 dB, except at the minimum position where it turns off the output signal completely.

Output Peak LED The Output Peak LED indicates when the output signal peaks at 0 dBFS. Bass Amp Room will not clip the peaks, it is only an indication.

Bypassing Amps or Cabs

You can choose to bypass the amp or the cabinet by selecting **AMP BYPASS** or **CAB BYPASS** from the small box in the lower right corner.

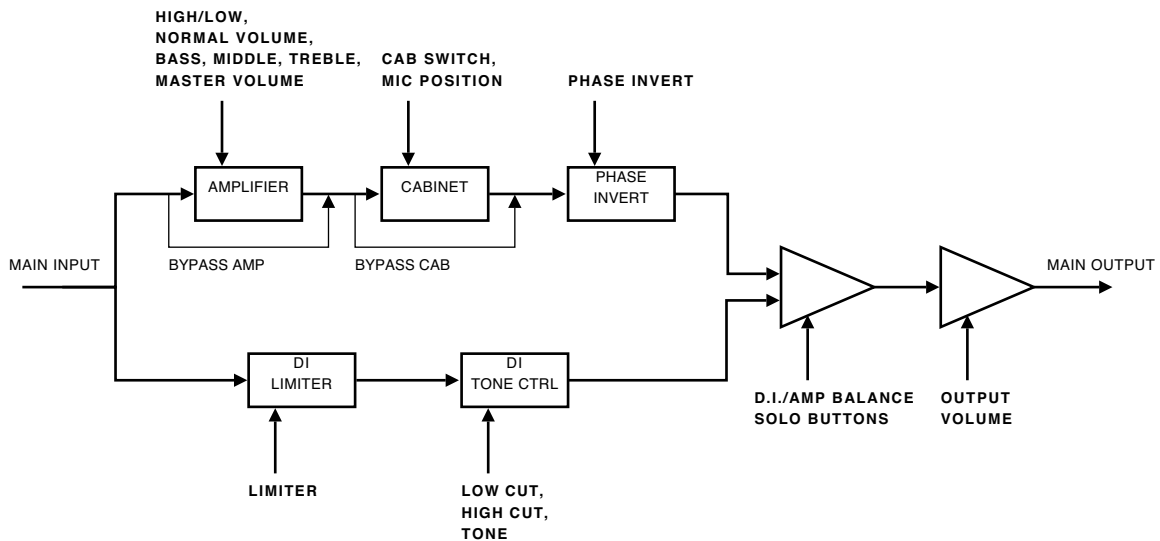
This is very useful if you want to use Bass Amp Room with an external speaker (or speaker plugin) or if you just want to add a cabinet simulation to a track, such as a recorded line out from your amplifier.

This will not bypass the D.I. section, since that can be easily done by setting the **D.I./Amplifier Fader** all the way to the right (on **AMPLIFIER**).

By bypassing the cabinets in Bass Amp Room, you can use the cabinets from the other Amp Room plugins. Just insert, for example, Metal Amp Room (with bypassed amplifier) on the track after Bass Amp Room (with a bypassed cabinet).

Block Diagram

The bass signal (Main Input) is split up into two identical signals, one that goes to the amp (top section) and one to the D.I. (bottom section). The two signals are mixed with the D.I./Amp Balance fader.



Buying Recommendations

If you like the sound from Bass Amp Room and would like to get that sound using the real deal (let's face it, a real amp is always sexier than a plug-in), here are some buying recommendations.

Amplifier

If you like this characteristic fat distortion and scooped-out tone stack, I would suggest that you get a two-channel Hiwatt DR103 from the late '70s/early '80s.

Cabinets and Mic

The leftmost cabinet was modeled after the industry standard of bass cabinets: the Ampeg 8x10" SVT810. If you like the 4x12 sound, I'd suggest that you try out a couple of different 4x12s, because we have absolutely no clue what kind of 4x12 we measured. It sounded excellent, had no labels on, and wasn't for sale... The small cabinet was a half-open 1x12" cabinet with a Celestion 20W/15ohm driver.

But you will also need a good sounding mic to get a good bass sound, and in our opinion nothing beats the Sennheiser MD421!

D.I. and Mix Panel

The tone controls of the D.I. and limiter was made by us and have no hardware counterpart.

Room

If you like the sound of the room, make sure to schedule a session at Care Of Sound in Norrköping.

BRAND NAMES ARE TRADEMARKS OR REGISTERED TRADEMARKS OF THEIR RESPECTIVE OWNERS.

Credits

Niklas Odelholm – modeling and sound design. **Oscar Öberg** – compressor modeling, framework programming. **Torsten Gatu** – interface programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering.



9 Console 1

Softube Console 1 is a new type of audio mixer. It is a hardware used together with your favorite digital audio workstation (DAW), and just like a traditional mixing console, you can route any or all DAW tracks to Console 1 and mix them from within that system. This gives you access to Console 1's world class sounding effects sections and lets you control everything from the hardware unit, using physical knobs and buttons.

There is a separate manual for Console 1. You'll find it by clicking on the Console 1 menu item, or on www.softube.com.

So what Console 1 gives you is the great sound that has made Softube world famous, in addition to a tactile and intuitive workflow that speeds up your mix work substantially compared to mouse mixing on the computer.

The SSL SL 4000 E Channel

Included in your purchase is Softube's model of a Solid State Logic SL 4000 E, one of the most famous and coveted mixing consoles in the world. SL 4000 E consoles have been installed in hundreds of the world's leading studios, and it is said that Solid State Logic's 4000 series have been used on more platinum selling records than all other console models combined! This is for good reason. The SL 4000 E has a signature sound that is transparent enough to work for all music styles, but still adds its touch of luxury. Things simply sound like a record when they've been run through the SL 4000 E.

With Console 1, you get the famous black knob E242 equalizer, the classic channel compressor, the very musical gate/expander and the sweet high/low cut filters from the SL 4000 E channel. We also modeled the unique harmonics, distortion and non-linearities you get from running the SL 4000 E into overdrive. Controlled by Console 1's Drive knob, you can use it subtly on all channels to glue the mix together like a slightly overdriven SL 4000 E does, or not so subtly to add edge and presence to selected sounds.

In addition, you get Softube's unique Transient Shaper, which is not modeled off the Solid State Logic SL 4000 E or any other unit for that matter.

Softube's model of the Solid State Logic SL 4000 E channel is only available as part of the Console 1 system and not as a single plug-in. Softube will continuously release more channel strip models to be used with Console 1, that can be purchased separately.

More Information

For more information, please visit www.softube.com or the Console 1 specific manual.



10 Drawmer 1973 Multi-Band Compressor

MORE CONTROL, MORE VERSATILITY, more definition—the **Drawmer 1973 Multi-Band Compressor** plug-in lets you micro-manage the dynamics of your sound with ultimate precision. Make your mixes wider and louder, shoehorn a piano into a dense mix, or curb the temper of an unruly bass guitar. And do it all with the unmistakable analog feel of a modern Drawmer classic.

Carefully modelled on the original Drawmer 1973 hardware in collaboration with the Drawmer research and development team, and approved by **Ivor Drawmer** himself, the Drawmer 1973 brings the award-winning three-band compressor bang up to date.

Mastering with the Drawmer 1973 gives you independent control over three frequency bands, often a miracle-cure for problem mixes. Pull out individual instruments,

brighten the mix, or tighten the lows ... or of course do it all! But the Drawmer 1973 is equally useful on single sound sources in a mix. Take away the boomy character of an acoustic guitar in a perfectly natural sounding manner. Control the low frequency dynamics of a bass but leave its top end alive and kicking. Or remove sharp consonant sounds from a vocal track without eliminating detail and clarity.

Staying faithful to the look and sound of the original Drawmer 1973, we've added some extra up-to-date features to the plug-in, including **Mid/Side** processing and sidechain, giving even more versatile compression and dynamic control. So the Drawmer 1973 plug-in lets you play with the mid portion of a sound without affecting the rest of that frequency, or link the gain reduction to an external sound source.



Low Band

Mid Band

High Band

User Interface

The Drawmer 1973 panel consists of four different sections, an Input/Output section that handles input and output volumes and signal routing. The lower section consists of three different frequency bands with similar, but not identical functionality. Here follows an overview of all controls, from left to right.

Input/Output Section

Input Gain Controls the gain going into the unit.

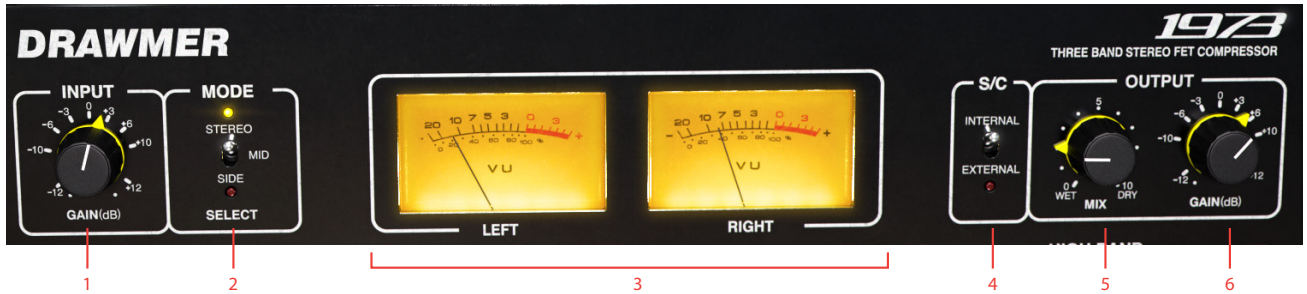
Mode Sets the 1973 in either **STEREO** or **MID/SIDE** modes—see a further explanation on using **MID/SIDE** below. The unit is in **MID/SIDE**

mode in both **MID** and **SIDE** positions. Selecting **MID** displays the controls of the **MID** portion of the sound, selecting **SIDE** displays the controls of the **SIDE** portion of the sound.

Please note that when you switch from **STEREO** mode to **SIDE** mode, the (yellow) knob settings from **STEREO** mode will get copied to the (red) **SIDE** mode. Therefore, all changes made in **SIDE** mode will get overwritten if you switch to **STEREO** from **SIDE** mode.

If you have made adjustments with the **Mid** or **Side** setting selected and then switch back to **Stereo**, the settings made in the **Side** mode will be lost.

Input/Output Section



1. Input gain
2. Mode Stereo/Mid/Side
3. VU Meter
4. S/C Internal/External
5. Mix
6. Output gain

VU Meter shows the output level for the left and right channels.

S/C Sets the sidechain mode. When set to **INTERNAL**, the 1973 operates in normal mode which means that the compression circuits respond to the incoming signal. If set to **EXTERNAL**, the 1973 will use an external sound source in the detector circuit. For example, external sidechaining can be used to get an entire mix to “duck” with every hit of a kick drum.

If **Sidechain** set to External without external signal routed to the 1973, it will operate as in Internal mode.

Please note that when in Mid/Side mode, sidechain signal will be split to Mid/Side as well. Mid part will trigger the Mid channel of the 1973, and Side part will trigger compression of the Side channel.

Mix Controls the relative level between dry signal (unprocessed) and wet signal (processed by the 1973's circuits). **Mix** is useful for easily setting up parallel compression, where a certain amount of the dry, unprocessed sound is mixed into the compressed sound. This achieves a sound where you get the dynamic control and coherence from the processed sound, but still keep a dynamic feel from the original sound source.

Output Gain Controls the output gain from the unit.



- | | |
|---------------------|----------------|
| 1. Big | 6. Air |
| 2. Low X-Over freq | 7. Attack |
| 3. Threshold | 8. Release |
| 4. Status | 9. Output gain |
| 5. High X-Over Freq | |

the frequency point at which the split between LOW and MID bands occurs. **High X-Over Frequency** (1 kHz - 14 kHz) sets the frequency point at which the split between MID and HIGH bands occurs.

Frequency Bands

Between the frequency bands sits the knobs that control the crossover frequency of the bands.

Low X-Over Freq and

High X-Over Freq Selects the crossover frequencies for the 6 dB per octave crossover filters. These are the frequency points where one band stops to process audio, and another band takes over. **Low X-Over Frequency** (50 Hz - 1.3 kHz) sets

Input and Output Gain, Mix and X-Over Freq knobs are global and affect both Stereo and Mid/Side modes.

Threshold Determines the signal level above which compression starts to take place. A low **Threshold** setting means more of the incoming signal gets processed. The 1973 is a soft knee compressor, which means that more compression is applied gradually as the input signal level increases.

Attack Controls how quickly the compressor responds to signals that exceed the threshold level. All times are nominal, the actual attack time is further modified by the release setting.

Release Adjusts how quickly the compressor ceases to process the sound after the input level has fallen below the threshold level. The first three switch positions are fixed and provide progressively increasing release times, while **PGM DEP** (“program dependent”) positions **F**, **M** and **S** (“fast”, “mid” and “slow”) cause the release times to vary in a manner which automatically adapts to the dynamics of the incoming signal.

Gain Turning this up compensates for the gain reduction which occurs when the compressor is active. As the Drawmer 1973 is a multi-band compressor, the three gain controls can be used to achieve subtle or radical changes to the overall equalization curve.

Hold Shift on your computer keyboard while adjusting a setting to link the corresponding control in the separate bands.

Gain Reduction

meters The meters operate in -1, -2, -3, -5, -7, -10, -15, -20 dB range.

Status Activates (**ON**), bypasses (**BYPASS**) or mutes (**MUTE**) the individual bands.

All the parameters above are common for all frequency bands. The controls below are specific for the low and high band respectively.

Big The Big circuit makes the low frequencies appear louder or “bigger”, by reducing the sidechain’s sensitivity to low frequencies. The **BIG** mode is useful on a stereo buss or in a mastering situation, where you want to control the dynamics but still keep a thick and warm low end tone. It is only available for the **LOW** band.

Air Air can be used to reintroduce high frequencies, which can sometimes be lost as a result of heavy compression. Activating **Air** makes the sound more intimate, detailed and transparent, but without making it sound harsh or introducing any noticeably unnatural artifacts. **Air** is only available for the **High** band.

In Use

The Drawmer 1973 Multi-Band Compressor splits the frequency spectrum into three bands—Low, Mid and High—and lets you add compression to these individually. This is useful in a number of cases. Here are some examples.

Controlling a Boomy Guitar or Piano

Sometimes, you will encounter a recording of an instrument where some notes seem to stand out in an annoying manner. Perhaps there is an acoustic resonance in the recording room, or in the body of a piano or an acoustic guitar, which is excited when certain notes are played. So every time a particular range of notes are played, the resonance just seems to explode. A similar situation often occurs with rhythm guitars in heavy metal, when the player goes back and forth between letting power chords ring out and “chug-chug” style palm muted playing. The palm muted sections will often have an overbearing bass boost, which you might want to control in the mix. In cases like these, simply cutting the offending frequency with an equalizer isn't the best option, as this affects that frequency at all times.

But with the Drawmer 1973, you can use one of the bands to zoom in on the disturbing resonant frequency and compress it whenever it gets too loud—and only then. First, decide if the **LOW**, **MID** or **HIGH** band is the suitable choice for your application. Set the other two bands in **MUTE** mode (with the **Status** switch), and adjust the crossover filter(s) until you have found the specific frequency range you want to control. Set the **Threshold** low enough for it to compress every time the frequency stands out too much, but high enough to leave the frequency unaffected at all other times. When you obtain a nice and even response, engage the two other bands again. Either set the 1973 to **BYPASS** them, or adjust their parameters as you see fit.

Create a Solid Bass Track

For very dynamic bass guitar playing styles, such as a funky slap style or a hard rocking picked bass, you may find yourself wanting to keep the low end of the bass tight, even and controlled without strangling the life and attack that you'll mostly find in the midrange and treble with playing styles like these. Here, you can use the Drawmer 1973's **LOW** band to do precisely that—smooth out the dynamics of the low frequencies to create a solid ground for the mix, but use lighter compressor settings for the **MID** and **HIGH** bands, or bypass them entirely.

Compress a Subgroup or the Master Bus in a Transparent Manner

With a full-band compressor, you will find that every time a part of the sound source exceeds the threshold level, the entire output gain will be reduced. This may be just the effect you are after, but in other cases, Drawmer 1973 can be used to create a transparent dynamics control. This is because none of the three bands will react to what happens in any of the others. So for example, a loud bass drum hit will not make the cymbals duck, as they would with a full-band compressor. This means you can control the dynamics, but in a less apparent manner.

Adjust Specific Sound Sources in a Stereo Mix

The Drawmer 1973 plug-in offers a mode which is not available on the hardware—the **MID/SIDE** mode (**MS**). In **MID/SIDE** mode, the Drawmer 1973 will not process the left and right channels separately. Instead, the sound is separated into the **MID** and **SIDE** portions of the sound. **MID** contains only the material which is common to both left and right channels (ie. the mono portion of the sound). This should not be confused with the **MID** frequency band. **SIDE** contains only the material which differs from left and right.

MID/SIDE processing is typically useful in a mastering situation, where you are working on stereo program material without access to the individual tracks. Perhaps you will find that a particular sound source—such as the lead vocal—has been mixed too low, and you want to bring it forward. But since you don't have access to the individual tracks, you can't just raise the lead vocal's level. You could apply EQ to bring up the frequencies that are most dominant to the vocal, but you would then also bring up all other instruments in that frequency range.

In this case, you could use the Drawmer 1973 in MID/SIDE mode. The vocal will typically be mixed dead center, so it will only be present in the MID channel. Therefore, you can select the MID channel with the **Mode** switch, and use its crossover filters to have one of the 1973's bands zoom in on the frequencies where the vocal is prominent. Boost the output of this band, and you will find that you have brought the vocal forward without affecting the chord instruments such as guitars and keyboards by much, even though these exist in the same frequency range as the vocal. Because these will typically be either panned left or right in the stereo panorama, or stereo recorded. That means they mostly reside in the SIDE channel and are thus not affected by the changes you made to the MID channel, at least not by as much as the lead vocal was.

Widen the Mix

MID/SIDE processing can also be used to make a stereo sound appear wider. You do this by slightly turning down the MID channel relative to the SIDE channel. But be careful not to go too far with this, as you might end up with a center “hole” in the mix. You may find that turning down the HIGH band of the MID channel, but leaving the LOW and MID bands as they are, gives a widening effect but still retains the integrity of the mix.

External Sidechaining

The typical example for sidechaining is to get an entire mix to “duck” every time a kick drum hits—this is a very common technique in electronic dance music. In order to achieve this, you set the 1973's S/C switch to EXTERNAL, and select which sound source should be routed into the sidechain of the 1973 (please refer to your DAW's manual for the specifics on how to do this). With this in place, the 1973 will start reacting to the sound of the external sound source that is fed into the sidechain. So in the above example, this sound source would be a kick drum. Every time the kick drum level exceeds the threshold level, the 1973 will react and start compressing.



Original Drawmer 1973 hardware.

External Sidechain in Mid/Side Mode

If you set the 1973 to MID/SIDE mode and use external sidechaining, the sidechain is actually also set to MID/SIDE operation. This means that you can use the external sound source to control only the MID or SIDE portions of the sound. For example, you might find that the kick drum and bass instrument struggle for the same space in a mix. In this case, you could feed the kick drum into the sidechain, use the Mid channel to zoom in on the low frequencies of the bass instrument (which is typically mixed dead center), and set the **Threshold** so the bass instrument ducks every time the kick drum hits. You will now have the bass instrument “wrapping around” the kick drum instead of conflicting. But everything else in the mix will be unaffected.

Parallel Compression

In any of the above use cases, you can use the **Mix** knob to apply parallel compression. This is a very useful technique in which you create a blend of the processed and unprocessed sound. The processed portion will add a full and dense sound, while the unprocessed portion retains the original dynamics.

Keyboard Shortcuts

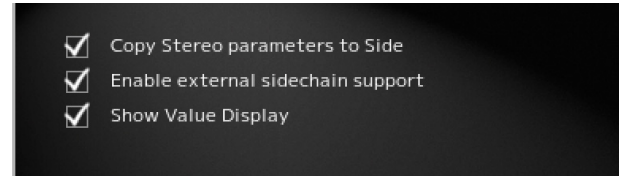
SHIFT-click a parameter to adjust it for all bands at the same time. If you SHIFT-click the “status” label, that band will be soloed, ie. the other bands will be muted.



The fastest way to un-solo (un-mute the other two bands) is to shift-click **On**, which will turn on all three bands.

Setup Options

The normal operation for the Drawmer 1973 is to copy



the Stereo parameters over to the Side parameters, whenever you go from Stereo to Mid/Side. It is possible to turn off this option by disable “COPY STEREO PARAMETERS TO SIDE” in the Setup menu.

Credits

Björn Rödseth, **Oscar Öberg** – modeling. **Paul Shyrinskykh** – product owner, manual. **Niklas Odellholm** – graphic design. **Kristofer Ulfves** – marketing. **Patrik Holmström** – framework programming, **Henrik Andersson Vogel** – additional manual copy. **Ulf Ekelöf** – 3D rendering.



11 Drawmer S73

FOR THE MASTERING ENGINEER, a multi-band compressor is a powerful tool. But as Spiderman's uncle Ben famously says: with great power comes great responsibility. A multi-band compressor that is not used diligently may create more problems than it solves. This is why we created Drawmer S73 Intelligent Master Processor.

Drawmer S73 Intelligent Master Processor features an enhanced multi-band compressor sound design, fine-tuned to improve your mix with a minimum of tweaking. The S73 does most of the job for you by making the choices a mastering engineer would make. Simply use the **Style** parameter to switch between ready made mastering processing techniques to quickly find the sound that suits your mix the best.



Original Drawmer 1973 hardware.

Handles It All

When we designed Drawmer S73 IMP, we wanted to make it sound transparent and suit any program material. We analyzed dozens of mixes and reached the conclusion that most of them have similar problems. So we created algorithms for S73 IMP to fix those problems. S73 can deliver both gentle sheen for a mix that's already great to begin with, as well as dramatic improvements for a more problematic mix.

Based on Drawmer 1973

Under the hood, the S73 is based on technology created by Drawmer for the **1973 Three Band FET Stereo Compressor**. This is an award winning multi-band analog compressor design famous for its precision and flexibility. Softube has carefully modeled all the characteristics of the original analog hardware, and have taken the concept a step further by adding mastering sound design to create the Drawmer S73 IMP. The unit offers the perfect way to balance and shape your mix, and get great analog sound in no time.



1. VU meters
2. Air
3. Style
4. Amount
5. Gain reduction meter
6. Mix
7. Gain

User Interface

Amount Controls the amount of dynamic and spatial processing applied. However, you should use **Mix** knob to control the overall precessing level.

Gain reduction meter Shows the overall gain reduction

Style

Switches between the different styles of mastering algorithms. The algorithms are divided into three groups: **CLEAN**, **BASS CONTROL** and **SPATIAL**.

CLEAN group contains styles that make a mix clearer by improving definition between vocals, instruments and other elements of the mix. Scroll through the styles in the group to find one that suits your music the best.

BASS CONTROL group lists styles that treat problems in the low end. If your mix sounds boomy or muddy, try De-Rumble style first. Punch style will add punch and definition to the kick drum and bass.

SPATIAL group includes styles designed to widen stereo image of your mixes, bring out ambience and emphasize lead vocals in the mix.

VU meters VU meters show output volume for left and right channel.

Mix Mix knob controls the balance between **DRY** and **WET** signal and can be used for parallel processing.

Gain Gain knob controls the output gain.

Air **Air** switch will bring more “air” to your mixes. Use **MIX** knob to control the amount of **Air** you’re adding to the mix.

In use

Using the Drawmer S73 is easy, since you really can’t do anything wrong. Play around with different styles, place several instances after each other, and listen to the result.

Select style

Fundamentally, the Drawmer S73 IMP is operated by auditioning the different **Styles** and choose the one that suits your mix the best. It’s good to start with 1-3 dB of gain reduction and increase the amount if needed. It’s safe to combine two or more instances of S73 with different styles. Just add them in series in the insert slots of your DAW’s master channel. For example, an S73 instance with **GENTLE** compression often works well followed by one set to **WIDE** mix. As does **DE-RUMBLE** followed by **CLARITY 1**. See the **Styles** description section for a walkthrough of all included **Styles**.

Use parallel processing

The Mix Knob can be used for parallel processing. This is a very useful technique in which you create a blend of the processed and unprocessed sound. The processed portion will add a full and dense sound, while the unprocessed portion retains original dynamics. To use parallel processing, try increasing **Amount** until meter shows 5-10 dB of gain reduction—which is in most cases is considered a great deal of compression—and then use the **Mix** knob to blend in the unprocessed signal until you find a pleasing balance. Activating **AIR** will add even more excitement to the master.

Tip! Use several S73 instances with different **Styles** at the same time, for example **De-Rumble** followed by **Clarity 1**.

Styles description

Here is a short description of the different mastering styles.

Clean

Clarity 1 Instant clarity and definition. Use gently.

Clarity 2 Adds definition in the midrange and tames high frequency harshness.

Neutral The most neutral Style, ideal to polish mixes that are already great, and just add some sheen.

Gentle Comp Adds glue and density to the mix. Keep gain reduction around 3 dB.

More Air Lets your mix breathe, and adds sheen to the top end.

This style allows you to control the amount of air by using the **Amount** knob, while the **Air** switch will add additional sheen.

Bass Control

De-Rumble Removes low frequency rumble from the mix and adds clarity and definition.

Punch Adds punch to a kick drum and bass.

Spatial

Wide Mix Gently widens your mix in the stereo field.

Ambience Brings out existing ambience in your mix.

Vocals Enhance Will bring the lead vocal upfront.

Please note that Spatial style group is not compatible with mono and multi-mono versions of S73

Credits

Björn Rödseth, Oscar Öberg – modeling. **Paul Shyrinskykh** – sound design, manual. **Torsten Gatu** – project management. **Niklas Odelholm** – graphic design. **Kristofer Ulfves** – marketing. **Ulf Ekelöf** – 3D rendering.



12 Eden WT-800 Bass Amplifier

MODERN MUSIC PRODUCTION IS ALL ABOUT THE BASS. Whether it's a Moog synth line, a throbbing EDM rhythm or a poetic fretless bass, you need a solid and clean low end. The Eden WT-800 Bass Amplifier is the ultimate bass recording tool for producers and bass players alike.

Eden's flagship bass amplifier **WT-800** is heralded as one of the all-time great bass amps. Every detail has been fine-tuned for bass. The WT-800 plug-in does not only deliver the powerful and present sound of Eden's high-end bass amp. It also includes models of Eden's top of the line 4x10" and 1x15" cabinets, mic'd with four high-end studio microphones each. The mics were selected and positioned by producer and bass player **Troy Antunes**, known for his work with **Justin Timberlake**, **NSYNC** and **AI**

Green. Troy Antunes has also provided some expertly designed presets. Unlike vintage style bass amps, Eden WT-800 delivers a modern, dynamic, and rock solid bass sound, flexible enough to be used for R'n'B, hip-hop, metal or jazz.

Eden WT-800 extends beyond a bass amp model—it's a fully featured tool for any bass instrument recording. Don't limit yourself to only using it with bass guitars—it adds weight and focus to synth basses as well.

High fidelity for low frequency.

Simple as that.

User Interface

If you are familiar with Softube's Marshall plug-ins, you'll recognize the hidden channel strip panel at the right. Click and drag the amp to the left, or click on the panel edge to reveal the channel strip.

Amplifier Parameters

Gain Adjust input gain to the amplifier. Set it so that the green light is blinking when you're playing.

Enhance The magic of Eden. This is a modern bass sound with the twist of a single knob. It's usually the first thing you adjust.

Bass, Treble If you find a three band parametric EQ a bit much for a bass amp, start with the **Bass** and **Treble**. They work just as you would expect.

The **Bass** is a type of shelving filter at around 100 Hz. **Treble** is a high shelving with center frequency at 4 kHz.

Low Gain/Frequency

Mid Gain/Frequency

High Gain/Frequency A three band parametric equalizer; enhance the mid, tailor the highs or tame the low end.

Compressor One knob compressor with a punchy attack and fat release. This compressor will **add more punch and attack to your tone**, rather than destroy the transients. If you play with a soft touch you'll get a looong sustain with a bit of compressor dialed in. The **Input Gain** will also affect the amount of compression, so a higher input gain will lead to more compression.



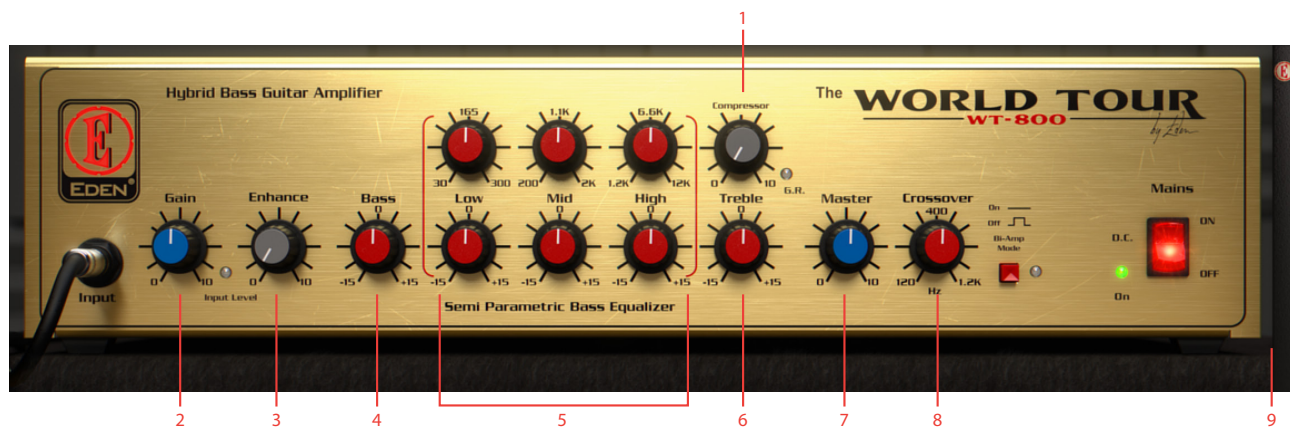
Amplifier



Amplifier

Channel Strip

Click side bar, or click and drag the amp, to reveal the channel strip



1. Compressor
2. Gain
3. Gain
4. Bass
5. Low gain/Frequency
Mid gain/Frequency
High gain/Frequency
6. Treble
7. Master
8. Crossover, Bi-Amp mode
9. Open channel strip

G.R. LED The gain reduction LED lights up when the compressor is limiting the output volume.

Master Main volume. Use this as the primary, overall, output volume knob.

Crossover, Bi-Amp Mode In **Bi-Amp Mode**, the output from the amp is split up into a low frequency and a high frequency part with the crossover frequency set by the **Crossover** knob. The low frequency signal is fed to the 1x15" cabinet, and the high frequency signal to the 4x10". With **Bi-Amp Mode** engaged, you can get a very clear and solid mid-range.



1. Mic selection 1x15
2. Mic selection 4x10
3. Tweeter level 4x10
4. Clip LED
5. Phase invert
6. On/Off
7. Level and channel meters

Channel Strip Parameters

The channel strip consists of three channels: the mic on the bassy 1x15" cabinet, the mic in front of the brighter and more well-balanced 4x10" cabinet as well as audio direct from the amplifier. This is not the signal from one of the original WT-800's built-in direct outputs, but an output taken from the end of the amplifier; after the **Master Volume**.

Mic Selection (only 1x15" and 4x10" channels)

Both the 1x15" and 4x10" cabinet has a selection of four different mics. Select which mic to use with these two columns with four radio

buttons. For more information regarding the microphones, see the section below.

Phase Invert (all channels)

Phase inverts the audio on that channel. Keep in mind, there is no "correct" phase when it comes to bass. Flip the phase and see what happens. If you like the sound, keep it, otherwise flip it again.

In (all channels)

Turn on the audio on that channel.

Level

(all channels) Set the level of the respective channel.

Tweeter Level

(only 4x10" channel) Turn up for more high frequencies from the 4x10" cabinet. The 4x10" has a built-in tweeter, with a level control on the back. This knob adjusts that level.

Meters,

Clip LED The channel meters show the peak levels of each channel. The **Clip LED** at the top right lights up when the output signal exceeds 0 dB.

Cabinets and Microphones

How do I get a clear and uncolored tone with lots of headroom? That was the question that started Eden's quest for the perfect bass cab. They started building bass cabinets in the late 70s and their designs quickly became an industry standard for touring musicians. Since you didn't have to trade low end with clarity anymore, it's no wonder that it became a success.

The cabinets we chose to include in the Eden WT-800 plug-in are the cabinets most associated with that amp: the **D410XLT** and **D115XLT**.

The 4x10"

The **D410XLT** has a 4x10" speaker layout and dual ported design, and offers focused tone with exceptional low end. It is the perfect option when you need to project your sound through the mix and drive for ultimate lows.



Troy Antunes during the Kore Studios sessions.

The 1x15"

The **D115XLT** is a tuned and ported professional cabinet with a fast responding 15" speaker. It offers a classic rounded tone with great detail and response, coupled with a high temperature voice coil and our Eden cast bell tweeter. It's the option for players looking for a little more growl.

Microphones

Put three bass players in a room and there will be heated and long discussion about mics. But thanks to moderating non-bass players we managed to put the differences aside and come up with a roster of five diverse and very different sounding mics.

The **AKG D112** has a classic scooped-out sound that just sounds right. Might feel a bit boring, but it does the job. **Electro Voice's RE20** is another classic, with not as much of the lowest and highest as the D-112, but a focused and warm midrange. **Neumann U 47** just has to be included. Full range, sparkling highs, nice and balanced. Can't go wrong there.

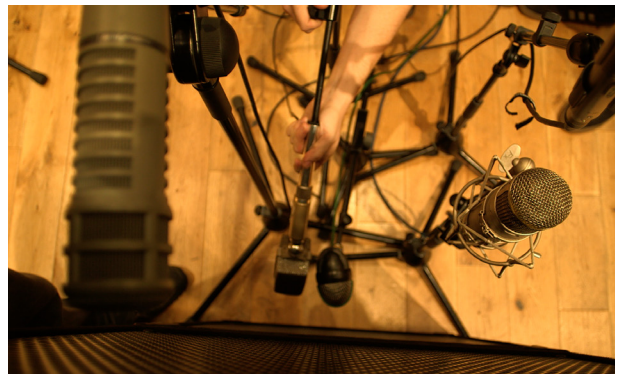
The final two mics are not studio classics for bass, but completes the palette in a very nice way. Royer's ribbon **R-121** has a super smooth top, and tames the 4x10" like nothing else. It's sounds both aggressive and smooth at the same time (a passive/aggressive mic?). And last, but definitely not least: the huge **Melodium Type R** ribbon mic is as big as it sounds. Nothing sounds like that one.

George Apsion, owner of Kore Studios, and lucky owner of that particular mic says: *"The Type R is a secret weapon for bass. It's an old French ribbon mic dating from the early 1940s. We were lucky enough to find one with the original ribbon intact and it has a huge low bump in the frequency response. There can't be many left out there, so it's a great feeling to hear it producing such immense sounds in a modern setting."*

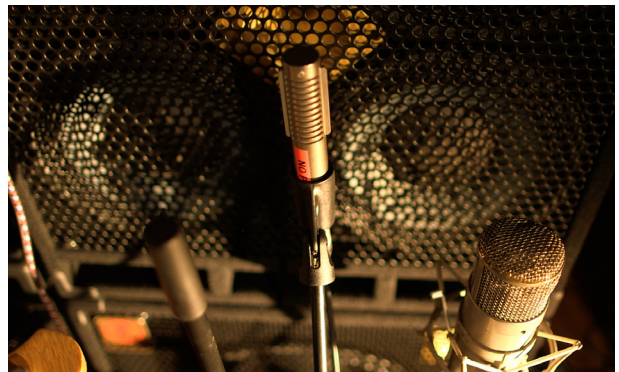
All cabinets and mics were recorded at Kore Studios, London, UK, by Troy Antunes, Niklas Odelholm, Kim Larsson and George Apsion.



Melodium Type R



Setting up mics for the session.



The Royer R-121 and U 47



The real deal.

Meters

The peak meters in the channel strip shows the peak level of each individual loudspeaker or direct from the amp. Even if all three meters are below 0 dB, that won't guarantee that the combined level is below 0 dB, since the sum of all can be higher. If the **Clip LED** at the top right corner of the channel strip is lit, you'll definitely be above 0 dB. Please note that the Eden WT-800 plug-in won't limit the audio level at 0 dB, but can output much louder audio than that. So if you have a DAW that allows levels above 0 dB, you don't have to bother too much about the **Clip LED**.

Mono and Stereo Operation

The Eden WT-800 is in its most basic form a mono plug-in, but if you place a stereo instance of the Eden WT-800 on a track, it will process both channels independently. You can thus use the stereo version of Eden WT-800 as an effects processor on stereo sources, such as a stereo synth bass line or drum kit.

Presets

Many of the presets have a carefully tuned equalizer, sometimes to match a bass player, style, song or tonestack of a vintage amp. When adjusting presets, reach first for the **Bass** and **Treble**, **Compressor**, **Enhance**, cabinets, and lastly the semi-parametric equalizer. **Compressor** settings depends a lot on your input gain and bass, so it will always be necessary to adjust the amount of compression to your taste.

Credits

Kim Larsson – modeling, programming. **Niklas Odelholm** – project management, sound design, graphic design, user manual. **Arvid Rosén** – extremely complicated Sharc related stuff. **Björn Rödseth** – programming. **Patrik Holmström** – programming. **Paul Shyrinskykh** – project management, quality assurance. **Troy Antunes** – sound design, presets. **Jerker Odelholm** – presets. **Ulf "Rockis" Ivarsson** – presets. **Ulf Ekelöf** – 3D modeling.

13 FET Compressor

Introduction

THE FET COMPRESSOR IS BASED ON the most famous hardware FET compressor, and very much effort has been spent modeling the real hardware to make sure that this one sounds exactly the same. If you only use the big knobs and the six fixed ratios you'll get exactly the same analog sound and functionality as the real deal. But that's just the beginning. With the help of digital technology we have added some useful features that still have that analogue sound – but features that would've been hard or even impossible to implement with analog electronics. That way you will be able to get the best of two worlds. (Not even mentioning how ridiculously many FET Compressors you will be able to fit into your project without running low on CPU.)

Not convinced yet? Set the **Input** on **MAX**, **Ratio** on **ALL**. Listen.

Design Philosophy

THERE ARE THREE MAIN IDEAS behind this design: first of all, we wanted to make an analog sounding digital compressor. Nothing must stand in the way of the sound. Every single part of the simulation contributes to making this piece unique. The extremely fast attack, all the subtle (and sometimes not so subtle) distortion that comes from the different parts of the compressor and the extremely careful way the signal is handled and conditioned through out the signal chain.



SECOND, WE ADDED FEATURES to make this product even more versatile and unique. The enhanced ratio, parallel compression, detector filtering and lookahead are such features. Third, these features must not stand in the way of the usability. The fewer knobs the better, and the faster the user gets the sound they want, the better.

This boils down to a single main objective: Every user should be able to get a good sound within seconds, and no user should be afraid to mess up the sound. An amateur should be able to make this sound just as good as a pro. And yeah, it has to look good.

In order to achieve these objectives we had to put special effort into the modeling. The original hardware has some quite quirky circuits, and a lot of its sound comes from far from ideal components and design. A lot of new ideas were developed during the modeling, and with the help of our expert listener and “golden ears” Dan Lumbye, we managed to put together a piece of software that should please even the most critical ears.



1. Input volume
2. Input VU
3. Gain reduction VU
4. Parallel inject
5. Output volume
6. Output peak meter
7. Ratio
8. Attack time
9. Release time
10. External sidechain volume
11. Low cut
12. High cut
13. Lookahead

User Interface

THE CONTROLS OF THE FET COMPRESSOR are divided into two sections.

First of all we have the big knobs (**Input**, **Ratio**, **Attack**, **Release** and **Output**), which are the knobs that you will use every time you use the compressor. These knobs are pretty standard and you probably already know what they do.

Then we have the smaller knobs (**External Sidechain**, **Low Cut**, **High Cut**, **Lookahead**, and **Parallel Inject**), which don't have to be tweaked every time you use the compressor. You don't even have to feel bad if you never touch them. People have managed to make hit records for 40 years without them. Nevertheless, they are extremely useful, and soon you'll have a hard time understanding how you ever could've managed without them.

In addition to this, we also added some metering so that you can monitor your signal all the time. An input VU meter (which is fun to drive into the reds all the time) and a stereo output peak meter make sure that you always know what is going on with your signal.

Input

The **Input** knob on this compressor works both as a gain control and a threshold control. If you increase the **Input** volume you will also increase the gain reduction, which makes it work sort of like a combination of a gain knob and a reversed threshold knob. You can drive the **INPUT VU** far into the reds without any unwanted digital clipping. Just make sure that the **OUTPUT PEAK** meter doesn't indicate any clipping. (If it does, just lower the **Output** volume a bit.)

The **Input** knob also works as a threshold control. More input gain will give you more gain reduction.

If you drive the input volume so that the **VU** start hitting the reds you will also add more distortion to the signal. If you want lots of distortion we recommend you to also set the **Ratio** knob on **ALL**.

More input gain also means more distortion (in a good way).

Ratio

At a first glance, the **Ratio** knob on the FET Compressor seems to work a little bit different than its hardware counterpart. The biggest difference is that the **Ratio** knob is continuous, meaning that it's possible to select settings between different fixed ratios, for instance between 12:1 and 20:1. Furthermore, it's also possible to select settings between 20:1 and the famous "ALL BUTTONS IN" setting.

If you would rather use the "predefined" settings that are identical to those on the original hardware: 1:1, 4:1, 8:1, 12:1, 20:1 and **ALL**, just click on the labels to jump directly to those settings.



"All Buttons In" Mode

The **ALL** mode is a peculiarity found on this type of compressor. Originally the different ratios were selected with push buttons, which made some mix engineers try out what happens if you press all buttons at once. From a technical perspective, pushing all buttons at the same time makes all bias levels go berserk. From a sound perspective, this means that you will get a very pumping compression with lots of distortion. This mode is often called the "British mode". By setting the **Ratio** somewhere between 20:1 and **ALL** you'll get a sound that's definitely unique for this product.

Attack and Release

Use the **Attack** and **Release** controls to change how fast the compressor starts to compress (**Attack**) and how fast it should recover from the gain reduction (**Release**). The attack time on this unit is very fast, ranging from about 20 μ s at the fastest setting to about 800 μ s (that's *micro*-seconds!) on its slowest setting. Other kinds of compressors often have much slower attack times. In comparison to the attack time the release times are much slower – ranging between 50 ms and 1.1 s. Please note that these numbers only give a general idea of the attack and release times. In practice (and just like in the modeled hardware), the attack and release times will be program dependent, ie. depend on the characteristics of the input signal.

Since a fast attack and release time leads to a fast gain reduction, sometimes those settings will cause unwanted "clipping effects". To remove this unwanted side effect, you can either slow down the release time or add some look ahead to the detector circuit. (See "Lookahead").

Input and Gain Reduction VU meters

The **Input VU** meter is used to monitor the input signal. If a stereo signal is present, the meter will display the maximum energy of both signals. This may seem like

a drawback, but since the gain reduction circuit (a.k.a. “detector” or “side chain”) works in mono, the VU will actually display the same thing the detector sees. (Unless you start fiddling with the small knobs, but we’re not there yet.)

The **Gain Reduction** meter is used to monitor the amount of gain reduction in the compressor. The FET Compressor has only one detector, so if a stereo signal is present the gain reduction will be the same for both channels. This prevents unwanted distortion of the stereo image.

If the **Ratio** knob is set at **ALL**, the **Gain Reduction** meter won’t show the same results as the original hardware. Instead it will display the actual gain reduction (which the original unit doesn’t). You will also notice that the **Gain Reduction** sometimes display positive values when the **Ratio** is set at **ALL**, which is correct. The **ALL** mode does indeed have negative gain reduction (=positive gain) for some settings.

Output

In comparison to the **Input** knob, the **Output** knob doesn’t do much more than just adjusting the output volume. Keep a close look on the **OUTPUT PEAK** meter while adjusting the output volume to avoid unwanted digital clipping. It is good practice to keep the levels below 0 dB, even if you are using a (native) host that allows level above 0 dB.

Output Peak Meter

The output peak meter is a fast peak meter with hold values. Single peak values up in the reds (at 0 dB) are okay, but don’t push it. This meter will display the left and right channel on the left and right meter if you use the plugin in **STEREO** mode, otherwise it’ll just show the same **MONO** signal on both meters.

Detector Controls

The most important part of any compressor is the detector. It is the detector that decides how the compressor should compress. All controls except the **Output** (and **Parallel Inject**) control the detector, but in this plugin we have chosen to bundle the four advanced knobs together and call them “detector controls”.

Common to all these controls is that they can be turned off by setting them in the **OFF** position. Their status LED will light up when ever they are active.

Don’t be afraid to keep these controls in the **OFF** position. Use these controls if there is something particular that you’d like to achieve or something that needs to be fixed. (If it ain’t broken, don’t fix it!)

External Side Chain

In some plugin formats, such as RTAS, VST3 and AU, it is possible to use an external signal as input to the detector. This is very common if you for instance want to compress a bass line using the bass drum as the trigger to the compressor. In that case, the bass line will be compressed when ever the bass drum is hit.

Even if an external side chain signal is present, you will have to activate the external side chain by increasing the **External Side Chain** knob until the LED lights up. If the **External Side Chain** knob is in its **OFF** position, the detector will always use the regular input.

Use the **External Side Chain** knob to set the input volume of the external signal (for instance the bass drum track). A higher input volume will lead to more gain reduction, just like the **Input** knob works without an external side chain.

Monitor the **Gain Reduction VU Meter** when you set the level of the external side chain. Just increase the **External Side Chain** until you get enough gain reduction.

Low Cut and High Cut

The **Low** and **High Cut** can be used to filter the signal before it reaches the detector. A good practice can be to remove some of the (unwanted) low frequencies so that they don't affect the gain reduction. Another trick is to apply filtering so that only certain frequencies cause the detector to compress the signal. If you for example compress a drum kit, you might want to keep the boominess of the bass drum, but compress the cymbals and snare. Set the **Low Cut** to 200-300 Hz so that the bass drum will get filtered out before it reaches the detector.

Please keep in mind that **Low Cut** and **High Cut** don't alter the direct signal, just the signal that enters the detector.

Please note that the 200 and 1000 Hz settings on the **Low** and **High Cut** knobs corresponds to the knob being set at 12 o'clock (where the little indicator lines are located). The easiest way to get to that setting is to click on the 200 or 1000 label on the panel.

Lookahead

The **Lookahead** can be used if the fastest attack time isn't fast enough, for instance if you want to apply some heavy limiting to a signal. The **Lookahead** will delay the signal path outside of the detector by up to 1 ms, leaving the detector signal unaffected. This has two effects:

1. The output signal will be delayed by up to 1 millisecond.
2. The detector will "see" the undelayed signal, but reduce the gain on the delayed signal. This means that the detector will be able to compress the signal before the actual transient comes (since the actual transient has been delayed).

The **Lookahead** time corresponds to the total latency of the plugin. The plug in will not report this latency to the host, since almost no hosts support compensating for a delay that depends on a parameter.

Some very fast **Attack/Release** settings will cause a "crackling" sound on transients, often sounding like some sort of digital clipping or saturation somewhere in the signal chain. This is very typical for this kind of compressor, but is usually an unwanted effect. The normal way to get rid of this "side effect" is to slow down the attack and release times but now you have another tool to use. By adding a little lookahead to the detector, many of these side effects can be avoided.

Parallel Inject

Parallel Compression is a technique where you blend the compressed signal with the original signal. Say that you have a heavily compressed drum track where all the transients have been lost (compressed). By adding a little bit of parallel compression you can blend in the original signal (with the original transients) with the highenergy compressed signal (without transients) and thus get the best of both worlds.

This is a technique that is very easy to achieve with a send bus, but this baby does a little bit more than that. First of all, you won't get any problem with the **Lookahead** since the dry signal is delayed with the same delay time as the compressed signal. Then the dry signal will be affected by the same analogue modeling mojo as the compressed one, which means that some of the distortion and coloring of the compressor will also have an effect on the dry signal. And at last, the **Output Peak** meter will of course show the sum of these signals, so that you can set a proper output level.

Whenever you change the gain reduction (for example by changing the **Input** volume) you will need to change the **Parallel Inject** since the volume of the compressed signal has changed. But be careful sometimes the dry signal is much louder than the compressed signal!

Status Indicators

The FET Compressor comes with a couple of handy status labels, located in the VU meter section.

Remote (Pro Tools only)

Whenever automation is used, or if an external control surface is connected, the **Remote** status indicator will light up. The knobs that are being controlled will also get a thin frame around them.



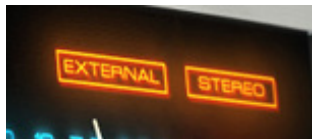
External

This indicator displays when an external side chain is present. Whenever an external side chain is present, it is possible to use the External Side Chain knob to set the amount of gain of the external signal to the detector.

In Pro Tools it will light up whenever you have connected a side chain bus to the plugin. This indicator will always be lit in hosts that use the Audio Unit (AU) format. In VST, it will light up if the host accepts side chain inputs, but this doesn't work for all VST hosts... So it goes.

Stereo

The **Stereo** indicator lights up if the compressor was instantiated in stereo mode.



Tweaking Tips

We didn't want to include too much of the standard "this is how a compressor work" theory, but rather focus on the parts that make this plugin special. Here are our favorite tweaks.

I want distortion!

There are two types of distortion in a compressor like this, first we have the input and output stage distortion and second the (frequency dependent) distortion caused

by the compressing. To get the first type of distortion, just increase the **Input** until you get enough distortion. You can even have **Ratio** set to 1:1 (=no compression). The other type of distortion is usually caused by a fast attack and release time. If you set **Ratio** on **ALL** and **Release** on **FAST**, you'll get the fastest release time possible. Try the *Distortion* presets.

The compressor isn't fast enough

If you find that the **Attack** time isn't fast enough (doesn't squash all the transients), increase the **Lookahead** until you're satisfied.

The compressor is too fast

The **Attack** times get slower if **Ratio** is set on **ALL**, so in order to get a late attack set **Ratio** on **ALL**. If you want to increase the transients but compress the rest of the signal, use the "all buttons in" mode. Try the *Transient* presets.

It makes crackling sounds on the transients

This is because of the extremely fast attack time. First you can slow down the **Attack** times so that the gain reduction becomes not as abrupt as it was. To compensate for the slower attack time, increase the **Lookahead**.

All the energy and attack of each note disappears

It is a fast compressor and chances are that it will compress fast transients (yeah?). That's why we added the **Parallel Inject**. By increasing the **Parallel Inject** you can blend in a little bit of the original signal (with the original transients). Try to balance it so that you get the transients from the dry path and the rest of the signal from the compressed path.

All the energy and attack of each note disappears (pt. II)

Another trick you could try, if you'd like to restore some of the transients is to narrow down the detectors frequency bands by adding some **Low Cut** and **High Cut** filtering. It doesn't work on all program material, but it's worth a try.

I want some cool drum bus tricks

Ok, here are our favorites:

1. **Fat Bass Drum:** Apply heavy compression on your drum bus. Increase the **Low Cut** knob until the bass drum doesn't trigger the compressor (at about 200-300 Hz). In most cases, this will make the bass drum sound fatter and louder compared to the rest of the kit.
2. **Pumping Drums:** Set **Ratio** on ALL, **Release** on SLOW and adjust the **Input** until you're satisfied. Blend in some dry signal with **Parallel Inject**.
3. **Sustained Drums with Transients:** Apply heavy 20:1 compression with fast release and attack times, blend in some transients with **Parallel Inject**.

Adding color

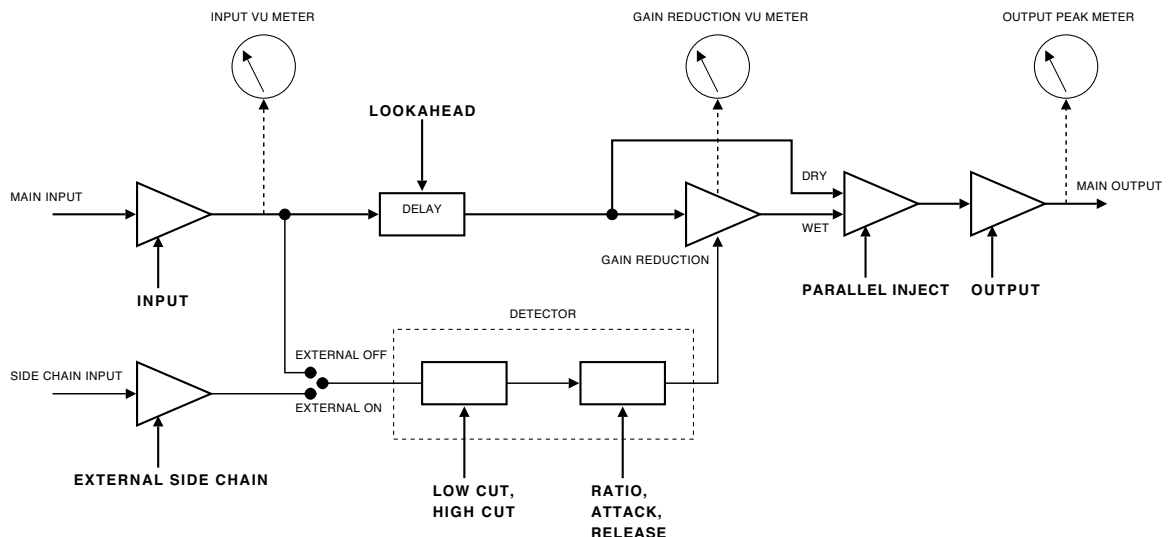
Finally you shouldn't be afraid of using this plugin in the 1:1 mode. You won't get any compression, but it will color the sound. And the meters look nice too.

Block Diagram

Below is a block diagram over the FET Compressor functionality. Please note that this illustration depicts the functionality of the plugin – the inner workings are quite different. The experienced reader will for example notice that some parts, like the feedback connection of the compressor, have been omitted in the illustration for the sake of clarity.

Credits

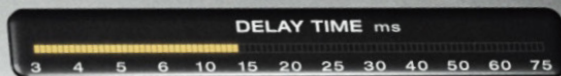
Oscar Öberg – modeling and implementation. **Torsten Gatu** – framework programming. **Niklas Odelholm** – framework programming and graphic design. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering and graphics. **Dan Lumbye** – A/B testing. **Stefan Fandén** – feature hunter. Thanks to **Lars Nygaard** and **Anders Bech** at Cyberfarm (DK) and **Per Åkesson** at Care Of Sound Studio for letting us use and abuse their equipment.





FROM THE MIND OF PAUL WOLFF®

doubler



14 Fix Doubler

DOUBLE TRACKING OR OVERDUBBING of tracks and instruments is a studio practice that goes back to the early roots of recording. Although pioneers like **Les Paul** experimented with overdubbing on recordings during the 1930s and 1940s, this method wasn't adopted as a studio practice until the arrival of the multi-channel tape recording machines. While overdubbing became a studio standard, especially on lead vocals, time constraints and demands from musicians led to the next innovation in dubbing: *Auto Doubling*.

While some engineers already had experimented with short slap-back echo to achieve a similar effect, this technique was further developed in the 1960s with the "*Artificial Double-Tracking*" of **Abbey Road Studios**. With the rise of digital signal processing in the 1970s, these techniques were further developed and gave birth

to the first digital harmonizers and doublers. Dedicated digital units were also developed by this time, but were strictly limited to studio works due to their size. Later, the Flanger effect, a side-effect of auto doubling, was popularized as an effect for guitar and drums.

This plug-in is based on the Real Time Auto Doubler AD-2 by Paul Wolff. It was a unique hardware designed to do classic tape-style doubling, and quickly became loved by musicians and sound engineers, but unfortunately never made it into serial production. The Fix Doubler is what the AD-2 was, and more. Paul Wolff has expanded the original concept and introduced several new techniques in order to give you an auto doubler that combines a classic sound with the versatility of the digital world. The Fix Doubler delivers texture and extraordinary richness on vocals and polyphonic instruments.



1. Engage
2. Stereo
3. Auto double
4. Dual sweep
5. Delay meter
6. Waveform type
Triangle, Sine, Random
7. Mix
8. Regen
9. Delay offset
10. Sweep
11. Rate

User Interface

The Fix Doubler panel contains the following ten controls.

Mix Sets the balance between the doubled and the direct signal. Set it to **INPUT** to only get the input sound, or fully to **DELAY** to only hear the delayed signal. Set it somewhere in the middle (30-70%) to get a nice doubling effect.

Engage Use Engage as a smooth bypass if you want to turn on or off the effect during a song.

Stereo Turn on to widen the stereo image, or turn a mono signal into a stereo signal. Make sure to insert the plug-in as a **STEREO** or a **MONO-TO-STEREO** plug-in for the widening effect to work.

Dual Sweep can also be used to widen the stereo image or turn a mono signal into stereo.

Regen Sets the amount of delay feedback. Turning **Regen** up makes the effect sound fuller, but too much **Regen** will make it sound flangy and hollow. This control is not active (greyed out) when **Auto Double** is in use.

Auto Double Turns on and off the **AUTO DOUBLE** mode. The **AUTO DOUBLE** mode will put the secret sauce in the stew and make everything sound great!

Delay Offset Sets the delay offset of the doubled signal. Longer delay times will yield a more slap back kind of doubling while shorter delay times will sound more flanger-like.

Sweep Sets the balance between the fixed **Delay Offset** and the sweep. Turning this knob clockwise will cause the sweep to affect the doubled signal more.

Rate Sets the speed of the sweep, ranging from 0.01 milliseconds to 6 milliseconds. The sweep amount is also reversely affected by sweep rate (see below), so that faster rates makes the sweep less apparent. This is by design to ensure a more natural doubling sound and a constant pitch change in the delayed signal.

Dual Sweep Turn this on to get two independent doublers out of phase with each other for the **LEFT** and **RIGHT** channel. This can make the doubling effect sound fuller, and also create a nice stereo effect for mono tracks.

Sweep Type Selects the type of waveform used for the sweep: **TRIANGLE**, **SINE** and **RANDOM**.

TRIANGLE: Constant pitch during the sweep, suitable for low **Sweep** settings and **Dual Sweep**.

SINE: Pitch is changing throughout the whole sweep, nice on monophonic instruments and higher **Sweep** settings.

RANDOM: The pitch shift is random, and will sound more natural when used sparingly. Low **Sweep** and **Rate** is preferred for this one.

Tweaking Tips

Here are a few pointers to guide you when using the Fix Doubler:

Delay Offset

Using 20 to 45 milliseconds of delay and subtly mixing the delayed signal with the original tends to “fatten up” the sound, especially when using the **STEREO** or **DUAL SWEEP** modes. This creates an illusion of more than one singer or player and is also an excellent method of adding lushness to string or horn sections. Instruments with sharp transients, such as rhythm guitars or drums, will typically require a shorter delay, vocals medium delays, and strings and horns slightly longer delay times.

Mix

Mix ratio is essential to the perceived “fatness” of the resulting signal. A completely wet signal (**Mix** fully clock-wise) can be OK when using more extreme settings in order to create vibrato effects, but normally a mix between 20% and 50% is recommended to create a subtle, natural sounding vocal doubling or chorus.

Sweep

The Sweep is the key ingredient when creating a subtle or fat doubling chorus. Too much **Sweep**, and it will sound pitchy and wobbly, too little and you’ll just get a slap back echo effect.

Chorus Effect

A natural sounding chorus effect is easily obtained by using the auto double function while setting the delay offset at a low setting (around 20–30 milliseconds), **Rate** at around 3–4 Hz and **Sweep** almost all the way to **OFFSET**.

Tape Wow and Flutter

By using a short **Delay Offset** at around 7 ms, **Sweep** set to 2–3%, **RANDOM** waveform with slow **Rate**, and **Mix** fully clockwise, you will get a tape machine like wow and flutter effect.

Original Auto Doubler hardware, the Real Time Auto Doubler AD-2 from 1979.



Vibrato

A vibrato effect can be achieved by using a completely wet signal, a fast sweep (**Rate** at 5–6 Hz), **Sweep** at 5–10%, and `SINE` or `TRIANGLE` waveforms. Use **Delay Offset** and **Regen** to get different characters of vibrato.

Credits

Paul Wolff – idea and original reference. **Arvid Rosén** – model analysis and implementation. **Niklas Odelholm** – programming and GUI. **Kristofer Ulfves** – User manual and initial testing. **Paul Shyrinskykh** – quality assurance. **Ulf Ekelöf** – 3D rendering and graphics.



15 Fix Flanger

FLANGING IS A MODULATION EFFECT that has fascinated music creators and sound engineers for decades. The instantly recognizable “swoosh” sound of a flanger has been widely used both subtly, to add weight and depth to horn sections, as well as heavily for very obvious sci-fi sweeps on everything from the sacral synths of **Jean Michel Jarre** to the big drums of **Led Zeppelin**.

The first flanging experiments are attributed to legendary guitarist and sound engineer Les Paul in the 1940s. But it was during the recordings of the **Beatles** classic *Revolver* that the effect was achieved when **John Lennon** and studio engineer **Ken Townsend** played back the same sound through two synchronized tape machines, and lightly pressed the flange of one of the playback reels about 20ms while recording onto a third machine. This slight delay mixed with the original signal caused a flanging effect—a swirly, metallic jet type of sound.

In the 1970s, dedicated flanging machines came up but were limited to studios due to their large physical size. Later, the flanger was popularized by the flanger guitar pedals that relied heavily on analog delay with the bucket brigade echo technique.

In 1979 **Paul Wolff** formalized a vision. This vision was a flanging and doubler unit with the sound and user experience of classic tape flanging and vocal doubling, but with the advantages of electronic circuits rather than mechanical tape reels. This vision was turned into a few hardware boxes loved among musicians and sound engineers alike, but they unfortunately never made it into mass production.

The **Fix Flanger** and **Doubler** is the reincarnation of his vision, two great plug-in effects featuring the versatility of digital world combined with warmth of the analog world.

For more information regarding the Fix Doubler, please see the Fix Doubler chapter.



1. Engage
2. Delay polarity
3. Stereo
4. Flange meter
5. Tap
6. Tap/4
7. VSO mode, Offset mode
8. VSO
9. Mix
10. Regen
11. Envelope
12. Delay Offset
13. Sweep
14. Rate
15. Auto/Man. Sweep
16. Servo

User Interface

The Fix Flanger consists of two main sections, **AUTO SWEEP** and the **MANUAL SWEEP**. These sections both control the flanging effect, but **AUTO SWEEP** acts more like a regular LFO swept flanger, while **MANUAL SWEEP** lets you control the flange just like you would if you had two synchronized tape machines. Outside of these sections you have two controls that globally affects the amount of flanging, **Mix** and **Regen**.

Mix Sets the balance between the flanger and the direct signal. The plug-in only outputs the original signal when set to **INPUT**, and only the delayed signal when set to **DELAY**. To hear the flanging effect between the input and the delayed signal, make sure that you set **Mix** somewhere around 50%.



Global Parameters

Auto Sweep Section



Regen Sets the amount of delay feedback in use. Full positive or negative regeneration will set the flanger into self oscillation. It's a nice effect, but watch your speakers!

Engage Use **Engage** as a smooth bypass if you want to turn on or off the effect during a song. The DAW's built-in bypass can cause artifacts when turning on/off the Fix Flanger.

Delay Polarity This is the polarity of the feedbacked signal. Different polarity of the feedback results in different sound of the flanging effect.

Stereo Enable **Stereo** to get a wider stereo image. If you are using a STEREO input and output, you will get a

wider and more vivid image. If you are using a MONO input with a STEREO output, the **Stereo** function will make a stereo signal from your mono input. If you are using this function in mono-to-mono, the flanging will sound a bit fuller and a tad different, but won't make much sense otherwise.

Auto Sweep section

Envelope Use **Envelope** if you want the level of the input signal to affect the flange. With **Envelope** active, a loud sound will force the delay of the flange towards zero, which is a very cool effect on drums. Use sparingly to get a more random flange effect on for instance vocals.

Delay Offset Sets the starting point of the **AUTO SWEEP**. With **DELAY OFFSET** set to 0, the sweep will be both positive (delayed signal) and negative (pre-delay). If you just want a typical delayed flange, set **Delay Offset** to a positive value.

Sweep Sets the balance between the **Delay Offset** and auto sweep. When **Sweep** is set all the way counter-clockwise, the **Delay Offset** determines the delay time of the flange. When set all the way clockwise, the full range of the sweep will make the delayed signal sweep back and forth +/- 15 milliseconds across the zero crossing point.

Rate Set the sweep rate, ranging from 0.01 Hz (100 seconds) to 4 Hz (¼ second).

Tap Tap repeatedly to set the sweep rate. This button cannot be automated.

Tap /4 Divide the current sweep rate by four. This button cannot be automated.

While using the envelope, the best results are often achieved by using a moderate amount of envelope modulation on the flange.

Manual Sweep section

VSO In **OFFSET MODE** the **VSO** knob sets the delay. In **VSO MODE**, it sets the speed of the delay change (see “VSO Mode” below).

Auto/Man. Sweep Sets the balance between **AUTO SWEEP** section and the **MANUAL SWEEP** section. When set fully anti-clockwise, only the **AUTO** section will affect the delayed signal. And of course, when set fully clockwise, only the **MANUAL SWEEP** section will affect the delayed signal and thus the flange effect.

Servo The amount the delay bounces when the delay is changed. When set fully clockwise, this knob will induce a form of “sluggishness” and smoothness when quickly changing the offset or simulated speed of the slaved tape (**VSO**). **Servo** will also affect the parameters on the **AUTO SWEEP**, but to a lesser degree.

VSO mode In **VSO MODE**, the **VSO** knob emulates how a **vso** (Variable Speed Oscillator) knob works on a tape machine. By having two tape machines, where you vary the speed of one machine while blending both outputs together, you achieved the original Beatles style tape flanging. The **vso MODE** emulates this behavior. This means that rather than directly setting the delayed offset between the dry and delayed signal, the **VSO** knob will



set the relative speed of the slaved tape recorder and thereby creating slightly different kind of flange behavior. When the **VSO** knob is set to -50 ms, also marked **HALT**, the tape reel will stop.

Offset Mode In **OFFSET MODE** the **VSO** can be used to directly set the delay the audio, ranging from -50 to 50 ms.

Tweaking tips

The Flange effect is highly dependent of the setting of the **Mix** knob, since the typical hollow flange is created by the differences between the delayed input and the original input.

Since the Fix Flanger can delay the signal with negative delay times, the dry signal passing through the plug-in will always have a 50 millisecond latency (look-ahead)

which makes it less suitable for using live. However, Fix Flanger will always work well as post-recording treatment, and all major DAWs will automatically correct this latency.

Through-Zero Flanging

For true “through-zero flanging” set the **Mix** balance to 50%, use negative **Delay Polarity** (lamp indicator is ON) and set **Regen** to OFF. This will completely cancel out the signal when the delay passes through 0 milliseconds.

Using the VSO mode

Since **VSO MODE** emulates much how an old-school tape flanging rig works, one has to imagine flanging as trying to keep “the second tape-machine” in near sync by speeding up and slowing down its speed against the dry signal. This is achieved by looking at the flange meter and trying to keep the flange as much as possible in the middle by “chasing” the through-zero pass-through with the **VSO** knob. The servo motor emulation amount can also be handy in order to induce some natural “motor lag” when chasing the flange up and down the sonic spectrum.

By using `VSO MODE` with a completely wet signal (**Mix** set to `DELAY`), you can emulate a tape stop by turning the **VSO** knob all the way counter-clockwise.

Chorus Effect

By using the `AUTO SWEEP` section and setting the **Sweep** knob almost fully to `OFFSET (2 - 5%)`, you'll get a nice chorus-like effect. Adjust the **Mix** between 25-75% to set the amount of the effect. **Delay Offset** should be around 10-15 ms, but by setting it to positive or negative offset you can get different styles of chorus.

Credits

Paul Wolff – idea and original reference. **Arvid Rosén** – model analysis and implementation. **Niklas Odellholm** – programming and GUI. **Kristofer Ulfves** – User manual and initial testing. **Ulf Ekelöf** – 3D rendering and graphics. **Paul Shyrinskykh** – quality assurance.



16 Fix Phaser

THE PHASER WAS ORIGINALLY INVENTED BY A SOUND ENGINEER in the mid 1960s while trying electronically emulate the effect of a rotating Leslie-cabinet. Although similar effects had already been produced by others with experimental equipment earlier, the first commercial product to feature the phasing effect did not surface until the late 60s.

A phaser effect is created by splitting input audio into two different paths and then mixing them together. One path treats the signal with all-pass filters, which preserves the amplitude of the original signal but alters the phase. The amount of phase change is frequency dependent. When this phase-altered signal is mixed together with the other path, the dry signal, the frequencies that

are out of phase will cancel each other out which creates the characteristic audio phase-notches. By changing the dry/wet mix the depth of these notches are altered. The deepest notches occur when the dry/wet mix ratio is 50% (this is the preset value in Fix Phaser).

The Difference between Flanger and Phaser

Historically, there has always been much confusion about what is phasing and what is flanging: The difference between phasing and flanging is that the latter creates an unlimited series of equally spaced notches and peaks, where phaser creates a limited number of notches with uneven spacing. This means that flanging is very much a similar sounding effect to phaser, but in the Flanger, the notches are linearly spaced in the frequency domain. Flanger is typically created by mixing a delayed version with itself and often tend to sound more pronounced and natural (“jetplane”), whereas phaser often sound more electronic and otherworldly (“space-whosh”).

Overview

The Fix Phaser is a very versatile effect with the ability to re-create everything from the phasing sounds of various classic gear to the more vintage sounding vibratos derived from the late 60s. Like the other Softube Fix products it has a dry/wet Mix knob giving the user the freedom to freely set their desired blend of dry input and the fully wet signal.

The Fix Phaser features three main phasing modes – Normal, Effect and Wide. Normal mode is the classic phase-shifting sound with all the liquid transparency, whooshing sweeps and tingling swirls we’re all used to. Effect mode adds a pronounced chasing stereo effect where the left channel's sweep is out of phase with the right channel's sweep, and Wide mode further expands

on this principle where a subtle widening effect is still apparent even when using small sweep amount.

The Fix Phaser is swept up and down by a sweep-generator with variable amount, rate and sweep type.

Finally, the Fix Phaser also feature an envelope-follower with fast and slow mode, positive and negative response, that makes it possible to use create auto-wah style effects among other things.

User Interface

The Fix Phaser panel contains the following twelve simple and easy to use controls. This is a brief description of each and everyone:

Engage (Out/In)

This button turns effect on and off. Effect is bypassed when Engage is set to “out”.

DAW Sync (Sweep/Sync)

This locks the rate of the Sweep generator to the tempo of your DAW when set to “Sync” (DAW sync lamp is lit).

Stereo (Normal/Effect/Wide)

- **Normal** – this is classic phaser mode where both sides of the stereo input is treated equally.
- **Effect** – enhances the stereo effect of the phaser by using the same phase-shift on left and right side but with different phase on the sweep. This gives the phaser more of a “chaser”-kind of effect where one side always is chasing the other.
- **Wide** – left and right phase will change in amplitude and polarity in a preset ratio to the sweep



- | | |
|---|------------------|
| 1. Engage | 9. Level meter |
| 2. DAW Sync | 10. Regen |
| 3. Stereo
(Normal/Effect/Wide) | 11. Envelope |
| 4. Slow | 12. Phase offset |
| 5. Modal centers | 13. Sweep |
| 6. Stages
(2, 4, 6, 8, 10) | 14. Sweep meter |
| 7. Sweep type
(Triangle/Sine/Elliptic) | 15. Rate |
| 8. Mix | |

in addition to the effect mode. Wide mode makes the soundstage wider and creates a subtle, musical movement which is very handy when mixing static program material.

Note: When using Phaser with Stereo mode set to Wide, the sweep modulation will always affect some aspects of the phasing with a preset value which is not affected by the Sweep (amount) knob. This is by design and intentional behaviour.

Stages (2,4,6,8,10)

These buttons set the amount of phaser stages used. Two stages creates one notch, 4 stages creates two phaser-notches etc. This is one of the key elements to look at when re-creating the phaser sound of classic gear.

Sweep Type (Triangle/Sine/Elliptic)

Selects the type of waveform used for the sweep:

- **TRIANGLE-waveform:** Suitable for creating a regular, ongoing phaser movement.
- **SINE-waveform:** Can be used for more natural sounding phaser movement.
- **ELLIPTIC-waveform:** This sweep type spends more time on high frequencies than lower ones.

Slow (Fast/Slow)

This switch determines how swift the envelope follower will react to an incoming signal. Use Fast response for drums, rhythmic guitar or clavinet picking, and Slow response for sustained chords and synth-pads.

Mix (Dry/Wet)

This knob set the mix between dry (unaffected) input signal and the fully wet signal. When set to fully wet the Fix Phaser can be used as a vibrato-like effect, bending input notes up and down in sync with the sweep rate.

Regen (0-100%)

Sets the amount of feedback for the end of all pass chain. A large amount of Regen make the Fix Phaser sound more electronic and out of this world.

Envelope (-100% - 0 - 100%)

This is the amount of which the envelope follower will affect the modal center movement. It is a polarized knob which means that the counter-clockwise position at -100% means that it will have a negative sweep impact on the phaser movement. AWith the knob at 12 o'clock it has no effect (default value), and at fully clockwise, it will have full positive effect on the phaser movement.

Phase offset (20hz - 20kHz)

Sets the offset of the phased signal.

Sweep (0-100%)

This knob sets the Sweep amount to affect the phaser movement back and forth.

Rate (0.01hz - 6hz)

Sets the speed of the Sweep, ranging from 0.01 to 6 hz. This means that, when Rate is set to lowest the lowest speed, 0.01 hz, it will take 100 seconds for the sweep generator to complete its cycle, while at the highest rate completion will occur 6 times per second.

When DAW sync is set to “Sync”, Rate is divided into 8 bars, 6 bars, 4, 3, 2, 1.5, 1 bar, half bar, dotted quarter, quarter and eighths parts of a bar. These values represent how many time-measures it takes for a sweep cycle to complete.

Meters

Modal Centers

This meter on the upper part of the plugin shows how many phase notches that are active and approximately on which frequency they are currently at work. More stages selected by the stage buttons means more notches.

The modal center display is also a handy tool in order to understand the envelope sweep and polarity, phase offset and sweep rate and range.

Level

This meter reflects output level. Keep this meter in the green or yellow for optimum response. Lower input volume or Regen if this meter is in the red.

Sweep

This meter shows the movement of the sweep generator at all times and is not affected by the Sweep (amount) knob.

Tweaking tips

Here's a few pointers to guide you when using the Fix Phaser:

Mix ratio is essential for getting the most effective phase-cancellation - the preset value 50% is will give you the deepest phase-cancellation. If the mix ratio is raised towards 100% wet, the phase-effect will get more "pitchy", especially with many stages selected.

This "pitchy" effect can be used to create vibrato effects with the Fix Phaser. Set Mix ratio to 100%, use 6 or more stages and a fairly high (30% +) sweep and you'll start hearing pitching effects.



By using extreme mix settings (100% wet) in combination with the Stereo “Wide” mode, tremolo, as well as combined “tremolo + vibrato” effects can be achieved in the same way.



The number of stages selected will greatly affect your phaser sound. Try for example a slowly sweeping phaser using 2 or 4 stages with high Regen. Now, try instead 10 stages and listen to the difference. The more stages, the more “hollow” and “electric” phaser characteristics.



You can also use the Phaser with a low dry/wet ratio (25-30%), low sweep (30%) with a fairly swift sweep rate (3 Hz), and no regen in order to create a “Chorus”-like effect.



The envelope-follower feature of the Fix Phaser is great for creating auto-wah like phasing. Use the Fast mode for staccato, funky fast playing or drums, and use the Slow Mode for sustained chords, strings and pads.



Finally, the Fix Phaser can also be used as a subtle or not so subtle stereo widener in the Stereo “Wide” mode. Using the Phaser with 60-70% wet mix, many stages selected and Sweep set at minimum (0%), the Regen and Offset knobs can be used to set the focus and “hollowness” of the stereo movement. The speed of this movement is set by the Sweep Rate knob.



Credits

Paul Wolff – idea, testing and feedback. **Arvid Rosén** – model analysis and implementation. **Paul Shyrinskykh** – algorithms testing and feedback. **Niklas Odelholm** – programming and GUI. **Kristofer Ulfves** – User manual and testing. **Joe Chiccarelli** – Presets and testing. **Igor Miná** – User Manual layout. **Ulf Ekelöf** – 3D rendering and graphics.

17 Focusing Equalizer

THERE ARE MANY EQUALIZERS OUT THERE. Some are extremely advanced with built-in spectrum analyzers and intelligent super modes. Others are simple and nothing more than your average digital equalizer. While some claim to possess super powers, the differences are usually just in the user interfaces, and not always in the sound. Others claim that there is only one way to make a digital equalizer, and therefore every digital equalizer sounds the same. That's not true. Modeling a three band parametric equalizer is more than taking the three separate bands and putting them together. The real electronics interact with each other so that the sum becomes more than its parts, and this sum isn't possible to re-create by stacking up a bunch of standard digital equalizers.

We set out to model two vintage equalizers as exactly as possible. The result are the **PASSIVE** and **ACTIVE** equalizer that are probably installed on your computer right now. The **ACTIVE** is the most traditional of the two. Use it as a standard equalizer and feel comfortable about it sounding the way the real gear does. The **PASSIVE** is a bit more quirky but just fantastic sounding. An excellent help when you need that glimmering high-end or warm low bottom or *je ne sais quoi...*

If you combine these two, you'll get what we call the **FOCUSING EQUALIZER**. It is designed to be a fast and efficient work horse that immediately produces the desired results. The combination of three bands that automatically follows the High and Low cuts and a compressor-saturator-distortion-mojomaker-tool makes it great tool to tidy up and fix your tracks.



We hope that you'll enjoy this set of equalizers and that you will find that they are excellent additions to your toolkit that will help you make great sounding music!

The Trio (“Passive-Active Pack”)

The **FOCUSING EQUALIZER** is a part of a trio together with the **PASSIVE EQUALIZER** and the **ACTIVE EQUALIZER**. The **PASSIVE** and **ACTIVE** have very different sounds, while this unit combines these two into a new type of equalizer with a classic sound. We can assure you that there are no other equalizer plug-ins that sound like these.

At a first glance, you will notice that these three units don't look like your average equalizer, and at a closer inspection you'll see that the knobs behave in a peculiar way. This is because we have chosen to keep the way the original units worked, and while it feels quirky in the beginning you will get used to it pretty fast.

The Story

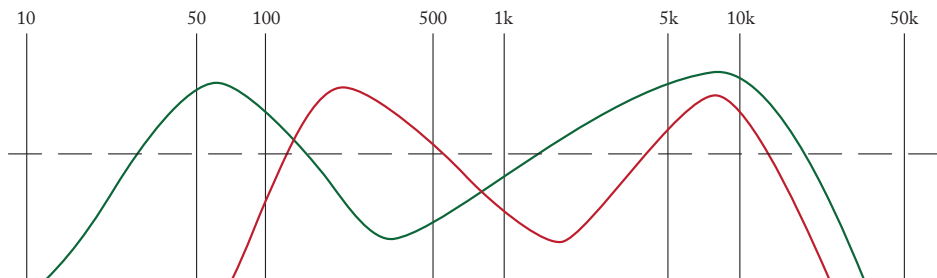
The **FOCUSING EQUALIZER** was actually the main goal of this project. We have had the idea for a long time, and we developed prototype after prototype to try the concept. It sounded good, but we had a hard time getting the right character. While showing it to one of our main dealers in Stockholm he asked us what kind of filters we used and why we didn't model a great sounding passive equalizer and put that inside the **FOCUSING EQUALIZER** instead of the off-the-shelf EQ algorithm we used in the prototype. At the store he had tons of equalizers and we listened to almost all of them, not really satisfied with what we heard. At the end of the visit he brings this Neumann and says, “Oh, I almost forgot this one. We had it here for some reparations, but the customer won't notice if you borrow it for a while!” We fell instantly in love with the sound, and when we put it inside the **FOCUSING EQUALIZER** we

immediately got what we had been looking for. One drawback with the Neumann is that it isn't very sharp, and sometimes difficult to use to really control and shape the sound. One of the runner-up equalizers we tried was the Filtek, but we didn't really want to model it because of the extreme complexity. Eventually (and with some persuasion from our testers) we realized that the Filtek was exactly the counterpart to the Neumann that was needed for the **FOCUSING EQUALIZER** and the **PASSIVE-ACTIVE PACK**.

What Is Unique About the Focusing Equalizer?

The combination of three bands that automatically follows the high and low cuts. So if you set your low cut at 2 kHz, the low band will automatically get a center frequency at or above 2 kHz. The mid band will have its center frequency somewhere between the high and low cut. Sounds easy enough?

The graph below shows two settings, using the **ACTIVE** filter type. **Low** and **High Bands** are on full boost, and **Mid Band** on full cut. The difference between red and green graph is the settings of the **Low** and **High Cuts**, 50 Hz-12 kHz (green) and 200 Hz-10kHz (red), respectively. As you can see, the mid dip follows the settings of the cuts.



Work Flow at a Glance

1. Set the frequency range with **High** and **Low Cut**. This will **automatically** change the frequencies of the **Low**, **Mid** and **High** bands).
2. Choose **Equalizer Type** (tip: start with **ACTIVE**)
3. Adjust the equalization (**Low**, **Mid**, and **High Gain**)
4. Toggle **Equalizer Type** to hear how the sound changes
5. Add some **Saturation** if needed.

Work Flow

First of all, you should disregard the **Saturation** controls and only focus on the **Low** and **High Cut** faders. Try to set these so that you filter out all the unwanted parts of the signal. For most instruments other than bass and bass drum, you can go pretty high up with the **Low Cut** without really changing the sound. By setting the **Equalizer Type** to **PASSIVE** you will get a smooth and musical cut, while on the **ACTIVE** setting you get a more brutal filtering, more like a standard parametric equalizer.

Step 1: Frequency Range Adjust (Low and High Cut) and Equalization

After adjusting the **High** and **Low Cut** faders, the next step is to use the **Low**, **Mid**, and **High Gain** to shape the signal. These equalizer bands have been automatically set to natural frequencies within the selected frequency band. In practice, this means that the **Low**, **Mid** and **High** will always make sense.

If the **Low Cut** is set at 1 kHz and **High Cut** at 5 kHz, the center frequencies of the **Low**, **Mid** and **High** will always be within the 1 and 5 kHz range.

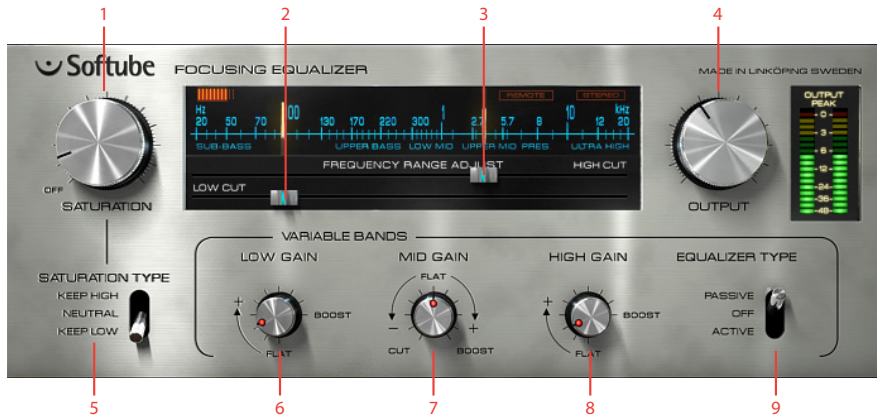
Step 2: Set Equalizer Type

When you have set the right character of your track, you can toggle between the **ACTIVE** and **PASSIVE Equalizer Type**, to see what character you prefer. In general, the **ACTIVE** is most suitable for narrow, focused sounds, while the **PASSIVE** is more suitable for wider sounds. The **PASSIVE** has (in our ears) an acoustic feel with a typical 60's glow, while the **ACTIVE** typically gives you a controlled 70's hard hitting sound.

Try to avoid setting the **Low** and **High Gain** on full blast while using the **Passive** equalizer type. By doing this, you will effectively cancel any effect from the **Mid Gain**.

Step 3: Use the Saturation and Saturation Type

The **Saturation** knob is a very effective way of boosting the energy of the signal. You can use the small saturation meter in the top-left corner of the frequency range window to see how hard it is working. It is not necessary to use the saturation at all, but it is a very useful feature.



1. Saturation
2. Low cut
3. High cut
4. Output volume
5. Saturation type
6. Low gain
7. Mid gain
8. High gain
9. Equalizer type

Knobs

Saturation Sets the amount of saturation. The saturation circuit is completely bypassed if the knob is set in the minimum position.

Saturation Type With **KEEP LOW** you will be able to distort the track without getting a farty bass sound. **KEEP HIGH** will do the same, but for high frequency material. **Neutral** will not do anything special with either the high or low frequencies.

Low and High Cut 6dB/octave (**PASSIVE** mode) or 18dB/octave filters (**ACTIVE** mode).

Low, Mid and High Adjust the gain of the filters. The **Low** and **High Gain** will only boost the low and high frequencies. If you need to cut, you will have to use the **Low** and **High Cut** faders.

Equalizer Type Choose between the **PASSIVE** type or an **ACTIVE** type equalizer. Affects all equalizer controls, but not the saturation circuit. You can also bypass the entire EQ circuit (and just use the saturation circuit) by setting this in the off position.

Output Output volume. Goes from nothing to +12 dB. At the 12 o'clock position, it's set at 0dB, and going clock wise will increase the output volume by +2 dB for each mark.

Meters

The Focusing Equalizers have the following meters and status indicators.

Saturation Meter This meter lights up whenever the saturation circuit is active and is adding saturation to the signal. Use this more as a visual indication than an absolute measurement of the amount of saturation.

Frequency Range

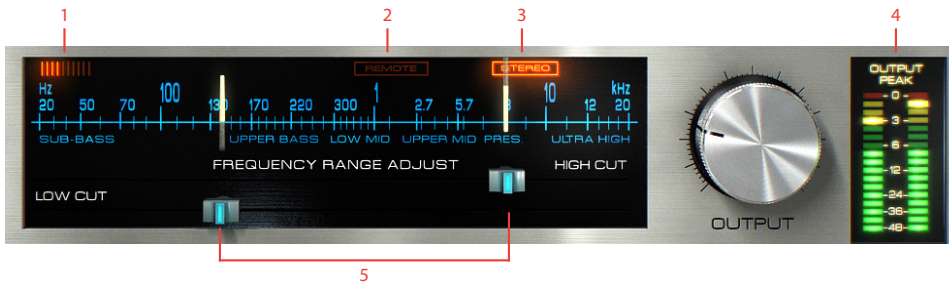
Guides These “tuner looking” guides help you see your selected frequency range.

Stereo Indicator Lights up if the plug-in has been activated in stereo mode.

Remote Control Indi-

cator (Pro Tools only) Whenever automation is used, or if an external control surface is connected, the Remote status indicator will light up and the (remotely controlled or automated) knobs will get a thin colored frame.

Output Peak Meter The output peak meter is a fast peak meter with hold values. If you use the plug-in in mono mode the left and right meter will show the same mono signal on both meters.



1. Saturation meter
2. Remote control indicator
3. Stereo indicator
4. Output Peak meter
5. Frequency range guides

Differences Between Passive and Active

This table highlights some of the differences between the three equalizers. Please note that there are a lot of other differences that make up for the characteristic sounds, and not many of these are easy to put in a table like this.

- 1) The Passive EQ doesn't have any cutting bell filters.
- 2) "Constant Q" wasn't defined at the time that the Filtek was built, but it shows many properties of a "Constant Q" equalizer.
- 3) The cut filters are shelving in the Passive Equalizer (due to parasitic resistance in the inductor) while this (unwanted?) property have been removed in the Focusing Equalizer.

Finally, a Note on Modeling

There are lots of buzzwords when it comes to recreating old gear in the digital domain. Many are those who claim they "model" stuff, few are those who actually do it. The advantage of real, component-based, physical/mathematical modeling is that you capture the entire behavior of a unit in one big sweep. The disadvantage is that the math can get very complicated, even for circuits

that doesn't look complicated at all. Even a simple thing like a bypass (such as in the Filtek) is complicated and causes problems when modeled, and it would be easy for us to take a short cut and mimic stuff when the modeling gets tough. But we don't do that at Softube, because we believe that you can hear the difference. So we do real modeling, for good and for worse.

Credits

Niklas Odelholm – modeling and graphics design.
Torsten Gatu – concept. **Oscar Öberg** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – graphics rendering. Thanks to **Stefan Fandén** and the crew at Deluxe Music for letting us borrow the gear!

	Active Equalizer	Passive Equalizer	Focusing Equalizer
Symmetrical Bell Shapes	Yes	N/A (1)	Symmetrical (ACTIVE), Non-symmetrical (PASSIVE)
Constant Q	Yes, sort of (2)	No	
High/Low Cut	18 dB/oct.	Shelving, 6 dB/oct. (3)	6 dB (PASSIVE), 18 dB (ACTIVE)
Filter Types	Bell 3-pole filters	Shelving 6 dB/oct. (High/Low) Bell 1-pole filter (Presence)	
Isolated Bands	Yes	No	



18 Harmonics

AFTER MORE THAN A DECADE OF analyzing and recreating the sound of classic studio hardware, we've hand-picked the best-sounding components from legendary units and placed them in one box to deliver a musical and versatile palette of analog flavors.

User interface



- | | |
|------------------------|------------------|
| 1. THD/Level meter | 9. Character |
| 2. Drive types | 10. High cut |
| 3. Meters | 11. Low cut |
| 4. Dry/wet | 12. Filter type |
| 5. Pre/post distortion | 13. Input |
| 6. Dynamics | 14. Input level |
| 7. Dynamics on/off | 15. Output |
| 8. Drive | 16. Output level |

Parameters

Drive types

Solid Inspired by units with hand-made quality solid state components from the early 70s. The distortion type will help any track to stand out in a mix, adding definition and separation in the soundstage. It will work great on vocals, drums and just about anything else.

Transf Inspired by the transformer sound of American consoles. The transformer will add the signature low-end tone of countless records from the analog mixing era. Full mixes, kick and bass tracks will shine when using this distortion type.

Master Distortion designed specifically for mastering/bus applications, leaving the transients intact. This type can be subtle, adding just the right amount of analog touch to the master bus, or, if driven hard, a smooth saturation bringing character and attitude to any program material.

Tube Inspired by a boutique tube circuit found in various pieces of iconic vintage gear from the mid-60s. Delivers the unparalleled warm tube sound that will benefit any bass or guitar tone.

Modern Inspired by an English all-valve modern distortion box with triode setting. This type adds instant warmth and punch.

Other parameters

Character Use this knob to adjust the character of distortion using mastering-grade filters. Turning clockwise favours high frequencies before the distortion, while turning counterclockwise favours low frequencies before distortion. The character filters are tailored for each distortion type.

Drive The Drive knob controls the amount of distortion.

Dynamics Allows you to add dynamics back to the processed/distorted material. It uses a unique algorithm that analyses parts of the incoming audio material before distortion and reconstructs the dynamics of the original waveform. Turn the knob clockwise to steadily increase the amount of dynamics. Turning the knob counterclockwise will add thickness and energy to the audio material, and bring up all the lower details. In combination with Drive it will add an unmistakable rich tone to drums, bass and guitars.

Dynamics On/Off Flip this switch to turn the Dynamics processing on and off.

Meter Use this switch to change the VU meters between input level display and THD (Total Harmonic Distortion).

Pre/Post The high and low cut filters can be applied before or after distortion. Use this switch to change their position in the signal chain.

High Cut Filter Lower the cutoff frequency of the High Cut filter to get soft, warm tone even when a lot of distortion is applied.

High Cut Filter Type Choose between 1, 2, or 3-pole filters to define the slope and sound of the High Cut filter parameter.

Low Cut Filter Raise the cutoff frequency of the Low Cut filter to roll off sub frequencies that can trigger excessive distortion in some program material.

Low Cut Filter Type Choose between 1, 2, or 3-pole filters to define the slope and sound of the Low Cut filter parameter.

Dry/Wet Blend the Dry and Wet signals with the Dry/Wet mix knob.

Input The Input fader controls the input gain.

Output The Output fader controls the output level of the plug-in.

Meters

VU The plug-in's main VU meters can display either input level, or THD. When in THD mode, the meters show the distortion amount. It is handy switching to this mode to track even the most subtle distortion levels.

Input/Output level These meters show the level of your signal at input and output.

In use

Bass

Every Drive Type will add different and unique flavours to the bass track. One of our favourites for bass and guitar is the Tube Drive Type. Set Drive to max, add a bit of Dynamics, turning the knob clockwise, and your bass track will shine with unique colors and life like never before. Add more definition to the sound by turning the Character knob clockwise. Add more rock solid low end by turning the Character counterclockwise and blending some Dry signal back by using the Dry/Wet control. You might want to add more Input Gain for even more distortion.

You can use several instances of Harmonics in series to add different character to top and low end.

Master bus

Choose the Master Drive Type and switch the THD/Level meter to the THD position. This will allow you to track the most subtle distortion. Please note: Drive in Master type doesn't only increase the harmonics amount, but changes the harmonic content as well, experiment with the position of the knob to find the sweet spot for your track. You may alter the flavour of the the Drive even more by turning the Character clockwise, adding mastering-grade sheen and clarity to the program material. Turning the Character counterclockwise will add punch and tightness in the low end. To adjust to the changes you might use the Low Cut and High Cut filters. It is recommended to use the 3-pole Filter Type on the Low Cut and the 1-pole filter on the High Cut to get the best results when working on your master bus.

Vocals

To add an analog touch and that extra clarity to vocals, choose a Drive Type to your taste and increase the drive until you hear a fair amount of distortion. Turn the Character clockwise to add more harmonics to the high frequencies. Use Dry/Wet to blend the Dry signal back and mix to taste.

Drum loops

To add some fatness and weight to the drum loops, turn the Dynamics knob fully counterclockwise and add Drive until the desired amount of distortion is achieved. You might want to blend some of the dry signal back by using Dry/Wet knob.

Credits

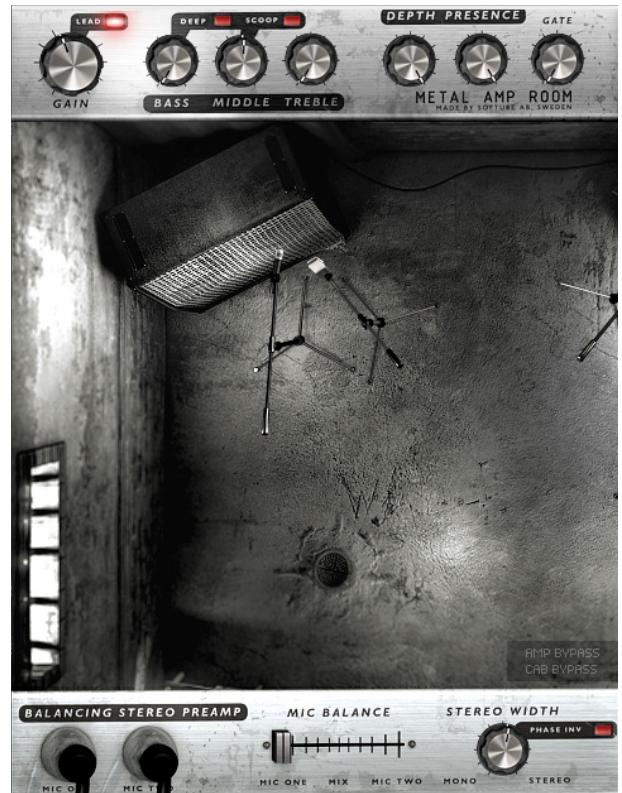
Paul Shyrinskykh – product management and sound design, **Kim Larsson** – modeling, **Niklas Odelholm** – graphic design, **Ulf Ekelöf** – 3D rendering, **Maxus Widarsson** – quality assurance, **Igor Miná** – user manual layout.

19 Metal Amp Room

Introduction

METAL AMP ROOM IS JUST WHAT IT SAYS IT IS. It's an amp room, just like the other Softube plug-ins Vintage and Bass Amp Room, but it's designed for Metal. It consists of three parts, of which the first part is the amp itself. It was carefully selected to have a aggressive sound with a smooth high end and never ending sustain, and special care was taken to make sure that the palm mute playing style hits you right in the stomach. We also added a lower gain (not to confuse with low gain!) preamp to accommodate for a more traditional playing style.

Second, and perhaps the most important of the building blocks, are the cabinets. You can choose between two different fourbytwelve cabinets, and each cabinet has two mics. A dynamic microphone for the classic metal sound, and a condenser mic with a fuller and richer sound. Just playing around with different cabinets and mic settings is a science in itself, and we have worked really hard to make it as easy as possible for you to find the sound you look for. To make this even easier, we had to design the third building block – the BALANCING STEREO PREAMP. This is basically two channel strips with a volume, a pan and a phase control each, that we combined into a single channel strip with a MIC BALANCE control instead of two volumes and a STEREO WIDTH control instead of two pan knobs. Just to make the handling easy, without compromising the sound.



And like our other products, such as Vintage Amp Room, Metal Amp Room has none of the added gadgets or weird-sounding presets sometimes found in simulation software. Focusing on quality and accuracy, it gives you a hard hitting metal sound that is authentically raw. The goal is not to give you a preproduced, ultimate guitar sound, but to provide you with the ultimate tool to create your own. Because after all, only you know exactly what sound you want.

Product Philosophy

Everything about this product is authentic. The raw and dynamic sound is an absolute replica of what you would get if you had this amp and these cabinets set up in a real studio. What you see on the screen are photorealistic 3D renderings of the complete setups, and you get to move actual mics around when deciding which positions are best, from near field offaxis to far field and back, continuously, without preset positions.

The idea is that using Metal Amp Room should be like working directly with this amplifier set up in a real studio, with two microphones in front of each cabinet, connected to your DAW. And we kept it simple. It has no knobs with dubious or unintuitive functionality, and no added gadgets or ridiculously fakesounding effects. We simply provide you with the same tools you have in a real studio, but in digital form. Then it's up to you to add compressors, EQ's, reverb, or any other effects, to get the sound you want. In short, you need the same skills to master this software as you need when dealing with a real studio setup. You can get back to doing what you do best, because music production is about using your ears, not your computer skills.

Supernormalize

The “supernormalize” feature makes sure that the output signal from Amp Room always has a reasonable volume. From a user perspective, this primarily means that the output never exceeds 0 dB and you don't have to fiddle with tons of output volume knobs. And even with low settings on the gain knob the output volume will be within a reasonable level and ready for digital recordings. Compare this to the real amp, which easily can have a dynamic range of 100 dB. From a technical perspective, this means that the signal path within Amp Room always has the highest dynamic range possible, and you won't lose any bits on the way.

User Interface

SIMPLICITY WAS THE GOAL when we developed this product—the amps and mics should work just as they do in the real studio. We have tried to imitate that as far as it is possible, and if you have experience of working in real studios, you will notice that Metal Amp Room handles and sounds just the way you expect it to.

Amp Panel



Room View



Mic Panel

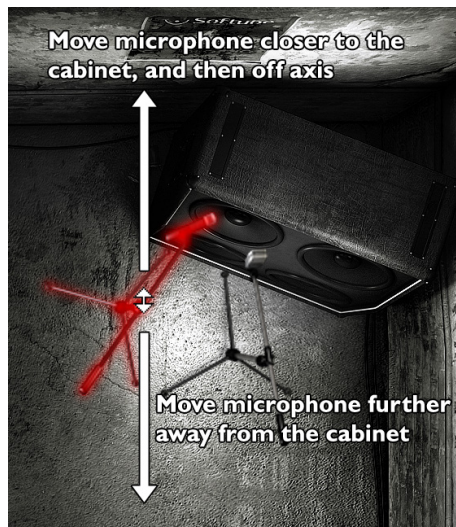


Amp Panel (Top Area)

In the amp panel you can alter the settings of the amp by clicking the knob and dragging the mouse up and down or left to right. Switches will be switched either by clicking on them, or by click-and-dragging the mouse. In some hosts you can change the behaviour of the mouse, but the default behaviour of Metal Amp Room is the one described above.

Room View (Middle Area)

In the room view you can do two things, select the other cabinet or change the mic's positions.



Mic Position

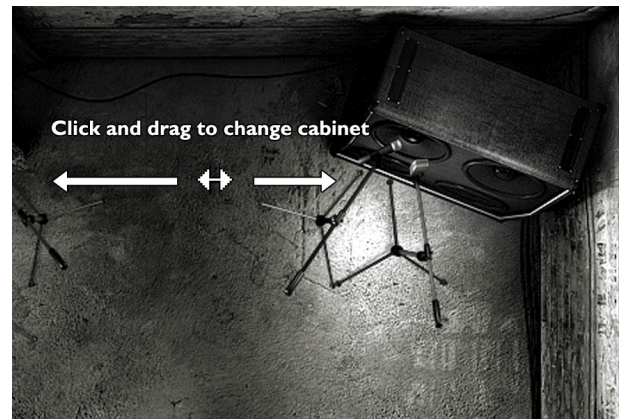
When the mouse is located over the mic stand, the mouse pointer changes to an “up-down” arrow and the mic gets a red light saber glow. Click and drag the mouse up or down to change the position of the mic stand. The microphone moves along a predetermined path, so you only need to move the mouse up or down, just as if you were changing a knob.

Linked Mics

If you press and hold Shift while moving a mic, the other mic will also move to the same position respectively.

Cabinet Selection

You can change the cabinet by clicking on the background and dragging the mouse to the left or right. The mouse pointer becomes a “left-right” arrow when you are pointing at the background to indicate that it is possible to change cabinet.



Mic Panel (Bottom Area)

In the mic panel you can change the balance and volume of the two microphones by setting the level from the two mics with the fader or the using the knob to alter the stereo width. The mic panel follows the cabinet selection so that you can have different settings for the different cabinets.

Clicking on the background while holding the Shift-key toggles through the cabs without any sliding animations.



1. Lead/Rhythm channel select
2. Deep bass
3. Scoop mids
4. Gate
5. Gain
6. Bass
7. Middle
8. Treble
9. Depth
10. Presence

The Amplifier

THE AMPLIFIER SIMULATED IN THIS PLUG-IN IS actually a hybrid between two different amps, where the **LEAD** channel (**Lead = ON**) represents one whole amp, and the **RHYTHM** channel (**Lead = OFF**) is a simulation of the same amp but with the input stage taken from an amplifier with a lower gain.

The **LEAD** channel is the main channel – the one to go to – and extreme care has been taken to make sure that the palm muted guitar hits you right in your face.

The **RHYTHM** channel represents a more classic rock sound, and is perfect for chords containing more than the root and the fifth...

We also took the liberty to add our own noise gate to the amplifier. It was specifically developed for guitar and focus on an extremely fast response. You will notice

that the gate closes (turns off the volume) more slowly if you let your notes decay slowly than if you mute your notes. So you can both play fast riffs and have the gate kick in between the chords, and play slow solos with a not so aggressive gate – all on the same setting.

Gain This knob controls the amount of distortion. When you have selected the **RHYTHM** channel, the **Gain** knob can be used to adjust the amount of preamp distortion. When you have selected the **LEAD** channel, the **Gain knob** is mostly used to shape the character of the distortion. When the **Gain** knob is set at 9 o'clock, you'll get a tight distortion with a fast response. As you increase the **Gain** you will get more distortion, but it won't be as tight as in the lower settings. A reasonable setting is around 12 o'clock.

Lead This switch selects between the **RHYTHM** channel (light is off) and the **LEAD** channel (light is lit).

Bass, Middle, and

Treble These knobs are the tone controls of the amplifier. This circuit is located after the preamp distortion and is used to shape the sound of the preamp distortion.

Deep Boosts the ultra low frequencies.

Scoop Scoops out the mid to create a tighter sound.

Depth, Presence These are used to control the amount of low and high frequencies in the feedback loop of the power amp and thus the amount of bass and treble in the power amp. Turn them up to get a low or a high frequency boost.

Gate A noise gate specifically designed for a guitar input. Mute your guitar and turn up the knob until the noise disappears.

The **Gate** is program-dependent, which means that if you play fast and tight riffs and end your notes distinctly, you will get a very fast gate. If you on the other hand play sustained notes, the gate will close more gently.

The Cabinets



The Metal Cabinet

The left cabinet is called the **METAL CABINET** since it has a stylish metal grille and an uncompromising metal sound. This is a predictable cabinet in the sense that there are very few surprises as you move the mics around. The **METAL CABINET** will give you an emphasis on mid-range frequencies, which makes it excellent for modern, tight styles. Very little post-processing (in the form of EQs, etc.) is usually needed since the raw sound has a produced character. Just do some minor tweaks to make it fit in the mix and you will be fine.

The Black Cabinet

To the right is the **BLACK CABINET**, which has a classic rock 4x12 sound. With this one, there are plenty of bass resonances as you move the mics around, producing a lively, sometimes almost unruly sound. To get a vintage sound, set the balance all the way to **MIC ONE** or blend in **MIC TWO** to get a more scooped sound. Depending on how the rest of the mix sounds and the exact positions of the mics, the **BLACK CABINET** could need a bit of work with the amp's tone stack and perhaps some external EQing to blend in properly.





1. Mic balance
2. Stereo width
3. Phase invert

The Microphones

Mic One

MIC ONE (to the left in both cabinet views) is a classic dynamic microphone. It has a tight mid range that often makes out the core of the sound. The character is focused and controlled when the mic is close to the cabinet, and gets a thinner ambient room quality (METAL CABINET) or more boomy and unpredictable (BLACK CABINET) as it is moved back. This mic is very directional, so when it is between the close onaxis and offaxis positions, small changes in angle will have a big effect on the sound.

Mic Two

MIC TWO (to the right) is a condenser microphone. Use it to add some scooped character to the mid range of MIC ONE or use it alone for a more open, broader, high bandwidth sound. Just like MIC ONE, the character is focused when the mic is close to the cabinet and gets more room ambiance (METAL CABINET) or more boomy sound (BLACK CABINET) as it is moved back. The far-field sound is still more controlled than the far-field sound of MIC ONE though.

The Balancing Stereo Preamps

EACH CABINET HAS THE “Balancing Stereo Preamp”. It isn’t modeled after some particular piece of gear, but was instead designed to make it as easy as possible to tailor the sound from the two microphones. The main ideas are:

1. It should be ridiculously easy to use without having to worry about anything but the sound you want to get.
2. It should be easy to monitor each mic (which is done by moving the **Mic Balance** fader fully to each side).
3. It should be easy to blend the mics and still get a stereo image that is balanced to the center.

Mic Balance Fades between the two mics. If it’s fully to the left you’ll get the sound from the left mic, and if it’s set to the right you’ll get the sound from the rightmost mic. If the **Stereo Width** knob is set at MONO, this is exactly like an ordinary cross fader. Set it as in the picture above, and you’ll get about 75% of the left mic (MIC ONE) and 25% of the right mic (MIC TWO).

Stereo Width This knob alters the stereo width of the signal.

Phase Invert This switch inverts the phase of MIC TWO.

Example Settings

One Mic

Mic Balance=0 (LEFT)

Stereo Width=anything

Only the left mic is selected, and since we only have one source (the left mic) the **Stereo Width** knob won't do anything and we will have the same output in both the left and right channel.

Both Mics, Mono Output

Mic Balance=0.5 (MIDDLE)

Stereo Width=0 (MONO)

With the **Mic Balance** in the middle, we have an equal amount of both mics, and since the **Stereo Width** is MONO, both the left and the right channel will sound the same. Both channels will have 50% of the left mic and 50% of the right mic.

Full Separation

Mic Balance=0.5 (MIDDLE)

Stereo Width=1 (STEREO)

We're in full stereo and equal amount of both mics. The left mic (MIC ONE) will only be heard in the left channel, and the right mic (MIC TWO) in the right channel.

Half and Half

Mic Balance = 0.5 (MIDDLE)

Stereo Width = 0.5 (halfway between STEREO and MONO)

Equal amount of both mics, but since the **Stereo Width** control isn't fully in stereo, we will have some bleed between the two channels. The left channel will contain about 75% of the left mic and 25% of the right mic, and vice versa with the right channel.

Mono and Stereo Operation

Metal Amp Room is designed to be a MONO-TO-STEREO plug-in. One guitar input, two mic outputs.

For best results, and if possible, insert the plug-in as MONO-TO-STEREO or a STEREO. If it has been inserted as a stereo plug-in, it will sum the two inputs (left and right) before processing the audio.

Using Pan and Balance

Be careful when you use the balance control of the stereo track. If you for instance have a great stereo sound using both the left and right mic, and then balances the track using the balance control in your host, you will change the sound of the output. For instance, if you want the output from Metal Amp Room to be fully panned to the right, set the **Stereo Width** control to MONO before you balance it to a side. Rule of thumb is that if your balance the stereo track halfway to one side, the **Stereo Width** control should be set halfway between MONO and STEREO.

If you balance the stereo output using the mixer in your DAW, some stereo information will get lost. Use the **Stereo Width** control to compensate for that. If you balance the track fully towards one side, set **Stereo Width** = MONO. If you balance the track halfway towards one side, set **Stereo Width** half-way between STEREO and MONO.

Bypassing Amps or Cabs

You can choose to bypass the amp or the cabinet by selecting **AMP BYPASS** or **CAB BYPASS** from the small box in the lower right corner.

This is very useful if you want to use Metal Amp Room with an external speaker (or speaker plugin) or if you just want to add a cabinet simulation to a track, such as a recorded line out from your amplifier.

By bypassing the cabinets in Metal Amp Room, you can use the cabinets from the other Amp Room plugins. Just insert, for example, Bass Amp Room (with bypassed amplifier) on the track after Metal Amp Room (with a bypassed cabinet).

Buying Recommendations

If you like the sound from Metal Amp Room and would like to get that sound using the real deal (let's face it, a real amp is always sexier than a plugin), here are some buying recommendations.

Amplifier

The sound of the amplifier is very much inspired by the one and only metal amp, the Engl Powerball. To get a nice rhythm channel, we utilized our soldering gear and bred our own JCM800/Powerball mutant.

Metal Cabinet

If the Metal Cabinet is your thing, run out and get a slanted Engl E 412 V 4x12 cabinet for live gigs.

Black Cabinet

For that classic vintage sound, we recommend a brand new straight Marshall 4x12 cabinet.

Room

The room is a very important part of the sound, and we had the opportunity of visiting The Haunted while they were laying down tracks for their new album in the In Flames studio (previously known as Studio Fredman). Tue Madsen and Patrik Jensen made sure that all the cabs and the microphones sounded just the way they should. You cannot get more metal than that.

ENGL IS A REGISTERED TRADEMARK OF BEATE AUS-FLUG AND EDMUND ENGL. MARSHALL IS A REGISTERED TRADEMARK OF MARSHALL AMPLIFICATION PLC.

Gate and Preamp

The noise gate and balancing preamp was developed by Softube, and has not yet had the chance to be embodied in a real piece of gear.

Credits

Niklas Odelholm – modeling and sound design. **Oscar Öberg** – concept, preamp modeling, sound and DSP-programming. **Arvid Rosén** – power amp modeling and sound programming. **Torsten Gatu** – interface and DSP programming. **Ulf Ekelöf** – graphics programming and 3D rendering. **Patrik Jensen** – amp selection and sound design. **Tue Madsen** – sound design and cabinet measurements.



20 Mutronics Mutator

THE MUTRONICS MUTATOR WAS AN analog hardware unit that gained a lot of popularity among musicians, music producers and recording engineers in the 1990s and early 2000s. It can clearly be heard on records that had a big part in defining that era, with artists such as **Radiohead**, **U2**, **Nine Inch Nails**, **Depeche Mode**, **Chemical Brothers** and **Daft Punk** among its users. Production of the hardware ceased in 2007 since a vital component was discontinued by the component manufacturer, and it has only been available in the second hand market since then.

In the development work with the Mutator plug-in, Softube has worked closely with Mutronics to ensure that every little nuance of the original unit is faithfully captured. But we also made a few modern additions that makes the plug-in even more useful.

Overview

The Mutator is a stereo analog filter and envelope follower with full control facilities. It contains two independent voltage controlled filters, similar to those found in analog synthesizers, which can be used to treat any external sound source. Each filter can be controlled from its own associated low frequency oscillator (LFO) and/or its own envelope follower section. The envelope follower essentially controls the filter and/or volume based on the amplitude of the incoming (internal) sound source, or on an external sound source that is fed into the unit's sidechain. This external signal can be any audio

source, eg. a drum sound, a guitar, a synth, a sampler output, or even a microphone. Thus the envelope characteristics of one sound can be superimposed onto the filtering contour of another.

Another switch on the envelope follower selects envelope follow mode or gate mode—in which the circuit detects whether a signal is above a certain threshold level before switching on.

In gate mode, the device can operate much like a dedicated noise gate unit, with the added capabilities of the analog filters. The LFOs have four different sweep waveforms each and may be switched via a stereo link to produce stereo panning effects.

The Mutator has two channels. The upper row of buttons control the left channel, the second row controls the right channel. It is possible to synchronize the LFO of the left to the right channel, as well as to the DAW host's song tempo.

The lowest row of knobs are additions Softube made to the plug-in, that were not part of the analog hardware. These include tempo syncing to the DAW host and a Mix button that lets the user blend the original sound with the mutated.

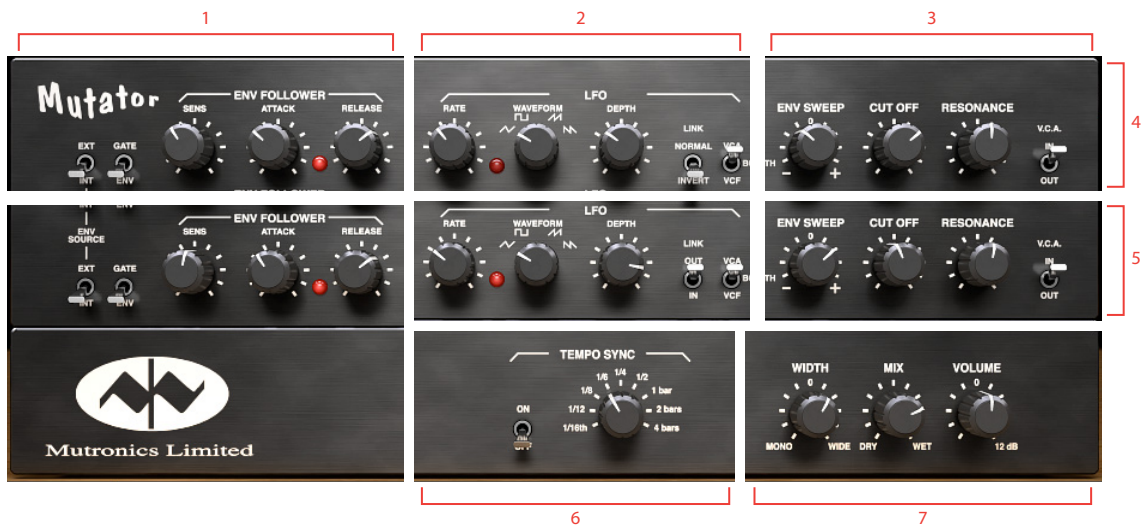
Knobs and Switches

The front panel controls for the Mutator are identical for both channels, with the exception of the **Link** switch which is detailed below.

Envelope Follower

Env Source Selects which signal drives the envelope follower circuit. In **INT** (internal) position, the sound source is also used as the control signal for the envelope follower. So with the **Env Source** switch set to **INT**, the filter section reacts to the amplitude of the sound source itself. In **EXT** (external) position, the envelope curve is determined by the incoming sidechain signal.

Env/Gate In **ENV** (envelope) mode, the envelope follower acts as its name implies, precisely following the amplitude of the control signal and producing a rapidly varying control signal which can be used to sweep the cut off frequency of the filter and/or volume level of the VCA. In **GATE** mode, the envelope follower is either full on or completely off, depending on whether the control signal is above or below the gate threshold as set by the sensitivity control.



- | | |
|----------------------|-------------------|
| 1. Envelope follower | 5. Right channel |
| 2. LFO | 6. LFO tempo sync |
| 3. Filter section | 7. Output section |
| 4. Left channel | |

Sensitivity In Env mode, this sets the optimum drive level for the envelope follower circuit. Too low and the **Envelope LED** only glows faintly, too high and the LED is driven hard. The optimum setting is when the LED peaks in brightness at the loudest points of the control signal. In GATE mode the sensitivity control sets the threshold level at which the gate operates.

Attack This adds an attack time (fade-in) at the front of the envelope. When set to ZERO (fully anti-clockwise) it has no effect on the envelope contour—fully clockwise it gives up to two seconds of fade-in.

Envelope LED The brightness of this follows the envelope contour produced by the envelope follower. It is useful for setting the sensitivity control and visually displays the effects of changing the **Attack** and **Release** controls.

Release This adds a variable release “tail” to the envelope. Even after the control input signal has died away, up to five seconds of fade-out may be added by this control. Setting this control to small amounts can be useful for smoothing out rapidly changing envelopes when in ENV mode.



1. Env source
2. Sensitivity
3. Attack
4. Release
5. Env/Gate
6. Envelope LED

LFO (Low Frequency Oscillator)

Rate This changes the sweep rate of the low frequency oscillator, from about one cycle every ten seconds to 100 cycles per second. Note that when the **Link** switch is on, only the **Rate** control of the left (upper) channel has any effect as it is being used to control both channels.

Rate LED A bright LED indicates that the LFO is at the peak of the waveform (high pitch, if it controls the frequency, or high volume if it controls the VCA).

Waveform Selects between the four available low frequency oscillator waveforms—**TRIANGLE**, **SQUARE**, **RAMP UP** and **RAMP DOWN**. Note that when the **Link** switch is on, only the **Waveform** control of the left (upper) channel has any effect as it is being used to control both channels.

LFO Depth Determines the amount of LFO sweep sent to the filter and/or voltage controlled amplifier.

Link Normal/Invert *Only on the left channel!* Via the **Link In/Out** switch, the LFO of the left channel can be sent to the bottom channel (right). The **Link Normal/Invert** switch inverts the output of the left LFO, which is being sent to the right channel. This means that every time the left channel LFO is at its peak, the right channel's LFO is at its lowest phase.

Link In/Out *Only on the right channel!* When set to **IN**, The left channel LFO is sent to both channels (for stereo effects). This means that the **Rate** and **Waveform** settings of the left channel affect both channels. When set to **OUT**, the LFOs affect their respective channel. Please note that all other settings are still active for each channel, **Link In/Out** only links the two LFOs.

VCA/Both/VCF Selects whether the output of the LFO sweeps the voltage-controlled amplifier, the voltage-controlled filter, or both simultaneously.

Filter Section

Env Sweep Determines the amount of filter frequency sweep (up or down) being sent to the voltage controlled filter from the envelope follower.

Cut Off Frequency This sets the centre roll-off frequency of the voltage-controlled filter.

Resonance Sets the Resonance (or Q) of the VCF. High values produce self-oscillation.

VCA In/Out Selects whether the final output goes through the voltage-controlled amplifier or is taken from the VCF output.



- | | |
|-----------------------|----------------------|
| 1. Rate | 6. Env sweep |
| 2. Waveform | 7. Cut off frequency |
| 3. LFO depth | 8. Resonance |
| 4. Link Normal/Invert | 9. Rate LED |
| 5. VCA/Both/VCF | 10. Link Out/In |
| | 11. VCA In/Out |

Tempo Sync

Tempo Sync On/Off

Setting **Tempo Sync** to ON means that the LFO rate of both channels are synchronized to the DAW host's tempo. With this set to ON, the **Rate** knobs of both channels are deactivated.

Tempo Sync

Allows the user to set the speed of the LFO in sub divisions of the DAW host's tempo.

1. Tempo sync On/Off
2. Tempo sync
3. Width
4. Dry/Wet mix
5. Volume

Output Section

Width This is essentially a Mid/Side matrix. Set to 0 (noon), the channels are panned hard left and right, respectively. When turned counter clockwise, the channels are both panned gradually towards the center. Both channels reach the center when the knob is turned fully counter clockwise. Turning it clockwise increases the apparent spread of the channels, making them sound as if they are wider than the actual loudspeaker placement.

Dry/Wet Mix Turned fully clockwise, you will hear only the sound of the Mutator itself, which is how the original hardware worked. By turning the knob counter-clockwise, you can blend in the original, unaffected sound source to your liking.

Volume Sets the overall output volume of the Mutator plug-in.



Operation of the Mutator

Even though the Mutator has a lot of knobs and switches, it's really very easy to operate. Basic idea is this:

What do we want to change? Amplitude and/or cut-off frequency?

We can change the amplitude by enabling the *VCA* (voltage controlled amplifier), or change the cut-off frequency of the filter by enabling the *VCF* (voltage controlled filter).

How do we want to change it?

We can change it either by letting the “loudness” of the input signal determine the change, by using the *envelope follower*. Or we can let a low frequency waveform, the *LFO*, determine the change. Or both at the same time.

Envelope Follower

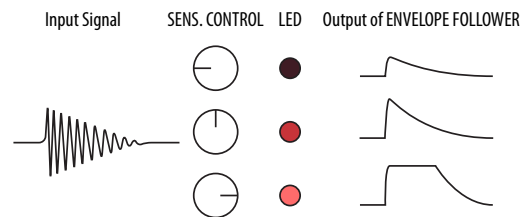
As previously mentioned, an envelope follower continuously tracks the amplitude of a signal and uses this envelope contour to control the cut off frequency of the filter. The signal that forms the base of this process is called the control signal. The control signal can either be the incoming signal itself (**Env Source** is set to *INT*) or an external signal that is fed to the Mutator using the sidechain functionality (**Env Source** is set to *EXT*). Extracting the countour from the internal signal could for example be used to create an auto-wah type of effect, where the louder parts of the sound source makes the cut off frequency of the filter move upwards.

A use case for the *EXT* setting would be to insert the Mutator on a piano track, and have the filter envelope on the piano be controlled by a drum loop on another track. In this case, the filtering on the piano would occur in time with the drum loop. Please refer to your DAW manual for instructions on how to set up a plug-in to accept an incoming sidechain signal.

Setting Sensitivity Control

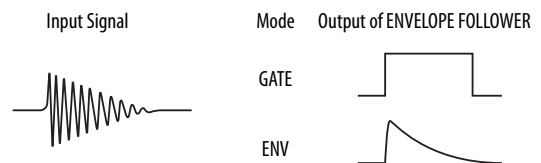
If the mode switch is in *ENV* position (and **Attack** and **Release** set to 0), turning up the **Sens** control will cause the LED to light when a signal is present—its brightness following the loudness of the signal.

For optimum performance, set the **Sens** control so that the LED reaches maximum brightness when the signal appears loudest. You may have to back the dial off slightly counter clockwise, as driving the circuit too hard will cause the LED to stay on maximum brightness even if the signal strength isn't quite at maximum.



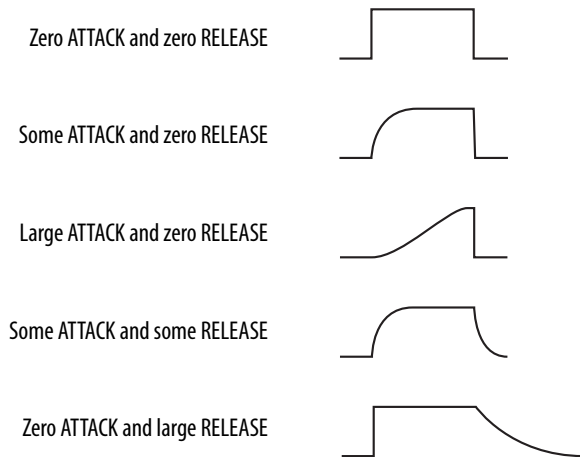
Gate Mode

If the mode switch is now flicked to the *GATE* position, the LED will come on at maximum brightness when a signal is present and be completely off when the signal falls low. Thus the output of the envelope follower will look like the diagrams below depending on the switch position.



Attack and Release in GATE Mode

In **GATE** mode, increasing the **Attack** and/or **Release** controls will change the shape of the output as shown below.



As you can see, a versatile range of envelope shapes may be constructed from the original gate shape.

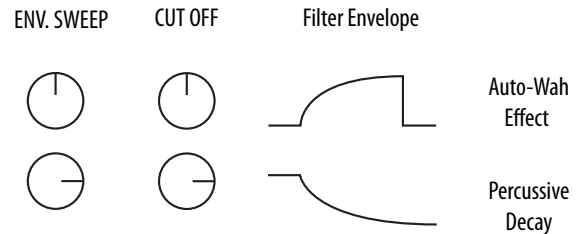
Attack and Release in ENV Mode

In **ENV** mode, the **Attack** and **Release** controls work as described above, but super-impose their contours on top of the original envelope shape of the signal. This can be very useful for smoothing out fast “wobblers” from a complex envelope!

Filter Sweep

The output of the envelope follower section is sent to its voltage controlled filter via the envelope sweep control in the filter section. This control has a centre zero (at noon) with both positive and negative sweep amounts available. Thus the arriving envelope contour can sweep the filter cut off frequency up or down from its original setting—which is determined by the **Cut Off** knob.

The same incoming envelope could produce entirely different effects on the sound depending on the settings of the **Env. Sweep** and **Cut Off** controls, for example:



If the **Env Sweep** control is set to 0, the filter cut off frequency will be manually changed by the **Cut Off** control.

The **Resonance** control allows a variable resonant peak to be added to the filter response, similar to that produced by an analogue synthesiser filter. At low settings the sound will appear fatter with subtle treble roll-off, much like a tone control. At higher settings a noticeable “twang” will be heard as the filter is swept, and at the highest settings the filter will begin to self-oscillate at its cut off frequency.

VCA In/Out Switch

After the filter stage, there is a voltage controlled amplifier (VCA) which, when switched in, controls the final output volume of that channel. The VCA is driven by the output of the envelope follower with the same envelope that is fed to the **Env Sweep** control. If an external source is selected for the envelope follower, and the gate mode is selected, and the VCA is switched in, the treated sound will be gated in and out by the external control signal.

The filter section can be effectively bypassed by turning **Cut Off** full up and its resonance to zero. Set this way, the VCF becomes “transparent” and with the VCA

switched in, the Mutator will function as a dedicated noise gate. If the cut off and resonance are altered, a filter sweep will combine with the gated effect.

The Low Frequency Oscillators (LFO)

The Mutator has two separate LFOs, one for each channel. The low frequency oscillators can automatically sweep the filter cut off (and/or VCA volume). Four waveforms are available: **TRIANGLE**, **SQUARE**, **RAMP UP** and **RAMP DOWN**. The sweep speed is controlled by the **Rate** control and the sweep range is adjusted by the **Depth** control.

The LFO output may be sent to the VCF cut off, the VCA level, or to both depending on the position of the **VCA/Both/VCF** switch.

Linking the LFOs

The LFOs of the two channels can be linked, in order to make them work in tandem. Note that the **Link** switches on the two channels are labeled differently and are used for different settings. The **Link In/Out** switch on the right channel (the second row of knobs) activates or deactivates linking. When activated, in the **IN** position, the LFO of the left channel affects both channels simultaneously. So with this setting, the LFO Rate and LFO Waveform of the right channel are deactivated. With this **Link** switch set to **OUT**, both channels' LFOs operate independently of each other.

The **Link Normal/Invert** switch on the left channel (upper row of knobs) can be switched between **NORMAL** and **INVERTED**. In **NORMAL** mode, the LFO affects both channels in exactly the same way. In **INVERTED** mode, the LFO's sweep is inverted for the right channel, essentially turning the waveform upside down. So every time the waveform is peaking on the left channel, it will be at its lowest level on the right channel, and vice versa.

Tempo Syncing the LFO

By turning on the **Tempo Sync On/Off** switch, the LFO rate controls are overridden by the DAW's tempo information and the Rate is no longer doing anything. By switching the **Tempo Sync** knob you can change the full period of one waveform to become a 1/16th note, 1/8th note, quarter notes, and all the way up to 4 bars. For extreme settings (say 4 bars at 30 BPM) the waveforms of the LFO might change a bit.

Resetting Waveform

By hitting play, most DAW's will send a reset command to the plug-in which will make the LFO start from scratch. For example, if the **Tempo Sync** is set at **1 BAR**, you will get a different result if you start playback on the downbeat or in the middle of the bar.

Output Section

The output section consists of a **Volume** control, a **Stereo Width** and a **Dry/Wet Mix**. If the left and right channels coming from the Mutator are very different, and you want to reduce the stereo width, the natural way would be to set the **Stereo Width** to somewhere between **MONO** and **0**, but in many cases you can achieve a similar (but not exactly the same) effect by settings the **Stereo Width** more to the **WIDE** setting. The main difference between **MONO** and **WIDE** is that in both channels will output L+R in **MONO**, while in **WIDE** one L channel will output L-R, while the other channel outputs the negated signal R-L.

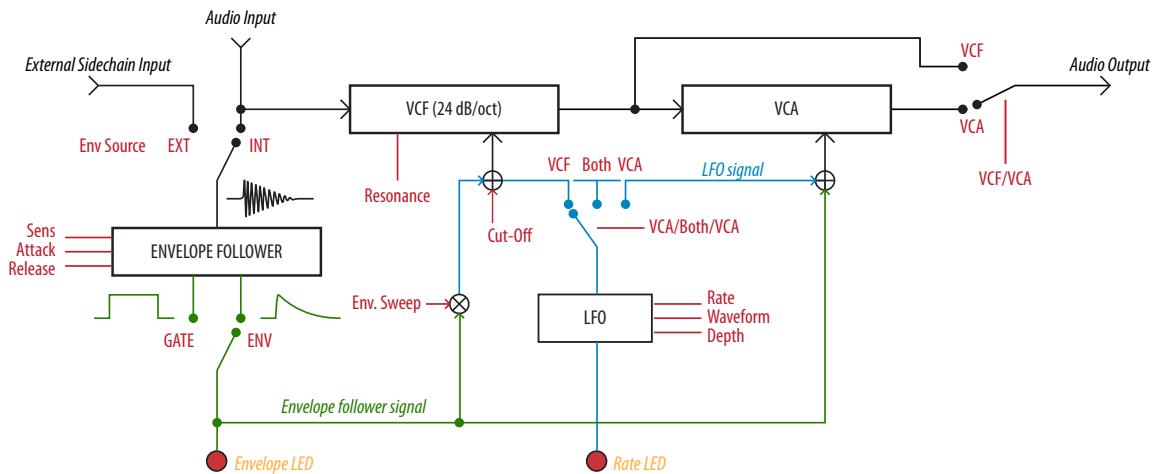
Block Diagram

The block diagram is a simplification of how it really works, but a useful help to figure out which knob to adjust.

One thing to note is that the Env Sweep control only affects the envelope follower, not the LFO.

Credits

Arvid Rosén – modeling. **Henrik Andersson Vogel** – manual, project management. **Niklas Odelholm** – graphics design, presets, math stuff. **Paul Shyrinskykh** – quality assurance. **Oscar Öberg** – math stuff. **Russ Hughes, Erik Putrycz** and **Paul Russell** – preset design. **Ulf Ekelöf** – graphics rendering. **Torsten Gatu** – framework programming. **Patrik Holmström** – framework programming. **Mattias Danielsson** – technical support.





21 OTO Biscuit 8-bit Effects

OTO BISCUIT CAME OUT IN 2010 and was the first product by French audio wizards **Denis Cazajoux** and **Stéphane Briat**. Denis had started out building stompboxes and synths for his brand *Cazatronics* already in the mid-2000s. One of the many boxes he built was the Biscuit, an effect housed in a plastic butter dish.

The original idea for the Biscuit was to simulate the sound of an old **Fairlight CMI** sampler. An 8-bit analog to digital converter was set up with a parallel output to an 8-bit digital to analog converter, and the sampling frequency could be set with a dedicated knob. A 12 dB per octave low-pass filter with a Q control was added to the prototype, as well as switches for muting and

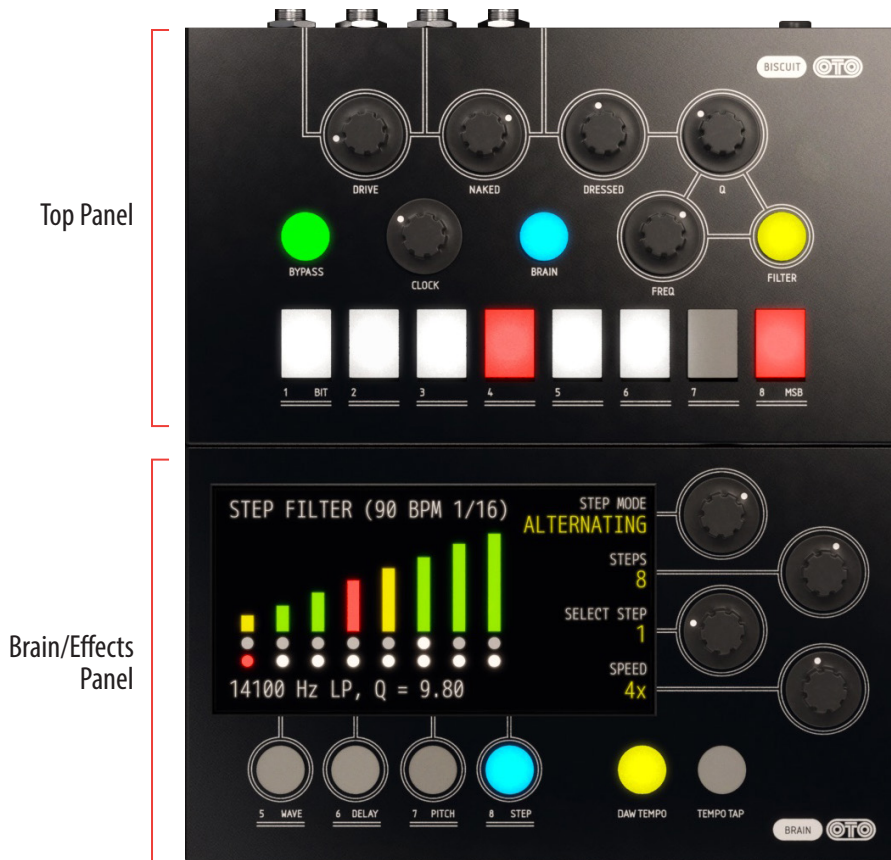
inverting the bits in the eight lines between the AD and DA converters.

But it wasn't until several years later, when Denis teamed up with producer/engineer Stéphane in a studio near Paris, that the two started expanding the Biscuit concept and turning it into a commercial product.

Overview

OTO Biscuit is a useful tool for all sorts of noise experiments. It's particularly useful with electronic sounds, but don't let that stop you from trying it on any sound source you can think of. At its core, the Biscuit is a bitcrusher, but some clever additions extend its use far beyond that.

It produces distortion by reducing the resolution and bandwidth of digital audio data. The resulting quantization noise can be varied from adding the familiar warm 8-bit character and lo-fi digital artifacts, to harsh numeric distortion. The "bitcrusher" section of the Biscuit is followed by an analog multimode filter (with low pass, high pass and band pass modes) and a simple mixing section (dry/wet). Furthermore, each of the eight data bits can be removed or inverted for more variation to the character, and there are four on-board digital effects—waveshaper, delay, pitch shifter and step filter.





1. Bypass
2. Drive
3. Clock
4. Naked
5. Brain
6. Dressed
7. Freq
8. Q
9. Filter
10. LSB (least significant bit)
11. Biscuiting switches
12. MSB (most significant bit)

Top Panel Controls

Drive Input gain knob. With a normalized signal gained above a certain level, the emulated diode clipping will be noticeable in the audio-chain and the **Bypass** button will flash red.

Naked Sets the amount of the dry signal mix, from silent up to 0 dB.

Dressed The amount wet signal mix, from silent up to 0 dB.

Q Resonance of the multi mode filter

Bypass This button bypasses **Biscuit when unlit**, green light means that Biscuit is active and processing. The button flashes red to indicate that the emulate drive circuit is clipping. It is usually desired to have some clipping happening.

Clock Sets the internal sample rate from 250 Hz up to 30 kHz. Changing the internal sample rate introduces additional gritty sonic artifacts as side-effects. This is a very recognizable effect that can be heard on many records.

Brain Toggles the Effects section on and off and thereby engages the last selected effect when lit.

Freq Cutoff frequency of the multi mode filter

Filter Sets the filter type. Its color indicates which type is selected:

GREEN: low pass

YELLOW: band pass

RED: high pass.

Biscuiting Switches

1-8 These switches mute or invert the 8 bits of audio stream through the Biscuit.

WHITE (lit) means that the **Bit** is ON

RED means that the **Bit** is INVERTED

GREY (unlit) indicates that the **Bit** is MUTED.

Inverting or muting bits of data can result in very harsh and loud noise, please be careful.

Inverting or muting the MSB (most significant bit), the bit furthest to the right, will result in more dramatic changes in sound, while doing the same to the LSB (least significant bit) yields more subtle results.

SHIFT-clicking on a switch will change all switches to the same value at once.

Brain/Effects Panel

Biscuit contains four different effects, of which one at a time can be used with the sample rate reduction, bit-crushing and analog filter:

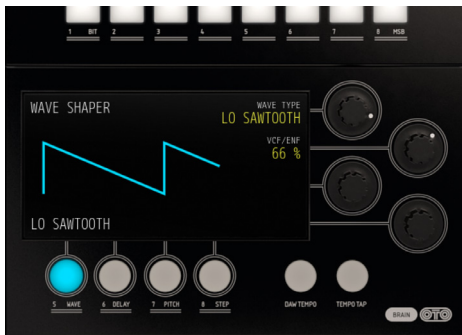
1. WAVE SHAPING
2. DELAY
3. PITCH SHIFTING
4. 8 STEP SEQUENCER FOR THE FILTER

Use the wave shapers to transform your sound, from crazy distortion to sub-octave effects. Add a gentle lo-fi delay or a screaming 100% feedback delay. Pitch shift

your melodies to add chorus or harmonies, or use the step sequencer to add DAW synchronized filter movement on your sounds.

The **Brain** button engages the last used effect when lit, and turns it off when unlit. The effect section can also be engaged by clicking on the desired effect button below the EFFECT DISPLAY.

Wave Shaper

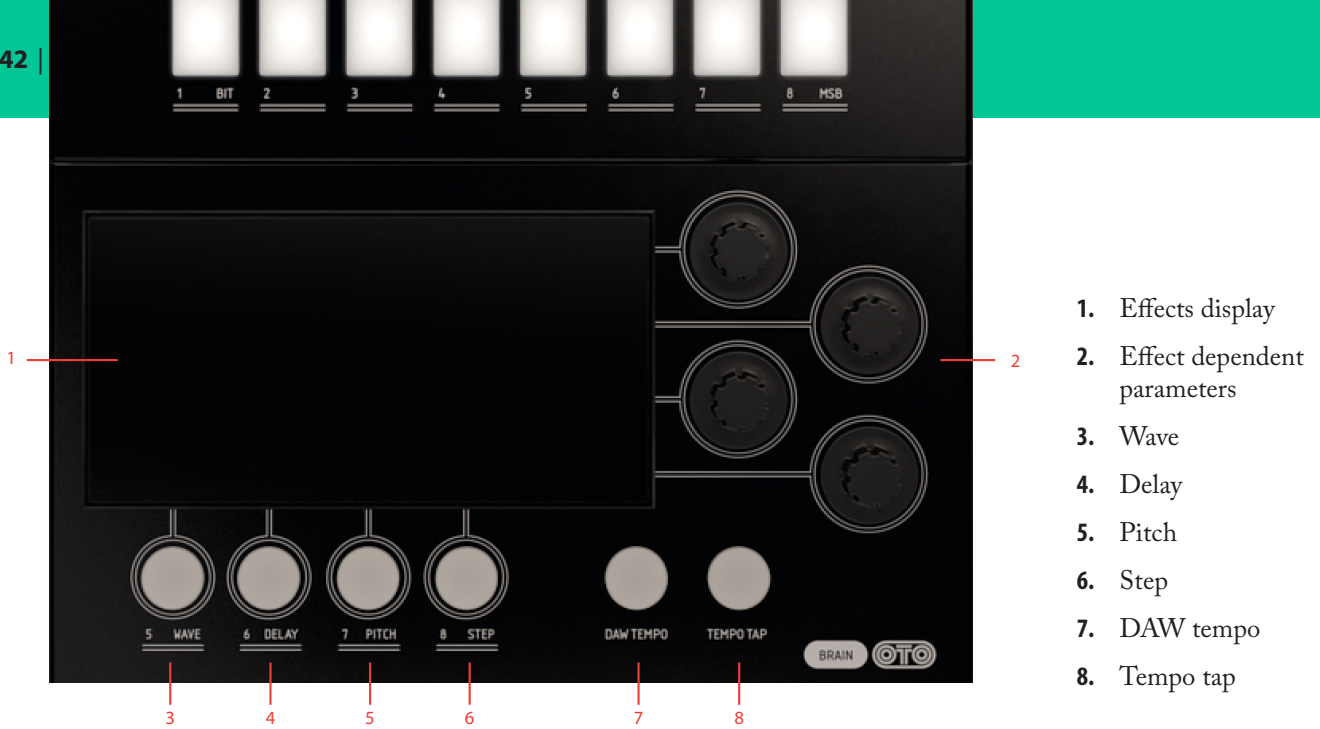


Delay

Pitch Shifter



Step Filter



1. Effects display
2. Effect dependent parameters
3. Wave
4. Delay
5. Pitch
6. Step
7. DAW tempo
8. Tempo tap

Effect Display Shows the user information on the currently active effect and its related parameters that are linked to the **EFFECT KNOBS** to the right of the display.

DAW Tempo Syncs the tempo to the DAW tempo for the **DELAY** and **STEP FILTER** effects.

Tap Tempo Tapping this button sets the tempo for the **DELAY** and **STEP FILTER** when **DAW Tempo** button is disabled.

Wave Turn on the Waveshaping mode

Delay Turn on the Delay mode

Pitch Turn on the Pitch Shifter

Step Turn on the Step Filter mode.

Each mode has one to four "soft parameters" that are specific for the different modes. For a description of those parameters, see the section corresponding to the effect mode.

Wave

The waveshaper effect can be used to distort, alter or dynamically track and filter your signal. Engage it by pressing **Wave**.

Wave Type Select the 8 different wave shapers. Shapers 1 to 5 will create distortion, shapers 6 to 8 synthesize a wave form.

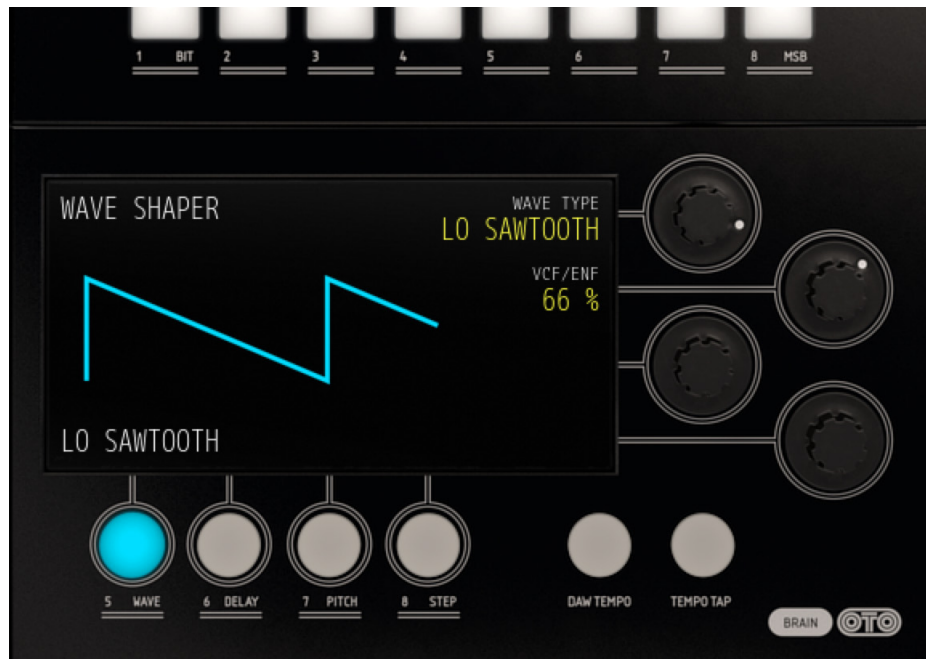
1. **RECTIFYING**: the negative parts of the incoming signal is inverted and added to the positive part. This creates an octave up effect similar to a fuzz octave pedal.
2. **ALTERNATE RECTIFYING**: only positive parts of the incoming signal is used. Creates a distorted "octave down" effect.

3. **BAT FUZZ**: the input above a threshold gets folded.
4. **BISCUIT**: "Fifth down" effect with some additional crunchy distortion.
5. **SWAP**: Bits 1 to 4 are swapped with bits 5 to 8 creating some radical distortion with a lot of high harmonics.

Waveshapers 6 to 8 in the Biscuit are small synthesizers, creating waveforms that track to the incoming signal frequency. These three waveshapers work best on bass synth sounds but can lead to exciting new sounds on other sources.

To allow proper tracking, crank up the drive knob properly so that the bypass button shows red to obtain a stable sound.

6. **SAWTOOTH**: Tracking SAWTOOTH waveform.



— Wave Type

— VCF/ENF

7. **LO SAWTOOTH:** Tracking **SAWTOOTH** waveform one octave down below source.
8. **LO SQUARE:** Tracking **SQUARE** waveform one octave down.

VCF/ENF Sets the amount of the envelop output that will affect the filter cutoff frequency. This effect is most apparent when lowering the cutoff frequency (**Freq** knob) on upper panel.

The **VCF/ENF** knob is only active in modes 6 to 8.

When using both the **Biscuiting** switches (bitcrusher) and **Wave Shapers** at the same time, it might be difficult to understand how these two effects interact. In order to have a better understanding of the **Wave Shapers**, we recommend to start using them with all the 8 bits in their normal state (all the 8 switches lit up white) at first.

1. Rectifying



2. Alternate Rectifying



3. Bat Fuzz



4. Biscuit



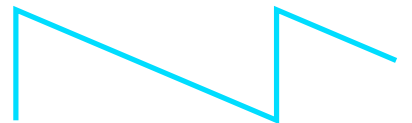
5. Swap



6. Saw Tooth (synthesized)

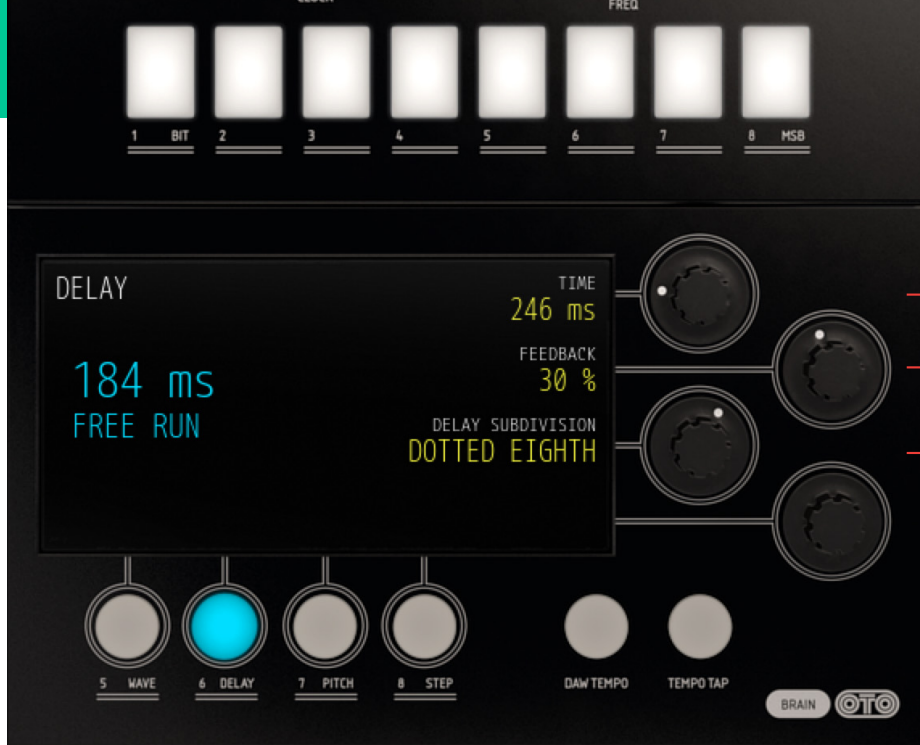


7. Lo Saw Tooth (synthesized)



8. Lo Square (synthesized)





1. Time
2. Feedback
3. Delay subdivision

Delay

The delay effect in Biscuit is a 8-bit stereo delay as opposed to the original hardware mono operation.

Only the delay will be heard via the **Dressed** Knob. This means that, if you want to hear both the source signal and the delayed echo at the same time, turn up the **Naked** and **Dressed** volume knobs accordingly.

Sample rate reduction and filtering will only apply to the delayed signal in **Delay** mode.

Time Sets the delay time of a quarter-note in free running mode. It also controls the time of the **STEP FILTER** (see below) free running mode which is using the same internal clock.

Time knob is disabled in **DAW SYNC** mode (as delay time is set automatically when **DAW Tempo** is lit).

Feedback Sets the amount of feedback, from 0% (one echo) to 100% (infinite build).

Delay Subdivision Sets the subdivision of the delay time into:

QUARTER NOTES (preset),
DOTTED QUARTER NOTES,
EIGHTH NOTES,
DOTTED EIGHTH NOTES,
SIXTEENTH NOTES, and
DOTTED SIXTEENTH NOTES.

The actual delay time is shown to the left in the effects display.

Pitch

The **PITCH SHIFTER** is a 8-bit mono effect that transposes the audio signal by using a short delay buffer and sends it to both left and right outputs via the **Dressed** knob.

Shift Sets the amount of pitch shift through a number of presets, ranging from -2 octaves up to +1 octave.

You can either use the knob, or click directly on the dots on the displayed keyboard to change pitch. Middle C is for creating chorus-like effects by mixing the **Naked** and **Dressed** signals. Click twice on the lowest C to alternate between -1 octave and -2 octaves pitch shift.



In the original hardware the sample rate in the pitch shifter is fixed at maximum rate to obtain as good pitch shift effect as possible.

Using both Biscuiting and sample rate reduction will make the pitch shifting algorithm work more like a pitch shifting delay, thus creating some cool effects not obtainable in the original hardware unit.

Step

The **STEP FILTER** is an (up to) eight steps filter sequencer where each step can be set to have its own **Filter Type**, cut off **Frequency** and **Q** value.

In **STEP FILTER** mode the effect display will show the **BPM** of the sequence related to division and/or **DAW sync** (if selected).

Each step in the sequence is shown in the display represented by a bar and two dots.

- The bar height represents the **Frequency** of the filter.
- The bar color represents the **Filter Type** (green for **LP**, yellow for **BP** and red for **HP**)
- The bar light intensity represents the **Q** value of the filter.
- The top row of dots indicates the "song position", ie. which step in the sequence that is currently playing.
- The bottom row of dots shows the currently selected step, the step that can be edited by adjusting the filter knobs. It also shows the total length of the sequence.

You can click directly on the display to select current step and change filter **Frequency**. To set **Filter Type** and **Q** value per step, you need to use the knobs on the upper panel.

A sequence can be 2, 3, 4, 6 or 8 steps long.

Step Mode Sets any of the four playback modes:

1. **FREEZE**: in this mode only the selected step is played. This is convenient for setting up the step's parameters (**Frequency**, **Q**, **Filter Type**) in detail.
2. **FORWARD**: the sequence is played from left to right.
3. **ALTERNATING**: "pendulum", sequence alternates between playing forward and backwards.
4. **RANDOM**: deterministic random, steps are played in random order.

Steps The amount of active steps of the sequence: 2, 3, 4, 6 or 8 steps long. Active steps are lit up and marked by white dots in the lower row beneath the display. Steps not active in the sequence are dimmed grey in the display.

Select Step Select the step to be edited. This can also be achieved by clicking in the display or on one of the rows of dots below the display.

Speed The speed division for each step in the filter step sequencer. Default is that each step is a quarter note which corresponds well to **BPM** free running.

When **DAW sync** is used division information (quarter, eighths, sixteenth and so on) is represented in the upper right corner in the effect display along with synced **BPM**.

Setting the Tempo in Free Running Mode

When the **DAW Tempo** is inactive (unlit), the **STEP FILTER** is in a free running mode using the same internal clock as the free running delay. This means that you can set the tempo either by tapping the **Tempo TAP** button or by adjusting the delay **Time** knob in the **DELAY** effect.

For an exact setting of tempo in the step filter, **DAW sync** is recommended.



1. Freq
2. Q
3. Filter
4. Step mode
5. Steps
6. Select step
7. Speed
8. Frequency
9. Select step
10. DAW tempo
11. Tempo tap

Linking Parameters

Some knobs and buttons have extra functionality (parameter linking) that can be accessed by shift clicking the knob/button.

Clock knob: change the clock speed and filter cutoff simultaneously.

Naked or **Dress** knobs: change both knobs at once with an inverse relationship to mirror that of a cross-fade operation.

Biscuit switches (bit inverter/mute switch): change all Biscuit switches to the same value at once.

Freq, Q or **Filter** in **STEP MODE**: set all steps to the same value. This can be useful for example when you want to go from a complex sequence with a lot of different use filter modes to a simpler one.

Tips

Mix the **Dressed** and **Naked** signals to achieve phased wave type sounds when sweeping cutoff and/or clock rate.

Inverting and removing bits will cause more noticeable effects the more significant the bits are (that is, the further right the buttons are).

When turning off or inverting the **MSB**, lowering the **Drive** level can act like a kind of a way to “gate” the signal and thereby dramatically alter the dynamics of the sound. Try it, you might like it!

The step sequencer can also be used for “gating” sounds, such as pad, in sync with the DAW. Use **LP** mode and set **Freq** on the desired “silent” steps to 0.

Using pitch shift and clock rate reduction at the same time can create some interesting results not possible in hardware.

By using clock rate reduction and **HIGH PASS** filtering of drums, nice additional hi-hat sounding parts can be created.

Using a combination of the **SWAP** wave shaper and the **HIGH PASS** filter mode can create nice additional high end “shakers” when applied to drum loops.

Credits

Björn Rödseth – modeling. **Kim Larsson** – modeling. **Arvid Rosén** – modeling. **Kristofer Ulfves** – product owner, user manual, testing and presets. **Niklas Odelholm** – graphic design. **Paul Shyrinskykh** – quality assurance. **Ulf Ekelöf** – 3D rendering.

22 Passive Equalizer

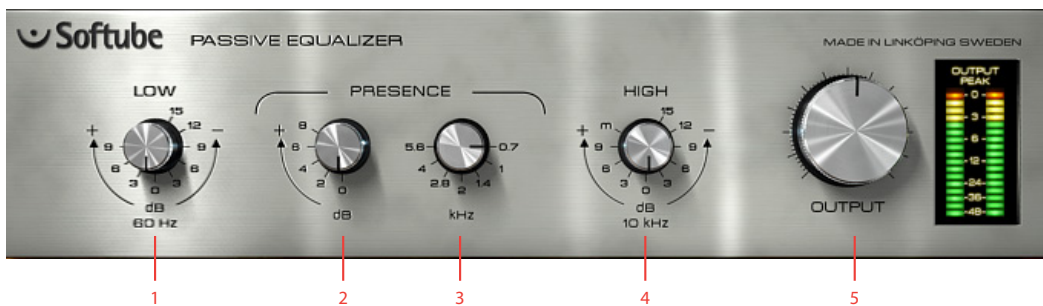
For more info about this product, please see the chapter about the Focusing Equalizer.

The Passive Equalizer, modeled from a Neumann PEV 930 console equalizer, has a very wide sound with a warm low end and a sparkling top. Excellent for adding that extra sparkle a track or a mix might need. The Mid set at 700Hz will give you a fat 70's sounding snare drum. Technically speaking, the High and Low filters are more of a shelving type than a bell filter, and the Mid filter is widest at low frequency settings, while getting more sharp for higher frequencies. Since this is a passive circuit, all knobs will interfere with each other and changing something in the treble might do something in the bass or mid as well.

The smooth and sparkling high boost filter is great for making up for a dull and lifeless track.

Knobs

Low Cut/Boost This knob will either cut or boost the low frequencies. The original unit is labeled as if it boosts or cuts with 3dB steps at 60 Hz, but in reality it changes the frequency of a shelving filter so that the attenuation/gain at 60 Hz will change. It would be more correct to label it with a fixed gain and a variable frequency, but we decided to leave it as it was on the original unit. And due to the non-ideal components in the unit, it does more things than that but that's too technical to put in a user's guide...



1. Low gain
2. Mid boost
3. Mid frequency
4. High gain
5. Output volume

Presence (Gain

Control) Use this to set the gain of the presence circuit in 2 dB steps (from 0 to +8 dB)

Presence (Frequency

Selector) Set the center frequency of the presence circuit (0.7, 1, 1.4, 2, 2.8, 4 and 5.6 kHz). The center frequency and gain have a tendency to move if you increase the **High** knob...

High Cut/Boost

Boost or cut the high frequencies. Just like the **Low** knob, this is more of a shelving type filter than a bell type. Boosting the **High** will change the character of the **Presence**.

Output

Output volume. Goes from nothing to +12 dB. At the 12 o'clock position, it's set at 0 dB, and going clock wise will increase the output volume by +2 dB for each mark.

The Presence filter is widest at low frequency settings (0.7 kHz) while getting sharper at higher settings (5.6 kHz). The 0.7 kHz setting is excellent for getting a fat snare drum sound.

Buying Recommendations

If you like the sound of the Passive Equalizer and would like to get your hands on the real stuff (which is an excellently built and beautifully crafted unit), you should look for a Neumann PEV 930-00 console equalizer. It's a fantastic sounding unit.

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Credits

Niklas Odelholm – modeling and graphics design.
Torsten Gatu – concept. **Oscar Öberg** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – graphics rendering. Thanks to **Stefan Fandén** and the crew at Deluxe Music for letting us borrow the gear!

23 Spring Reverb

Introduction

THE SPRING REVERB WAS AN essential part of almost all albums produced before the age of digital processing. With a sound that doesn't sound at all like a real-life reverberation, it has made its way into the music production hall of fame just because of its peculiar character. There is really no reverb that sounds like a spring reverb, and if you want to get a vintage vibe on your production, that's about the only way to go.

The Spring Reverb plugin can for example be used as an insert effect together with the Brown amp from Vintage Amp Room in order to get a really bluesy sound, or as a send effect on vocal tracks to get that vintage vibe.

The Springs and Tension parameters let you morph between a typical guitar amp reverb and a smoother studio reverb, thus making the Spring Reverb plugin a very versatile tool in your studio rack.

As with all our products, Spring Reverb is the result of exhaustive research. It was developed after nearly a year's analysis of the mathematics and mechanics of spring reverbs. New simulation techniques had to

be developed in order to simulate the springy characteristics of the reverb, and we're proud to say that we found what makes a spring sound springy. The user interface was designed to be as simple as possible, with a small set of parameters that make a difference. The Tension and Springs parameters are additions that we didn't find on any real reverb, but that we just couldn't live without.





1. Mix (Dry/Wet)
2. Bass
3. Treble
4. Springs
5. Tension
6. Shake!

User Interface

The Spring Reverb has the following controls: **Mix**, **Springs**, **Tension**, **Shake**, **Bass**, and **Treble**. The following pages will give you a brief description of all the controls.

Mix Controls the balance between the reverb and the direct signal. Outputs only the original signal when set to **DRY**, and only the reverb effect when set to **WET**.

Set this control to blend the wanted amount of reverb with the dry signal. Make sure that you set **Mix** fully clockwise on **WET** every time you use Spring Reverb as a send effect.

Springs Controls the number of springs in use. There are three sets of springs in the reverb unit and you can choose how many of the sets you would like to use.

Set the control to **ONE** for a more pronounced springy sound and to **THREE** for a smoother sound.

Together with the **Tension** slider, this control is an excellent tool to change the overall character of the reverb.

With **Springs** set to **TWO** and **Tension** in the left most position, you will get a typical Accutronics® Type 4 sound, as used in many American guitar amps.

Shake BOOOM! Since we don't want you to smack your computer to get that thunderous springs-being-shaken sound, Spring Reverb features a slider that can be used to accomplish the same thing safely. Fully automatable of course.

Be aware that when the slider is moved to a position other than halfway between far left and right, the reverb will go "spoing", so storing presets with this knob set to something other than in the middle, it will produce a (perhaps unwanted) noise when the preset is loaded. For this and other reasons, when the slider is adjusted in the GUI it will spring (ha!) back to the middle when the mouse button is released. This does not work when the parameter is being controlled by a MIDI surface or other controller.

The **Bass** and **Treble** controls do not affect the sound of the shake effect, since they are applied before the signal goes into the reverb tank.

For the most violent shake effect: set **Tension** to the left and **Springs** on three before you use the **Shake** parameter.

Tension Tunes up or down the tension off all three springs. Adjusting the **Tension** parameter will change the sound of the strings from slow and smooth (left) to quick and harsh (right). The idea behind this parameter is that you are able to

get the sounds from many different reverb tanks by just changing the tension and the number of springs. A cheap guitar reverb tank will usually have a high tension and two springs, while a more expensive reverb tank might have three springs and a lower tension.

Bass and Treble Controls equalization of the wet signal. In general you would need to lower the **Treble** if you use the reverb on percussive sounds, while too much **Bass** will make the reverb sound very dark and take up a lot of energy in the mix.

Tweaking Tips

There are a couple rules of thumb to achieving the sound you desire with Spring Reverb.

Mix

Mix is without doubt the setting that will have the biggest impact on the overall sound you achieve and probably the one that is hardest to master.

First of all, when used as an insert effect, the **Mix** knob will typically be set between 0 and 20%. It is easy to drench your recording with a reverb that is as characteristic as Spring Reverb.

When used as a send effect, the **Mix** knob should be set to 100%, but the send/return volumes of your DAW should be kept low. Easy does it!

Controlling the Character

In terms of character, Spring Reverb goes from vintage grit, full of character, to semi smooth, still with character but toned down.

The grittier sounds are for example great with electric guitars, electric pianos, and vocals. Setting the **Springs** knob low and **Treble** high will bring out the spring characteristic, producing more pronounced echoes. Then **Bass** can be used to shape the overall frequency characteristic and **Tension** to control the length of time it takes for the reverb to stop ringing.

For more percussive sounds, such as drums or a full mix, the character will probably need a bit of toning down. Bring the **Springs** knob all the way up to let the echoes of the three springs intermingle and create a more solid reverb tail. Then reduce **Treble** to decrease the metallic character and finally do final tweaks with **Tension** and **Bass**.

Using Spring Reverb with Vintage Amp Room

Spring Reverb is an excellent companion to the Brown amp in Vintage Amp Room. In the real amplifier, the spring reverb is placed in between the preamp and power amp. With the Spring Reverb plugin, you can use it both as an insert *before* Vintage Amp Room and as an insert *after* Amp Room and get very different results.

Placing Spring Reverb before Vintage Amp Room

If you place Spring Reverb before Vintage Amp Room, the reverb effect will be distorted in the same way as the guitar signal, and you will usually need to lower the **Mix** knob and use a bit less **Treble** than if you place it after the plugin. This placement will often make the reverb sound more authentic, although it is a bit harder to dial in.

Placing Spring Reverb after the Vintage Amp Room

This is the “normal” way to use Spring Reverb. All presets have been tailored to suit this use. Just place Spring Reverb (preset: “GUITAR DIRTY”) after Vintage Amp Room (preset: “BROWN BLUES”) and you’ll get the most authentic blues sound available from a plug-in. Don’t forget to use a beaten up ol’ guitar with single coil mics.

Chorus-y Effect

It is very easy to get a very special chorus effect if you automate the **Tension** knob. Since many hosts (such as Cubase and Pro Tools) have the ability to write in automation curves that are sinusoidal or triangle shaped, it is very easy to get an unique sounding chorus effect from Spring Reverb.

Credits

Peter Möller – research and mathematical modeling.
Torkel Svensson – research and mechanical analysis.
Oscar Öberg – modeling and implementation. **Torsten Gatu** – framework programming. **Niklas Odelholm** – framework programming and graphic design. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering and graphics.

24 Summit Audio EQF-100 Full Range Equalizer

Introduction

IT'S ALWAYS DIFFICULT TO DESCRIBE how an equalizer sounds, but everyone we've asked has described the EQF-100 in the almost the same way, with *rich*, *creamy*, and *smooth* being the most common adjectives. And there is definitely some sort of magic in this equalizer.

With a wide bandwidth setting you'll experience an open sound that will breathe life into your tracks, while a narrower bandwidth gives you a no compromise solution for precision tone shaping.

The four full bands of equalization plus the low and high cut gives you plenty of flexibility for any track or style.

Add a tube stage amplification to perfect the rich tone that comes from the passive EQ design and you'll get a warm and fat distortion when the output VU meter hits the red.

Added Functionality

To make an already sexy equalizer even sexier, we decided to add two beautiful VU meters to the unit. We also added the **Output Volume**, so that you easily can level match the equalizer.

A Note on Distortion

As usual we have included all the distortion that the original unit has, with most of the distortion occurring at the output of the EQ. We've calibrated the distortion so that it will start to saturate when the output VU meter goes up above 0 into the red area. If you do a lot of boosting with the EQ bands and get too much distortion, you can always lower the output volume in order to lower the distortion. The distortion will kick in at around -3 to 0 dBFS.

The sound of the distortion in this unit is very thick and *woody*, as opposed to thin and metallic, which in technical terms means that it contains a strong 2nd order harmonic. For example, the distortion in the TLA-100 compressor has a different character, with more edge than the EQF-100.

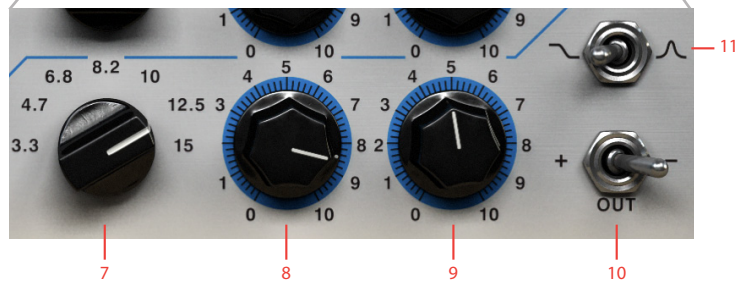


User Interface

The user interface consists of four separate bands of equalization (low, low mid, high mid, and high), two cut filters (high and low), the meter section and the **Output Volume**.



1. VU meter and output volume
2. High mid band
3. Low band
4. Low mid band
5. High band
6. Low and high cut
7. Hz, kHz frequency selector
8. Bandwidth
9. Gain
10. Boost/Cut
11. Type Shelving/Bell



Frequency Bands

Each of the four frequency bands consists of seven fixed frequencies with fully adjustable bandwidth (Q) and gain. All bands can be bypassed separately. The lowest and highest bands can also be set to shelving mode.

Hz, kHz	
Frequency selector	Selects one of seven frequencies on which the band boost/attenuate and bandwidth controls operate.
Bandwidth	Determines how much of the frequency spectrum is affected by the boost or cut operation. 0 is a narrow bandwidth and 10 is a wide bandwidth.
Gain	Determines the amount of boost or cut that takes place.
Boost/Cut	This switch sets the section for boost or cut; the center position turns the sections off.
Type	
Shelving/Bell	Changes the high or low band between a shelf or peak type of response curve. <i>This switch is only available for the highest and lowest bands!</i>

Low and High Cut

Each cut filters consists of three set frequencies and bypass.

Filter - Hz

Low Cut Determines the low frequency cut off.

Filter - Hz

High Cut Determines the high frequency cut off.

Output Section

Output Volume Adjusts the output volume (before the output distortion) from -30 to +10 dB.

If you want more distortion, just increase the **Output Volume** until the output VU meter hits the red area!

Applications

The following paragraphs are from the original hardware manual, but are of course just as valid for the plug-in as for the hardware. (Well, except for all that stuff about hooking up XLR cables.)

Basic setup

Plug the EQF-100 directly into the insert jack on the mixing console. Put the insert send into the XLR input, and the XLR output back into the insert return of the desired channel. If outboard preamps are being used, try sending the output of the preamp into the EQF-100. This EQ is also perfect before or after a compressor, and for side chain effects such as de-essing.

Boosting a frequency

Select which frequency you want to boost, put the Boost/Cut/Bypass switch to plus and increase the gain. Use the bandwidth control to affect more or less of the surrounding frequencies. A lower number on the bandwidth knob means a narrower bandwidth. If the frequency you want to boost is on the high or low band, set the Shelving/Resonant switch to resonant.

Setting up a low or high filter

Set all four Boost/Cut/Bypass switches to Out. Adjust the two filter settings to the desired frequencies.

Notch filtering

Find the approximate frequency to be attenuated (low, mid-low, mid-high, high). Set the bandwidth to wide (ten), the Boost/Cut/Bypass switch to Cut (minus), and the filter type to resonant (high and low bands only). Turn the gain all the way up (for the highest amount of cut) and start narrowing the bandwidth. Use the frequency select and bandwidth to narrow in on the frequency to be attenuated. Decrease the gain knob (decreasing the amount of cut) until the frequency is cut with the least effect on the surrounding frequencies.

Low or high shelving

Set the low or high EQ sections to shelving filter. Select the frequency where roll-off or boosting is to begin. Select plus or minus on the Boost/Cut/Bypass switch and turn up the gain knob to cut or boost above the high frequency selected or below the selected low frequency.

Mono and Stereo Operation

The Summit Audio EQF-100 plug-in can operate in both stereo and mono modes. In stereo both channels are linked, but processed individually. How the different modes (stereo, mono and mono-to-stereo) are selected depends on your host software.

Credits

Arvid Rosén – modeling, **Niklas Odelholm** – extra features and modeling, **Mattias Danielsson** – testing, **Torsten Gatu** – framework programming, and **Oscar Öberg** – framework programming, **Ulf Ekelöf** – 3D rendering.

25 Summit Audio Grand Channel

The Tone Sculpter

WE KNEW FROM THE OUTSET of our development of the TLA-100A compressor that we wanted to create a beautiful looking, extremely versatile and gorgeous sounding channel strip plug-in which would combine the TLA and the EQ beast known as the EQF-100. But the EQF-100 proved to be a more difficult creature than we had first imagined.

After having modeled plenty of equalizers we felt confident that absolutely nothing could surprise us and that we would keep all deadlines. We couldn't have been more wrong. The passive design of the EQF-100 was intricate, well-designed and just as beautiful as we thought it would be. We just didn't realize how complicated it was.

But after plenty of work developing a new technique for modeling of big passive circuits we feel confident that you will enjoy this monster.

EQF-100 Full Range Equalizer

Four full bands of equalization, two cut filters, two VU meters and a warm output distortion makes this equalizer extremely versatile. It can be used on any type of source, and once you get to know it, it will become your best friend.

TLA-100A Tube Leveling Amplifier

It doesn't have many knobs, but can create a vast amount of different sounds. With an opto-like compression



circuit, you'll get a smooth compression with a very soft knee. But due to its modern and intricate design you can set it to be faster than any other opto-compressor, which makes it more versatile than its competitors.

Different Kinds of Distortion

The TLA-100A has a separate saturation control which lets you adjust the headroom of the unit, so that you can go from a clean sound with almost no audible distortion at all, to a fully saturated leave-no-transients-intact sound. The distortion in the TLA is therefore well-suited for taming transients in drums, but maybe not the first choice when you need to warm up a vocal track.

The distortion of the EQF-100 has a totally different character. Boost the EQ, or crank up the output volume and you'll get a warm and fat type of distortion, due to its strong 2nd order harmonic, that is very well suited for vocals, bass or brass.

User Interface

The user interfaces of the individual units are the same as the individual plug-ins, so please see their respective chapter for more information.



EQF-100
Full Range Equalizer

TLA-100A Tube
Leveling Amplifier

TLA-100A Extra Panel

Bypass and Routing Panel

EQF-100 Full Range Equalizer

A fully parametric four band equalizer with two cut filters.

TLA-100A Tube Leveling Amplifier

The classic TLA-100 compressor with adjustable attack and release times and a smooth soft knee character.

TLA-100A Extra Panel

The additional panel adds modern features to a classic compressor. Detector low cut, parallel compression and an adjustable saturation.

Bypass and Routing Panel

The routing panel is used for engaging or bypassing the separate units, and also to select the processing order.

EQ Bypass Bypasses the EQF-100 equalizer. Right position: **BYPASSED**. Left position: **ENGAGED**.

Compressor Bypass Bypasses the TLA-100A compressor. Right position: **BYPASSED**. Left position: **ENGAGED**.



1. EQ bypass
2. Compressor bypass
3. Compressor before EQ

EQ Before Compressor

Compressor Before EQ Selects if the equalizer should process the sound before the compressor (default) or the other way around. Left position: EQ BEFORE COMPRESSOR. Right position: COMPRESSOR BEFORE EQ.

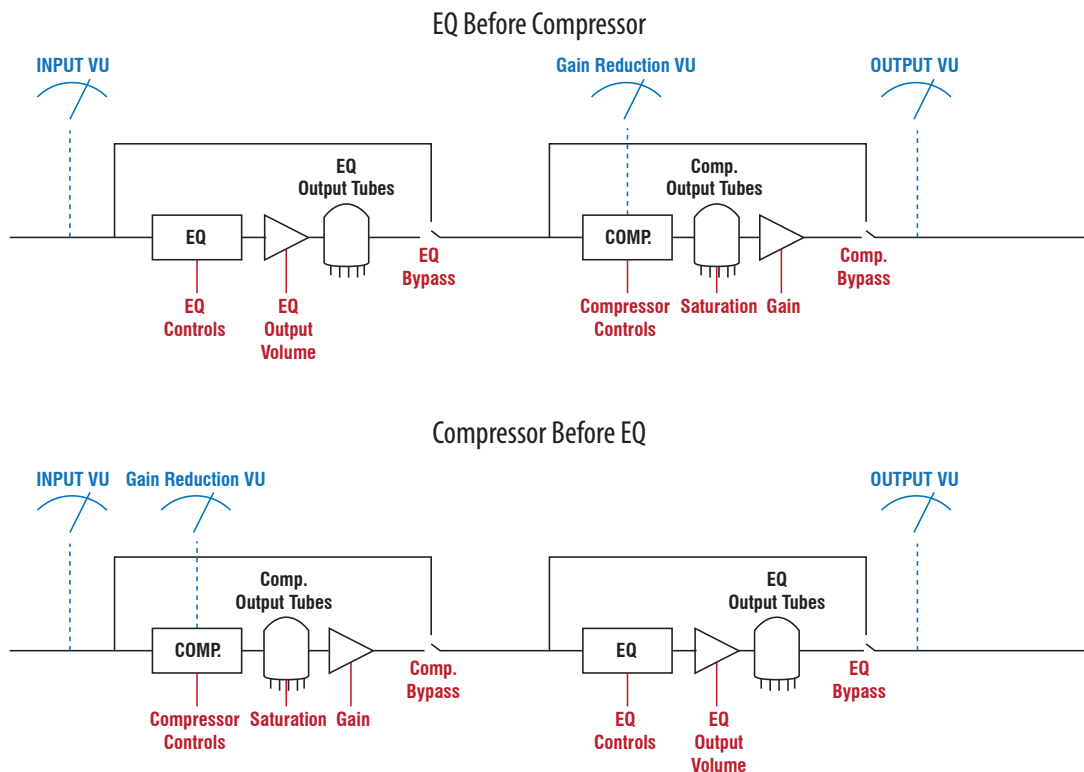
It is also possible to click on the pilot lamps of each unit to engage or bypass them.

Signal Routing

The signal routing is very straight forward, except for the VU meters on the EQF-100, which are always placed at the master input and the master output. See the block diagram for more information.

Gain Staging

There are two main volume controls on this unit, the Output Volume on EQF-100 and the Gain knob on the TLA-100A. The best way to use these volume controls is to make sure the apparent audio level stays approximately the same when you bypass the unit. That way it will be easier to bypass and compare the audio, and also to switch order of the compressor and equalizer.



VU Meter Calibration

Just like the stand-alone TLA-100A plug-in, the Grand Channel's VU meters are calibrated so that a -9dB_{RMS} signal reads 0 VU.

Sidechain and the Equalizer

No, the equalizer does not affect the external sidechain of the compressor.

Credits

Oscar Öberg – modeling, **Arvid Rosén** – modeling, **Niklas Odelholm** – modeling, **Mattias Danielsson** – testing, **Torsten Gatu** – framework programming, **Ulf Ekelöf** – 3D rendering.

26 Summit Audio TLA-100A Compressor

More Than a TLA-100A

THIS SUMMIT COMPRESSOR IS A CLASSIC, but we felt that there are a couple of features that could really bring this unit into the 21st century. So we decided to add an extra panel below the unit. If you don't care about fancy features, just disregard that panel and the TLA-100A will work just as the original unit.

Low Cut

It is always handy to have the ability to filter out the lowest frequencies so that the compressor doesn't react on them. Here you also have the possibility to choose whether you want to put the low cut on the entire signal (just as if you would have had a Low Cut inserted before the TLA-100 plug-in) or put on just the detector part of the compressor (a.k.a. "sidechain filtering")

Parallel Inject = Parallel Compression

Parallel compression, ie. mixing a compressed signal together with the original signal, is a standard practice today. So instead of using several mix buses to achieve this we added a knob.

Saturation = Headroom = Mix Level Adjust

It's always difficult to choose what level the virtual 0 VU should correspond to in the digital world. Some people use drum loops that peak at 0 dBFS, while others use material recorded at -24 dBFS. That's why we added the Saturation control. With that control you can tune the headroom and the amount of saturation from the tube



output stage in the compressor. Without it, you would have to use the Gain control to set the desired amount of distortion and compensate for that loss or increase in volume with a separate volume control. The Saturation does all that. In one knob.

User Interface

The user interface consists of two panels, the upper panel, containing the original TLA-100A knobs and switches, and the lower panel that offers extra functionality.



1. Attack Release Meter select
2. Gain (output volume)
3. Reduction/Output meter
4. Gain reduction
5. Sidechain

Upper Panel

Attack A three position switch with **FAST**, **MEDIUM** and **SLOW** settings, which corresponds to the time it takes the compressor to respond to the input signal.

Release Controls the time it takes the compressor to return to unity again. A three position switch with **FAST**, **MEDIUM** and **SLOW** settings. The release time is also effected by the program material. The slower the release time setting, the more the program material determines the release time.

Meter Select Allows monitoring the output level (**OUTPUT**) or the amount of gain reduction taking place (**REDUCTION**).

Gain Output volume. Unity gain is at “25”. The value display in the lower menu bar will show 0.0 dB.

Gain Reduction Sets the amount of gain reduction and the operating point where gain reduction begins. The higher the gain reduction, the higher the ratio becomes. Works almost like a threshold control.

Sidechain If you have an external sidechain present, switch to ext to use the external sidechain as target signal for the compressor. Use when you want a signal to duck under the external signal (for instance ducking strings under a bass drum, or music under a voice over).

Lower Panel

Low Cut Adds a low cut filter on either the detector or the input. From OFF to 600 Hz. OFF disables the low cut functionality.

Low Cut Selector Select if the low cut filter should affect the input: like a normal low cut filter placed before the TLA-100A, or just the detector so that low frequencies doesn't trigger the compressor.

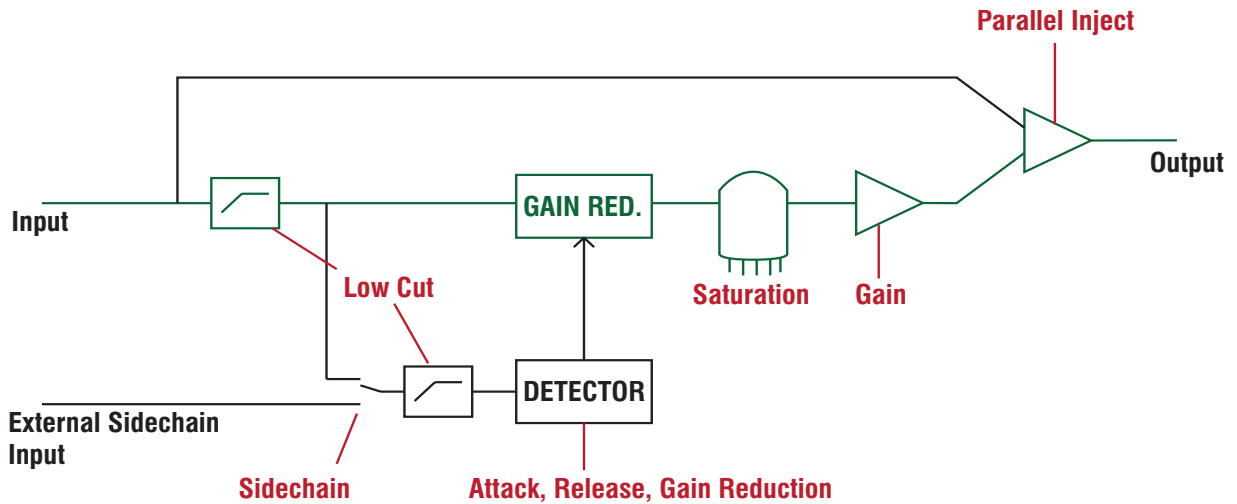
Saturation Sets the amount of headroom in the unit. If you need more distortion, turn up this knob. Distortion can sometimes be very useful together with slower attack times when you get loud transients. Use the **Saturation** knob to limit the transients.

Saturation LED Indicates distortion at the output. You will get the best result if the saturation is used sparingly on percussive sounds, like a drum bus. Let it take care of occasional peaks, and do not run it too hot.

Parallel Inject Adds the un-affected dry signal to the compressed (WET) signal. If you are using the compressor to shape the sound instead of compressing the volume (a rather normal use case these days), it is often useful to blend some of the original signal with the compressed track. An overcompressed drum track has few transients, which will make it sound dull and without energy, but if you blend some of the original transients into that track you'll get the energy back while maintaining the overcompressed sound.



1. Low cut (frequency)
2. Low cut selector
3. Saturation saturation LED
4. Parallel inject



Block Diagram

The drawing above illustrates the functionality of the plug-in. Green lines illustrate the main signal path and red labels the parameters.

The **Low Cut** parameters switch between either the detector low cut filter or the input filter.

Only one of the low cut filters is active at a time or both bypassed (by setting the **Low Cut** frequency parameter to OFF).

Mono and Stereo Operation

The Summit Audio TLA-100A plug-in can operate in both stereo and mono modes. In stereo mode the plug-in will work as the real unit in “stereo link” mode. This means that both channels will get the same amount of gain reduction (which will preserve your stereo image). How the different modes (stereo, mono and mono-to-stereo) are selected depends on your host software.

Credits

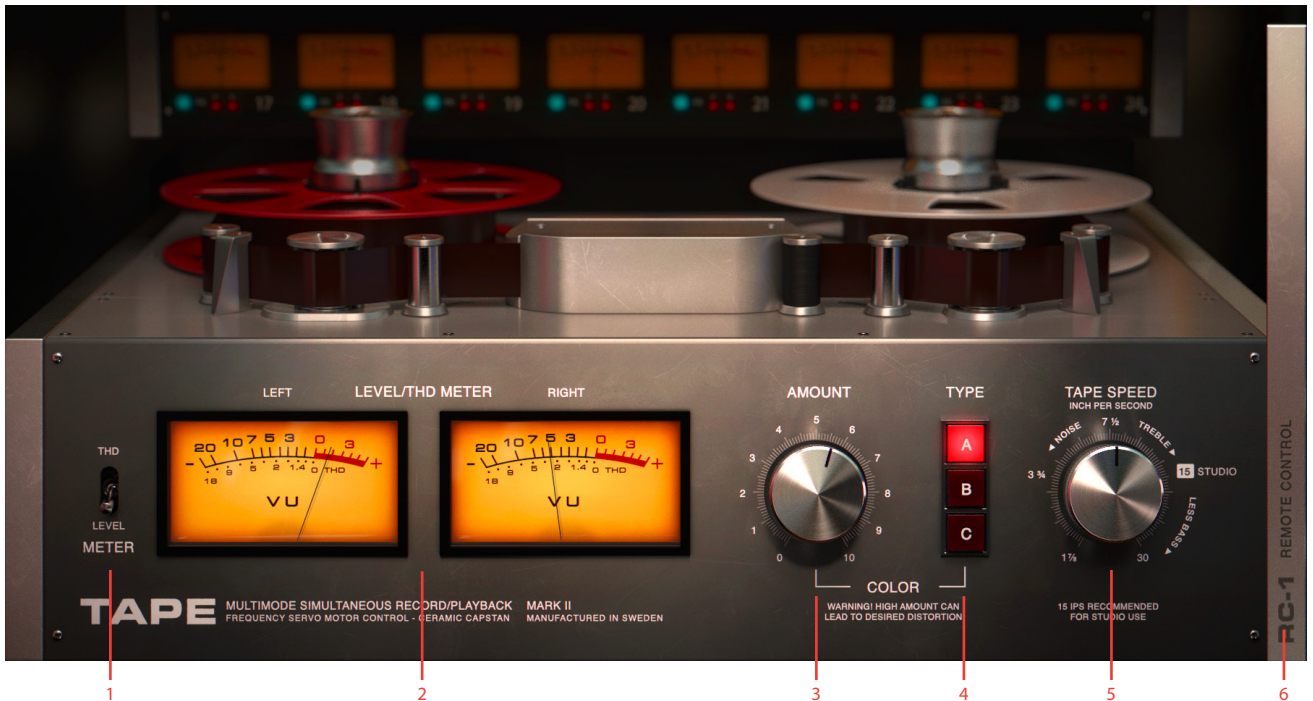
Oscar Öberg – modeling, **Niklas Odelholm** – extra features and testing, **Torsten Gatu** – framework programming, and **Arvid Rosén** – framework programming, **Ulf Ekelöf** – 3D rendering.



27 Tape

AFTER YEARS OF RESEARCH, WE decided to create our own tape emulation plug-in that could easily stand out among other tape emulations. We've modeled the characteristics of the tape machine circuits, tape head, the tape itself and took it even further in a versatile **Tape** plug-in, that can deliver a range of colors from subtle tape vibe to saturated warm tones. Geared with mastering grade **High-Frequency Trim** filter, Tape will shine on master bus and individual tracks as well.

Main View



1. THD/Level meter
2. VU meters
3. Amount
4. Type
5. Tape speed
6. Open side panel

User interface

Clicking the sidebar on the right side opens the panel with additional parameters.

Amount Controls the amount of processing.

Type Selects color types. Each color type is an essence of years of different circuits research combined to a set of iconic flavors delivering instant and rich tape vibe.

Side Panel with additional controls



1. Dry/Wet
2. Speed stability
3. High frequency trim
4. Crosstalk
5. Noise
6. Input/Output faders and meters
7. Run/Stop

TYPE A is an emulation of Swiss tape machine that gained its popularity in the 60s.

TYPE B tape sound was inspired by a machine with a transformer-based circuit.

TYPE C was inspired by British tape machine with a distinct vintage vibe.

Tape Speed Selects available tape speeds. 30 IPS (inches per second) and 15 IPS are the most neutral speed settings that will deliver mastering grade results and will work great on the master bus. **Speed** settings 7.5 ips and below are extremely useful for creating warm and fat tones.

THD/Level meter Switches the meters between input level meter and THD (Total Harmonic Distortion) meter.

VU Meters Show VU input level and THD amount. When in Level mode, the meters show the VU input level. Keep the input level below the red zone for subtle distortion amount. When in the red zone, the plug-in will add a larger amount of distortion and color.

When in THD mode, the meters show the distortion amount. It is handy switching to this mode to track even the most subtle distortion levels

Peak Meters Show input and output levels of the plug-in. They are useful to match the input and output gains for sound comparison.

Dry/Wet Controls the amount of dry and wet signal.

Speed Stability Controls the amount of tape wear and tear from stable to wobbly.

Using **Speed** stability together with **Dry/Wet** may lead to chorusing sound.

High-Frequency Trim A mastering grade filter that regulates the amount of high-frequency compensation. For the most linear frequency response set the parameter to **FLAT**. Increasing the amount towards **BOOST** will add beautiful sheen and clarity to the audio source. For vintage warm tones, move **High Frequency Trim** knob towards **CUT**.

Crosstalk Is a model of a magnetic tape crosstalk. The parameter controls the amount of crosstalk between channels of a stereo source. When used in Studio One MIX FX mode, it controls the amount of crosstalk between the channels in the session.

See more information about Studio One Mix FX mode at <http://www.presonus.com/products/Studio-One>

Noise Turns on and off tape noise.

Noise levels differ for each **Tape** speed.

Input Input fader controls the input gain. Use it in conjunction with Amount knob to achieve the desired amount of color.

Output Output fader controls the output level of the plug-in.

Run/Stop Run and Stop buttons emulate tape speed up and speed down behavior correspondingly.

In Use

Here are a couple of tips to get started.

Getting Started

tape **Types** are input dependent, so make sure to have correct gain staging, you can check the **Input** or **THD** meters for optimum levels. Insert **Tape** on every track or group of tracks and set the **Amount** until THD meter reads 1 - 1.4. If necessary, increase **Input** gain for more coloration.

Master Bus

To get started, choose 30 or 15 ips and adjust **Amount** to tame excessive transients first, then compensate high end with **HF Trim**. If the mix lacks punch and definition use crosstalk, it will bleed left and right channels in a way magnetic tape does. Use it to reinforce instruments in the mid (vocals, bass and kick). For most mastering applications you'll need up to 40-50% of the crosstalk. If the vocals get too prominent, use **HF Trim** to tame some of the top end.

Use **TYPE B** if subtle coloration is needed on master bus hence **TYPE B** has more tolerance than **TYPE A** and **C**. You may want to use **TYPE C** on material lacking punch in the low end.

Electric Guitar Overload

Speed at 3 3/4 and 1 7/8 can deliver a rich harmonic tone to electric guitars. To drive electric guitar, you may need to increase the **Input** gain and **Amount** to the maximum. For warmer guitar tone, use **HF Trim** to tame some of the top end.

Credits

Kim Larsson and **Björn Rödseth** - modeling, **Niklas Odelholm** - graphic design, **Henrik Andersson Vogel** - marketing, **Ulf Ekelöf** - 3D rendering, **Paul Shyrinskykh** - product management and sound design.

28 Tonelux Tilt

Foreword by Paul Wolff

After the success of the TILT control on the MP1 and MP1a Mic Preamps, I felt that a rack mount unit would be a nice addition to the Tonelux product line. The decision was made to make it an 8 channel unit, with polarity, in/out and access with D-subs for ease of use.

After using the TILT rack unit on a few sessions, I was shocked at how useful it was, and how smooth and sweet it made tracks. I tried it on everything from Bass to Overheads, Guitars, Vocals etc. It did so much with only one knob that the only logical move was to make a plug in. While designing the features, Softube and Tonelux thought that having a few extra options would really fill out the module.

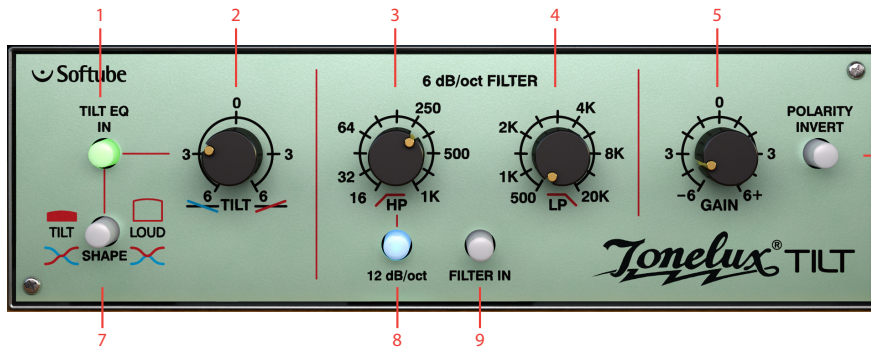
We decided to model the transformer in the Tonelux products, and added a very smooth high and low pass filter, not to fix problems, but to have the ability to use a filter that sounded like it wasn't there. On top of that, we added a loudness feature to the TILT knob, allowing the engineer to boost both low and high at the same time, much like a loudness control used in post production, to simulate near and far positioning. So far, the few friends that we have sampled it to have loved it, with comments like "it's just smooth" or "beautiful on over heads, just enough sheen".

After showing the TILT rack unit at a trade show,



it was suggested that we consider a live version, which might include some different features, one of which was a way of limiting the boost, but retaining the TILT effect, to prevent feedback in wedges and side fills, so Softube came up with a unique "BOOST CEILING" control, where you can set the amount of maximum boost above normal, to prevent feedback, but when using in-ear monitoring you can still have the original effect. It works really well to compensate ear fatigue without altering a complex EQ or certain individual channel EQs. It never really gets "louder", it just gets "clearer".

*Paul Wolff
Designer and Founder of Tonelux*



1. Tilt EQ_{in}
2. Tilt
3. High pass
4. Low pass
5. Output gain
6. Phase invert
7. Shape Tilt/Loud select
8. 12 dB/oct select
9. Filter in

Introduction

THE TONELUX TILT PLUG-IN IS a joint development effort between Tonelux designer Paul Wolff and Softube, bringing the famous Tonelux sound to the digital world. The innovative Tilt design was first featured on the Tonelux MP1a discrete mic preamp module and has been a godsend for engineers and producers that need to take control of their sound in a fast and effective manner. Often, the Tilt knob is all you need to make a track sit better in a mix or to make a track “warmer” or “cooler” and it’s perfect in a live situation where you need to tweak the whole mix to adjust for ear fatigue.

The TILT plug-in features not only the original Tilt design but also adds a couple of new features. A special “Live” version of the TILT plug-in is included which is optimized for the DIGIDESIGN VENUE system and contains features requested by live sound engineers. The TILT also includes modeling of the transformer found in the MP1a mic preamp, which adds a subtle distortion for bass frequencies that fattens up the low end of a track.

User Interface

The TILT is all about getting the sound you want fast and easy. The interface is made up of three sections which will give you flexible and powerful control of your sound.

First section: **Tilt Eq In, Shape and Tilt.**

Second section: **High Pass, Low Pass, 12 dB/oct and Filter In.**

Third section: **Gain and Polarity Invert.**

Want to do a fast mix? Try inserting a TILT on each track of your mix and you’ll find that often the **Tilt** knob is the only EQ you’ll need.

Tilt Section

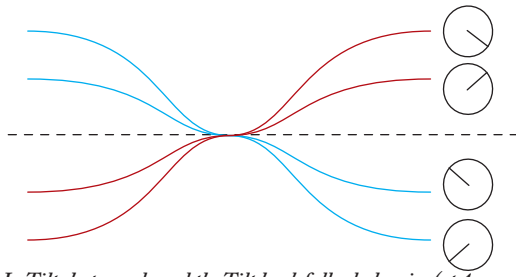
The Tilt section consists of **Tilt Eq In**, which engages the Tilt/Loud equalizer, the **Shape** switch, which lets you select between a Tilt style equalizer or a Loudness equalizer, and finally the **Tilt** knob, with which you adjust the amount of the equalization

When using the **Tilt** knob less is more! Remember that when for example cutting high frequencies, the perception can be that you're boosting low frequencies. Thus you will have double effect when turning the **Tilt** knob.

Tilt Equalization

The **Tilt** control rebalances the whole sound of a track with just one twist of a knob, going from a bright and shimmering top to a warm and wide low-end.

The filter of the **Tilt** is shaped so when one end of the frequency spectrum goes down, the other end goes up, thus cutting frequencies in one end while boosting in the other. The center frequency of this equalizer has been carefully selected to work on a wide range of material and provides a safe way of altering your sound without



In Tilt shape mode and the Tilt knob fully clock-wise (at 4 o'clock), you will get a high frequency boost and a low frequency cut.

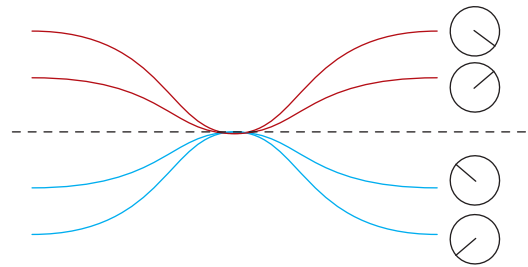
it getting too harsh or too boomy. The gain of the filter goes from 0 to 6 dB.

Loudness Equalization

By pressing the **Shape** button, making it unlit, the function of the **Tilt** knob is changed from TILT to LOUD.

The **LOUD** setting changes the shape of the filter to that of a loudness control, meaning that when you turn the **Tilt** knob clockwise you will get more bass and more treble and turning it counterclockwise will give you less of both.

Turning the **Loud** control from min to max can give the illusion of a sound going from far away to up close. This is a handy



In Loud shape mode and the Tilt knob fully clock-wise (at 4 o'clock), you will boost both high and low frequencies.

effect for post production.

The **EQ In** button will engage or disengage the TILT/LOUD filter making A/B comparisons easy.

Try using the **Loud** filter with different settings on different sounds that needs to be separated. For example, imagine a guitar track and drum track getting in the way of each other. Boost the **Loud** filter on the drums and cut it on the guitar will place the two in the mix with less clashing frequencies.

Filter Section

The filter section with its two filters will further shape the sound or help you get rid of problematic frequencies in a track. The **Low Pass** filter cut-off frequency goes from 16 Hz to 1 kHz and will attenuate the signal 6dB per octave. When the **12 dB/Oct** switch is engaged the signal will be attenuated 12 dB per octave.

The **High Pass** filter cut-off frequency goes from 20 kHz to 500 Hz and is set to attenuate the signal 6 dB per octave.

By engaging/disengaging the **Filter In** button the **HP** and **LP** filters will be enabled or in bypass respectively.

Output Section

Depending on what version of the TILT you are running, the TILT or TILT LIVE, the knob in the output section will be either a **Gain** control (TILT) or a **Boost Ceiling** control (TILT LIVE). The **Boost Ceiling** is explained below in the TILT LIVE section.

The **Gain** control is a standard make up gain going from -6 dB to +6 dB.

The **Polarity Invert** button inverts the phase when pressed (lit up).

Tilt Live

The TILT LIVE plug-in is optimized for the Avid VENUE system and contains features requested by live sound engineers. The TILT LIVE version does not include the transformer modeling to preserve DSP power and features the **Boost Ceiling** control.

The TILT LIVE is perfect for in-ear systems when you want to tweak the mix to adjust for ear fatigue, without changing the actual volume of the mix.



In normal operating mode (**Boost Ceiling** on full), the filters in the Tilt/Loud circuit will attenuate some frequencies while boosting others, just like the normal TILT plug-in. For live performance purposes this can be a drawback, since boosting frequencies can sometimes cause feedback in a live monitoring system.

By setting the **Boost Ceiling** on min, you have limited the amount of boost to 0 dB, i.e., no boost at all. Whatever setting you put on the TILT LIVE plug-in, it will never boost any frequencies.

For monitor systems on stage the TILT LIVE can help you changing the balance of a mix without having to risk getting feedback from increasing the volume.

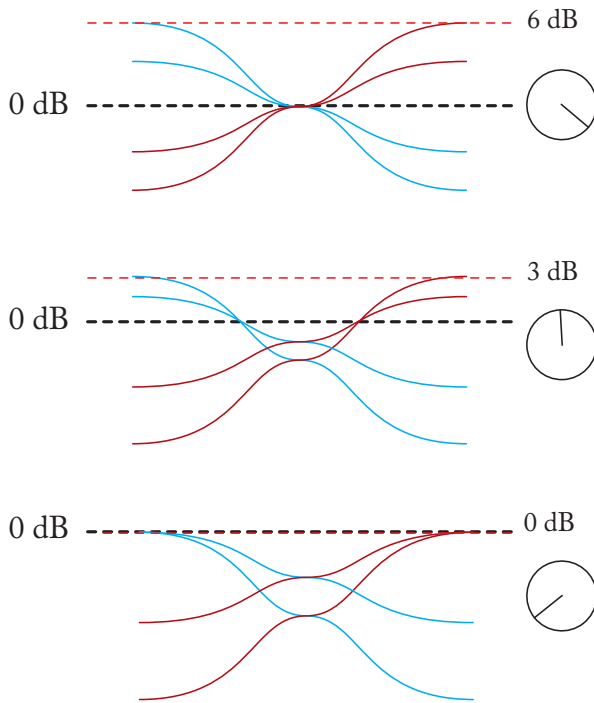
Boost Ceiling

When using the TILT LIVE the **Gain** control will be replaced with a **Boost Ceiling** control.

The **Boost Ceiling** control makes the TILT filter behave differently depending on the **Boost Ceiling** setting.

When the **Boost Ceiling** is at its minimum the audio level will never be amplified. The filters will have the same shape but will always be below 0 dB.

When the **Boost Ceiling** is set to its maximum it will work as the ordinary TILT. When set between, there will be some boosting of frequencies, but never more than you dialed in



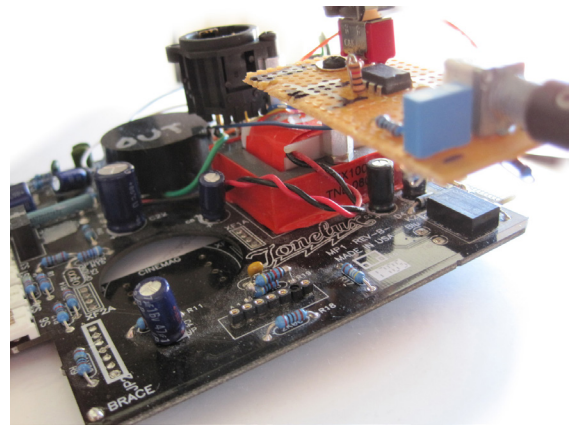
Tilt filter curves for different settings of the Boost Ceiling knob. Top: Boost Ceiling at “In Ear” setting (6 dB), middle: 3 dB, and bottom 0 dB (“Wedge” setting)

Buying Recommendations

The TILT and TILT LIVE plug-ins aren’t based on a single piece of hardware. Designer Paul Wolff wanted to create some extra functionality for the plug-in version of his Tilt module, and worked night and day to design a hardware prototype that is the basis of these two plug-ins. However, if you are desperate to find the Tilt control in a hardware package, we can strongly recommend the fantastic sounding Tonelux MP1a mic preamp module



that features the Tilt filters, or the eight channel Tonelux Tilt rack unit.



Here you can see the eight channel Tilt unit by Tonelux, and below is a photo of the prototype Paul Wolff sent us. It looks like, well, a prototype, but sounds fantastic!

Credits

Niklas Odelholm – modeling, **Oscar Öberg** – DSP programming. **Torsten Gatu** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **Paul Wolff** at Tonelux.



29 Transient Shaper

ONE OF THE MOST POPULAR FEATURES of the Console 1 is the addition of the Transient Shaper in the Shape section. The ability to quickly get expanding or compressing effects by a quick tweak of a single knob is extremely useful and in many cases completely eliminates the need for additional dynamics processing. While developing the original Transient Shaper for Console 1 we discovered that the underlying algorithm was a very powerful dynamics tool, and that with a couple of small additions we could unleash the true power of it. And in true Softube spirit we wanted to make a product with few simple controls that become a powerful combination that allows you to completely transform your audio track.

The secret ingredient? A cross-over section for the punch or sustain to affect the high or low frequencies. Try it out yourself, and you'll notice how powerful it is. Add a bit of high frequency sustain to some strings or voice. Remove some low frequency sustain on overheads, or combine the two by increasing the high frequency punch and removing some low sustain on a kick.

Shaping vs. Compressing

So how is the Transient Shaper different from your standard dynamics tools, like a compressor? First, it is level independent. This means that it doesn't *need* a threshold control and will react the same even if you change the gain of your track. Furthermore, it doesn't care about the overall level of a track, it only cares about the transients (**Punch**) or lack of transients (**Sustain**).

While it might be easy to think of the **Sustain** and **Punch** as "threshold" controls on a auto make-up compressor, it is simply wrong to do so. For example, going from the minimum setting on the Sustain up to the maximum, you will start with an effect that is more related to expanding than compressing, and end up with something that sounds similar to a hard "all buttons in" mode compression.

But while it might sound similar to compression and expanding, that's not really what it is, but we won't delve further into those details today.

Knobs and Switches

Here is a brief overview of the knobs and switches of the Transient Shaper.

Sustain Increase or decrease the amount of sustain in the audio. A setting below 0 dB will effectively reduce sustain, for example unwanted ringing in toms or a kick drum. A setting above 0 dB adds more sustain.

Sustain Band Select if the **Sustain** should affect high frequencies, low frequencies or the whole sound (full bandwidth). Increasing high frequency sustain will make a audio source sound more vibrant without cluttering up your mix.

Punch Increase or decrease the transients. Add **Punch** to get more pronounced transients, for example to make it sound as if the drummer is hitting the drums harder. Lower the punch to make the drum hits softer. Punch is only effective on sounds with fast transients, like a drum kit or plucked bass. For “slower” instruments, like voice or piano, it is more effective to work with the Sustain control instead of the punch, although it is possible to soften hard sounds in a vocal track by reducing punch.



1. Sustain
2. Sustain band
3. Gain change
4. Clip LED
5. Output level
6. Punch
7. Punch band
8. Punch type
9. Crossover
10. Output level
11. Clip on output

Punch Band Select if the Punch should affect high frequencies, low frequencies, or the whole sound (full bandwidth). Increasing high frequency punch is an effective way of adding more “click” to an audio source, such as a kick drum, while decreasing high frequency punch is very useful to tame sharp transients without taking away too much energy from them.

Punch Type Adjust the timing of the punch. A **SLOW** punch type is smoother while the **FAST** punch is more aggressive.

Crossover The center frequency of the crossover filter.

Level Output volume

Clip Turn on or off output distortion. When on, the output soft clips at 0 dB. Clipping in conjunction with increased transients is a very nice way of adding power to each transient without getting too loud output levels. Clipping is indicated by the **Clip LED** in the meter window. It is also a good idea to set **Clip** to **CLIP ON** when you’re dealing with really loud transients.

Meters

Gain Change These two meters indicate the gain change in the High and Low frequency bands.

Clip LED Indicates if the output is clipping

Output Volume Output peak metering

Slow vs. Fast Punch

There are two main differences between **SLOW** and **FAST Punch Type**,

1. *Transient detection:* the **FAST** punch has faster detection, and is better at detecting sharp transients. If the source transient has a slow build-up, it is better to use the **SLOW** mode, since **FAST** mode will miss a slow transient.
2. *Transient shaping:* the **FAST** punch will shape a shorter part of the transient than the **SLOW** punch, and will therefore sound snappier, as compared to the more fat punch from the **SLOW** mode.

Typical Use Cases

Here are a couple of typical use cases for the Transient Shaper.

Kick Drum

To get more click in the kick, add some punch in the high band:

Punch: +
Punch Band: HIGH
Punch Type: FAST
Crossover: 700 Hz
Clip: ON

Snare Drum

Decrease low frequency rumble and ringing by decreasing sustain for the low band:

Sustain: -
Sustain Band: LOW

A snare often needs a slower punch to increase the “fatness” of the attack. Let the output clip if the level gets too high, distortion on a snare drum works very good when you want to beef up a drum track:

Punch: +
Punch Band: WIDE
Punch Type: SLOW
Clip: ON

Overheads

Add room and shimmer. Avoid amplifying the low frequency rumble by only increasing the sustain for the high band:

Sustain: +
Sustain Band: HIGH
Crossover: 1-2 KHZ

Drum Kit Enhancement

If you want to shape the entire drum kit, you'll need to try both SLOW and FAST **Punch Type** to see which one is best suited for your recording. Just add a tiny bit of punch for the high frequencies:

Punch: 1-4 DB
Punch Type: SLOW OR FAST
Punch Band: HIGH
Crossover: 2-4 KHZ

Vocals

Add shimmer and breath:

Sustain: +
Sustain Band: HIGH
Crossover: 1-2 KHZ

Tame plives by decreasing the punch for the entire signal. The punch detector will only react on the worst offenders, leaving most of the vocal track intact:

Punch: -
Punch Band: WIDE
Punch Type: SLOW

Bass Guitar

Add body by increasing the low frequency sustain:

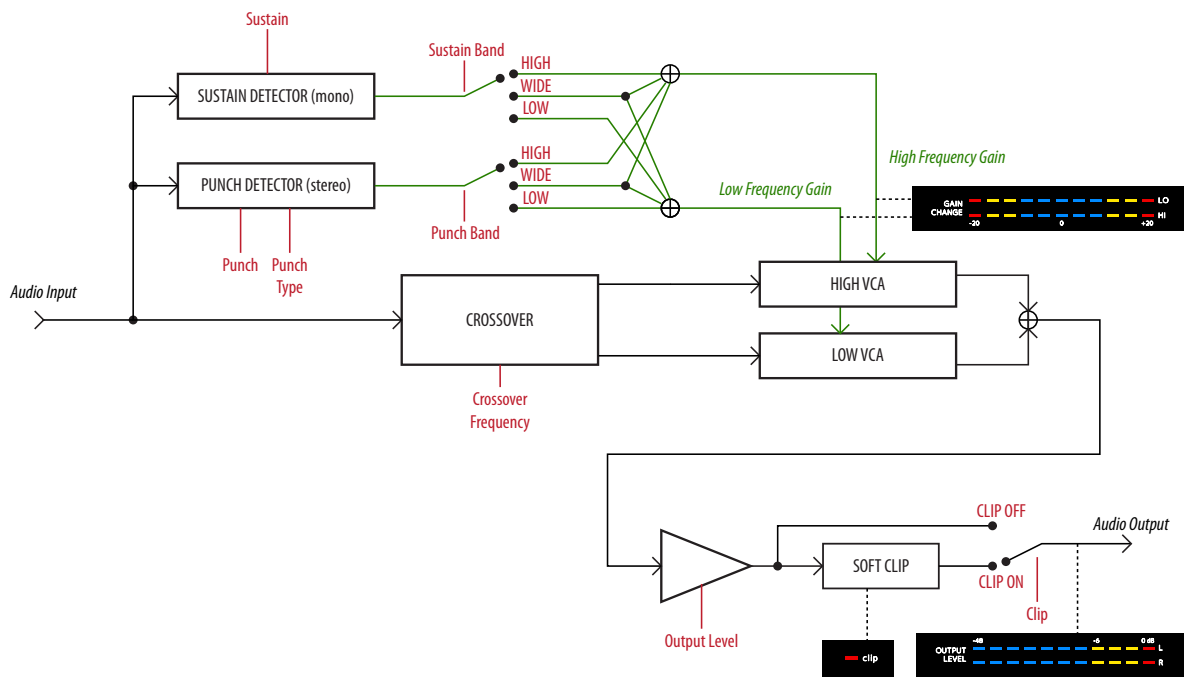
Sustain: +
Sustain Band: LOW
Crossover: 100-200 HZ

Stereo/Mono Operation

The Transient Shaper operates in both stereo and mono. The sustain detector operates on the combined LEFT + RIGHT signal, while the punch detector operates in true stereo.

Block Diagram

The block diagram is a simplification of how it really works, but a useful help to figure out which knob to adjust.



Credits

Niklas Odelholm – concept, modeling, sound design. **Paul Shyrinskykh** – quality assurance. **Patrik Holmström** – framework programming. **Henrik Andersson Vogel** – presets and marketing. **Arvid Rosén** – framework programming. **Oscar Öberg** – framework programming. **Ulf Ekelöf** – graphics rendering. **Torsten Gatu** – framework programming. **Mattias Danielsson** – technical support.

30 Trident A-Range

Foreword by Malcolm Toft

I have evaluated the Softube Trident A-Range equaliser plug-in at my studio with some very experienced recording engineers. After exhaustive listening tests with many different instruments and vocals, I am pleased to say that we all agree this is an incredibly accurate software recreation of my original A-Range design from 1971.

It exhibits all of the nuances of tone from the original, right down to the 'saturation' control which emulates the overdriven artifacts from the original when large amounts of equalisation are applied or the input is driven hard.

Well done to all the guys at Softube.

*Professor Malcolm Toft
Original designer of the Trident A-Range console*

*Torquay, Devon, England
December 17,
2009*

Background

THE TRIDENT™ A-RANGE® IS A LEGENDARY piece of equipment. As only thirteen A-Range consoles were ever made, it remains a holy grail for sound engineers and producers around the world that hope to create equal works of art to those that were originally recorded on the A-Range – David Bowie's Ziggy Stardust, Queen, Elton John, the list goes on...

The A-Range was originally designed in the early 70's for Trident Studios in London by Malcolm Toft and Barry Porter. Toft was employed as a sound engineer at Trident Studios and when the studio needed a 24 track recording console, he started Trident Audio Developments to design and build the Trident A-Range. Other products followed and found their way to studios around the world. Toft remains active to this day, developing products under the Trident name.

The A-Range plug-in from Softube is the first and only native plug-in that has been fully endorsed by Toft and Trident Audio.



About the Trident A-Range

The plug-in version of the A-Range is modeled after a channel on one of the original 13 Trident A-Range consoles. This particular channel strip is channel number 15 from Sweet Silence Studio 'B' in Denmark and was borrowed from Danish producer Flemming Rasmussen. The A-Range console has been in Rasmussen's care for a long time and used by Flemming Rasmussen when recording albums such as Metallica's *Ride the Lightning*, *Master of Puppets* and *...And Justice for All*.

The A-Range channel features four bands of equalization and high and low pass filters, the sound of the filters are often referred to as "colorful" and "musical". When running a hot enough signal through the original unit and boosting the bands it is possible to get it to distort, the sound is a hairy and effective saturation that is just right in some situation and just wrong in others! Even though the unit was never designed to distort, it has definitely been used this way by renowned producers and we thought it was only appropriate to stay true to the original unit and add the saturation to the plug-in as an option!

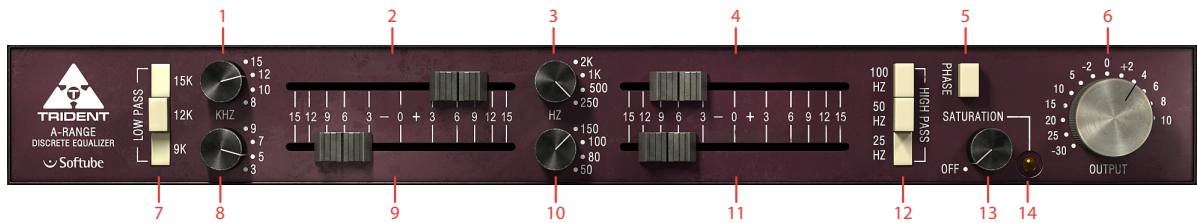
User Interface

The layout of the A-Range equalizer is simple and easy. You have four different bands of equalization, with a frequency selector knob and a gain fader each. **Low** and **High Pass** filters can be activated by pressing the off-white buttons, and a handy **Phase Invert** switch is found on the right side of the panel. The **Saturation** knob engages the console saturation, and is best used as a subtle effect. If you don't wish to get saturation, just put it in the OFF position and all saturation circuits will be de-activated.

The A-Range console was actually designed to have very little distortion, but it

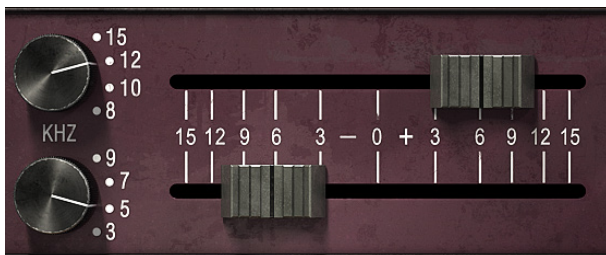
was nevertheless possible to crank up the input volume and equalization bands to make it distort, an effect that has been of much use in lots of recordings. The actual amount of saturation depended heavily on the audio material and amplification before the console, so we added the **Saturation** knob to make things easier for you. If all bands are set to zero gain, **Saturation** on its default position (12 o'clock) and **Output Volume** on 0 dB, the A-Range plug-in will have unity gain.





- | | |
|------------------|--------------------|
| 1. High freq. | 8. High mid freq. |
| 2. High gain | 9. High mid gain |
| 3. Low mid freq. | 10. Low freq. |
| 4. Low mid gain | 11. Low gain |
| 5. Phase invert | 12. High pass |
| 6. Output volume | 13. Saturation |
| 7. Low pass | 14. Saturation LED |

The four equalizer bands have two controls each – a “frequency selector” knob and a gain fader. The fader will boost the signal up to 15 dB by dragging the fader to the right and attenuate the signal by 15 dB by dragging it to the left.



High Shelving Band The frequency options for the **High Shelving Band** are 15 kHz, 12 kHz, 10 kHz and 8 kHz.

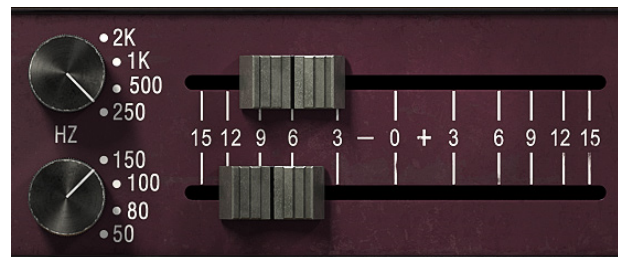
When boosted, the High Shelving Band gives a beautiful high-end shimmer to just about any material. Don't be afraid to try it on a master bus, as the result can be fantastic!

High Mid Band Bell type with the following center frequencies: 9 kHz, 7 kHz, 5 kHz and 3 kHz.

The bell filters interact with each other in a rather funny way. For example, even if the gain of the Low Mid Band is set to zero, changing the frequency of that band will affect the frequency response of the High Mid Band filter significantly. The results are quite unpredictable and this is part of the reason for the unusual sound of the A-Range!

Low Mid Band Bell type with the following center frequencies: 2 kHz, 1 kHz, 500 Hz and 250 Hz.

Low Shelving Band The frequency options for the **Low Shelving Band** are 150 Hz, 100 Hz, 80 Hz and 50 Hz.



Low Pass and High

Pass Filters The **Low Pass** and **High Pass** filters are controlled by three buttons respectively, which sets the cut-off frequency for the filters – 15 KHz, 12 kHz and 9 kHz for the Low Pass and 100 Hz, 50 Hz and 25 Hz for the High Pass. The cut-off characteristic is unusual and not exactly coherent but the slope is about 12 dB per octave.

The three settings work independently and can be used in any combination which means that the more buttons you engage, the more attenuation you get. For example, pressing all three Low Pass buttons will actually result in a filter with an even lower cut-off frequency than 9 kHz.

Phase Invert The **Phase** button inverts the phase when pressed.

Saturation The **Saturation** or distortion of the A-Range is placed after the equalizer bands and is depending on the gain settings of the bands. A louder or “hotter” signal will make the unit distort more and a weaker signal will make it distort less.

To be able to handle signals of different volumes and to make up for equalizer band settings we added a **Saturation** knob that basically is a volume compensated input gain for the saturation circuits. The LED next to the saturation knob indicates when the signal is being distorted. If you want to be sure not to add any saturation to the sound, click the OFF label and the saturation circuit will be bypassed.

You will get the best result if the saturation is used sparingly on percussive sounds, like a drum bus. Let it take care of occasional peaks, and do not run it too hot.

Output Volume The Output knob controls the output volume from the plug-in. The range of the output volume is -30 dB to 10 dB.



VENUE Stereo Operation

VENUE stereo is not supported due to DSP limit constraints. You will need to use the MULTI-MONO mode instead of STEREO mode. Normal STEREO mode is however available in AAX.

Buying Recommendations

We always make buying recommendations in our manuals for a hardware equivalent of the plug-in. In this case, it's really simple and really hard – If you can find an A-Range channel and can afford it, get it!

Credits

Niklas Odelholm – modeling, **Oscar Öberg** – DSP programming and modeling. **Torsten Gatu** – framework programming. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **Malcolm Toft** and **Barry Porter** at Trident Audio Developments. The original unit is owned by **Flemming Rasmussen** at Sweet Silence Studios.

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This is a photo of channel 15 and 18 from one of the original Trident A consoles. These two are the ones we measured for the A-Range plug-in. A few cosmetic changes have been made to these units while at the Sweet Silence Studios: the original flip switches on the Mute button has been changed to big red switches, the gray fader handles have been changed to bright yellow, and three frequency selector knobs have also been replaced. But it has the original face plate color, a strange but beautiful aubergine-style purple!

31 TSAR-1 Reverb and TSAR-1R Reverb

**MODULAR
READY**

FOR A LONG TIME WE HAD A DREAM about creating the perfect reverb. The most obvious requirement was of course that it should sound extremely good—like a natural room, and better. The reverb tail should be classy and natural, but yet lush and vibrant. It should range all the way from sophisticated halls to a more modulated, sparse and dreamy reverberation. A reverb that surrounds the listener just as a real room does.

While experimenting with different novel reverb designs we realized that the biggest challenge would be to create a product that our users actually could learn how to use. And that led us to the second requirement: it need to be easy to use! We didn't want a reverb with fifty parameters, so complicated that no-one ever dares to change anything, and by that forces the user to use presets.

So once the algorithm was finished we started the work of getting it down to a few, but meaningful, parameters. But don't be fooled by its simplicity, it's one of the most powerful reverbs ever released.

After the original reverb, the TSAR-1, was finished, we

decided to go even further and make a natural sounding reverb with just three parameters—and that became the TSAR-1R. It has the same astounding sonic qualities as the TSAR-1 (it's exactly the same algorithm), but focuses solely on natural spaces. And with only three parameters: Time, Predelay and Color, getting a good reverb has never been easier.



Introduction

TSAR-1 STANDS FOR True Stereo Algorithmic Reverb Model 1, and that's just what it is. But TSAR is easier to remember than, well, what ever it stands for...

True Stereo

Both the left and right channels contribute uniquely to the left and right outputs, so the right channel doesn't only affect the right channel but also the left side.

Unlike many other reverbs, which might have a two different engines for the left and right sides, or just one engine with different outputs for the left and right sides, a true stereo reverb needs four unique engines to process the audio. This is true stereo, and this is what the TSAR-1 engine does.

Algorithmic Reverb

Even if there are tons of convolution reverbs out there, and anyone with a simple piece of software can create their own set of impulses, there are nothing that compares to a well designed algorithmic reverb.

An impulse response is indeed a very good finger print of a room's acoustics, and does a good job in mimicking a sampled space if you have a completely dry recording. You'll get all the echoes, tails and coloration of that room. An algorithmic reverb is on the other hand designed to add space or ambience to a recording without coloring the dry signal and without interfering with the already recorded ambience. The ability to tailor the reverb to co-exist with the recorded ambience is one of the strengths of an algorithmic reverb, and one very strong reason to why you will always need well designed algorithmic reverbs.

Anatomy of a Reverb

EVERY REVERB DESIGNER has her own way of implementing the user adjustable parameters, but many parameters are the same in most reverb designs and are today just as common in reverbs as bass and treble are in a guitar amp.

Predelay

The Predelay sets the time between the dry signal and the reverb tail. It is often used to achieve the impression of a large room, by making the reverb tail arrive later. A 34 ms predelay corresponds to the time it takes for the sound to travel 10 m. In modern music production, predelay is often used to achieve rhythmic effects, often set at longer times. An 8th notes slap back each in 120 bpm corresponds to a 250 ms predelay.

Reverb Time/Decay Time

This is the time it takes for the volume of the reverb tail to drop 60 dB. A real world analogy for the reverb time could be how much the walls of a room reflect the sound. A big room with much acoustic treatment has shorter reverb time than a stone-wall church of the same size.

Density/Size/Diffusion/Shape

These parameters are used to adjust the sound of the reverb tail. A room with a large size often have longer time between reflections than a small room, and the sound of the reverb becomes less dense. Low density reverbs are very handy if you need a reverb with a long tail that doesn't take up too much energy in a mix. Diffusion alters the build-up of the reverberation, and can have a dramatic effect on percussive sounds.

Early Reflections

The early reflections are the very first echoes that reach the listener and help the listener to decide the size of the room.

User Interface at a Glance

Early Reflections Type The early reflections give the user a sense of the room dimensions.

Early Reflections Mix The mix between the early reflections and reverb tail.

Diffusion High diffusion gives a smoother sound, but also a sound that takes more space in the mix.

Modulation Normal mode is **RANDOM**, but you can set it to **SLOW** or **FAST** for a more chorusy effect.

Reverb Mix The mix between the direct signal and the reverb signal (including early reflections).

Output Volume Sets the output volume of everything (including dry signal).

Predelay The amount of delay between the early reflections and reverb tail.

Time The time it takes for the reverb tail to decay.

Density Higher density gives a thicker, smoother reverb with more reflections, but also the impression of a smaller room.

Reverb Tone Adjusts the tonality/color of the reverb tail.

High Cut Limits the high frequencies for both the tail and early reflections.



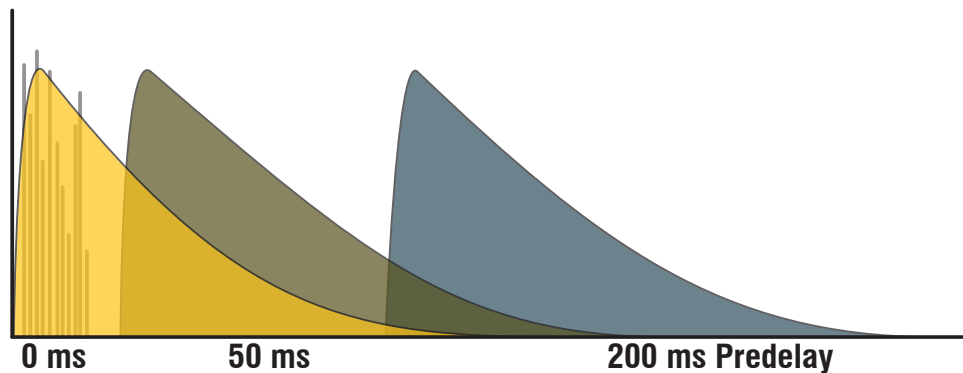
Reverb Parameters

Predelay

Sets the time between the *dry signal* and the reverb *tail*.

The early reflections (gray in the illustration) are **not** affected by the **Predelay** setting.

For a natural sound, there should not be a distinct gap between the initial echos (either from early reflections or from the dry signal) and the **Predelay**. Depending on the the character of the dry signal and the **Early Reflections Type**, the **Predelay** should usually be set below 50 ms.



Predelay and Early Reflections

Since the early reflections are not affected by the **Predelay** and by themselves have an inherent and independent delay, you need to tune the **Predelay** so that it match the length of the early reflections to get a natural sound. For the largest early reflections type, a time of 40–80 ms can be useful. See the **Early Reflections** section for more info.

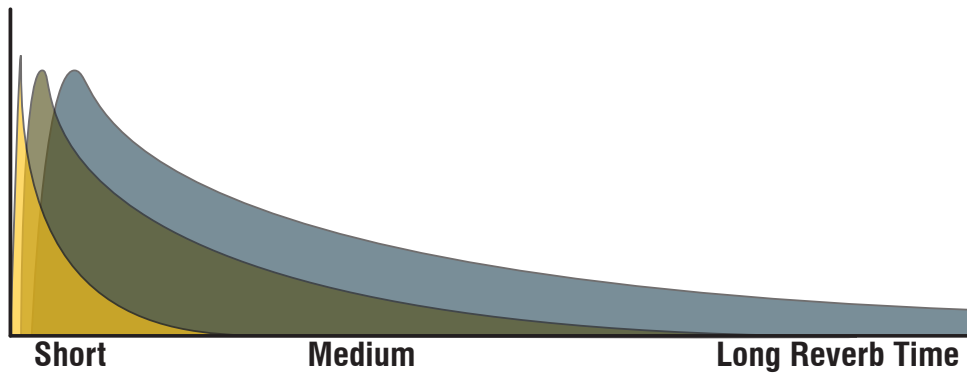
Predelay as an Effect

The **Predelay** is often used to achieve a rhythmic effect, such as a slap back echo on a snare drum, and then with much longer times (about 80–250 ms). A shorter **Reverb Time**, with **Diffusion** and **Density** set to **LOW** makes this echo more distinct.

Reverb Time

Sets the decay time of the reverb tail.

A longer reverb time gives the impression of a room with more reflective surfaces. If a long **Reverb Time** is used together with a low **Density**, you will get the sound of a large hall. If you on the other hand have a short **Reverb Time** and high **Density**, you will get the sound of a small, tight, studio.



Large Hall Setting

A large hall has often both a long **Reverb Time** and a lower **Density**. Set **Reverb Time** to 1.8–5 s and **Density** to 25–40%. **Early Reflections** on *LARGE* and about 40 ms **Predelay** enhances this effect.

Room and Chamber Settings

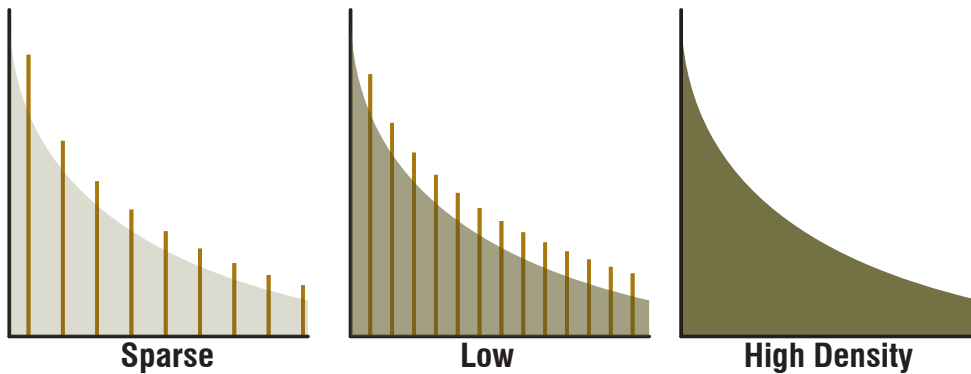
A shorter **Reverb Time** and higher **Densities**, you get the sound of a room or a chamber. A typical room has less than 1 s **Reverb Time** and a **Density** above 50%.

RT60

RT60 is a measurement of the time it takes for the reverb tail to decay 60 dB. The time specified in the display is an approximate RT60 measurement, since the actual RT60 time also depends on the **Tone**, **Density**, **High Cut** and **Diffusion** controls. As usual, use your ears instead of depending on numerical displays.

Density

Sets the density, “thickness”, or “smoothness” of the reverb tail. A low **Density** is similar to the sound of a large space, while a high **Density** corresponds to a small space.



Low Density Reverbs

To avoid coloration and muddy-ness in a mix, it is often good to use a less dense reverb. On a non-percussive or legato instrument, such as strings or voice, a sparse reverb will often sound clearer than a dense reverb. Less dense reverbs take up less energy in a mix and is often easier to use, even if they may sound a bit unnatural on its own.

Density vs. Size

Use a high density reverb to get the sound of a small space, and a low density reverb to get the sound of a large space.

Plate and Chamber Settings

To get a sound similar to that of a vintage plate or an echo chamber, you will often need higher density reverbs.

Gated Style Reverbs

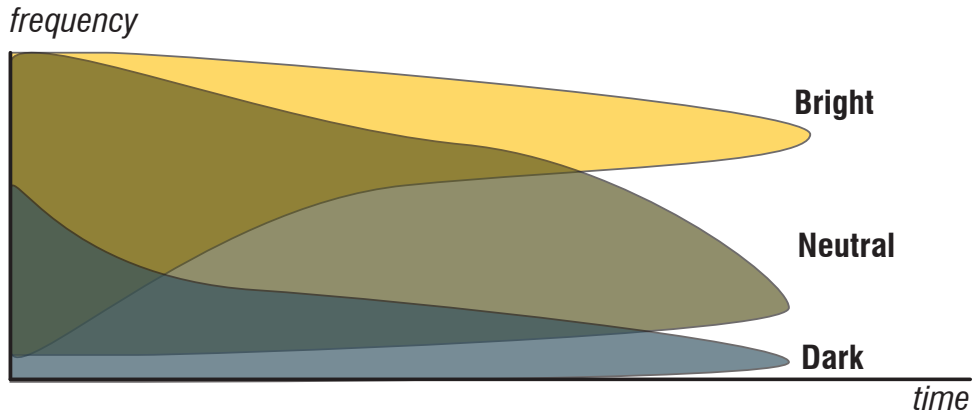
A “gated reverb” is an effect where you gate the reverb signal with the dry signal as side chain to achieve an abrupt cut-off of the reverb tail. You can mimic this sound by setting **Reverb Time** to its shortest value and **Density** below 20%.

Reverb Tone

Use the **Tone** control to adjust which frequencies that have the longest reverb times. With a **BRIGHT** setting, the high frequencies will decay more slowly, and vice versa with the **DARK** setting.

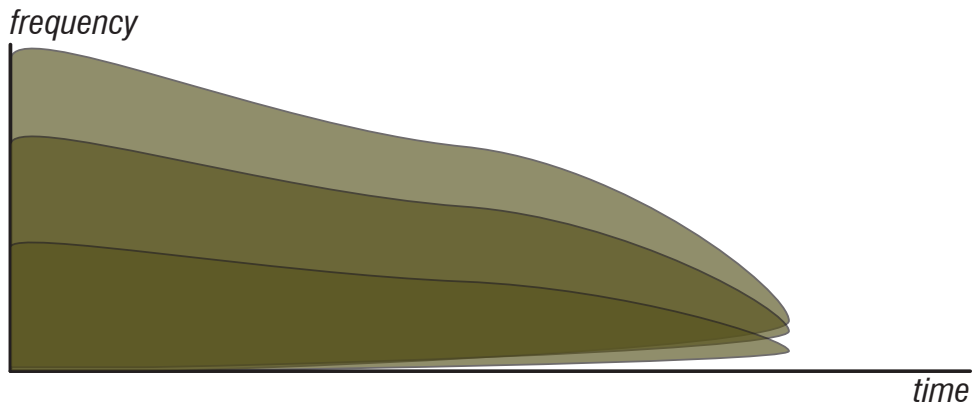
Settings between **NEUTRAL** and **DARK** will often give the most natural sounding reverb tails.

The **Reverb Tone** will only affect the reverb tail, not the early reflections.



High Cut

Sets high frequency attenuation for both the reverb tail and the early reflections. A cut in the high frequencies often results in a more natural sound.



Early Reflections

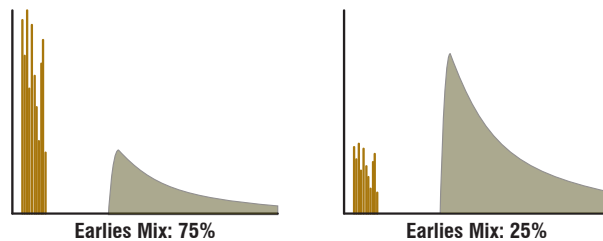
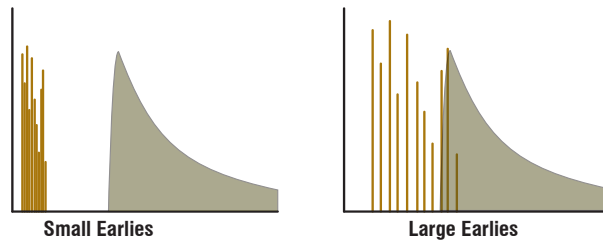
The early reflections give the listener a perception of the geometry and space. Set the type of geometry with the **ER Type** switch and the blend between late and early reflections with the **ER Mix**. If your material was recorded in a nice room with lots of early reflections, you might not want to impose a new set of early reflections on the recording, and you can thus lower the amount of early reflections, or completely turn them off.

For a close mike recording with very little ambience, early reflections can be very effective in placing the recorded material in a space.

ER Type vs. Predelay

Since the early reflections have built-in delays, it is often necessary to adjust the Predelay according to the **ER Type**. A large **ER Type** may need a longer **Predelay** than a small **ER Type**. The approximate delay times for the three different **ER Types** are listed below:

SMALL: 9–16 ms, MEDIUM: 30–70 ms, LARGE: 45–80 ms

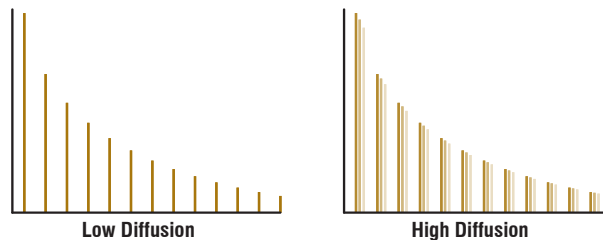


Diffusion

Sets the amount of “smear” in the reverb.

Low diffusion leads to a more immediate sound, sometimes with distinct reflections if the **Density** is low. Set **Diffusion** to HIGH to enhance the sound of percussive sounds.

Although high diffusion often leads to a more natural sounding reverb, it will also make the reverb take up a bigger place in the mix, which sometimes result in a muddier sound. If you, for your application, don't hear much difference between different settings of **Diffusion**, you might as well leave it at LOW.



Modulation

Select between three different types of modulation:

RANDOM: Designed to give a lush and uncolored reverb, without any “pitchy” or “chorus-y” effects. The default setting for most purposes.

SLOW: A more chorus-like modulation with a slow modulation rate. Suitable for extremely long reverb times and legato instruments.

FAST: Same type of modulation as **SLOW**, but with a faster modulation rate.

All three types of modulation are based on randomization, but **RANDOM** is just a bit more random than the other two...

Reverb Mix

Sets the mix between the direct and effect signal (reverb + early reflections). When using the TSAR-1 as a send effect, the **Reverb Mix** should always be set at 100% (wet). If you use it as an insert effect, a setting between 15–40% is a good starting point.

Output Volume

The **Output Volume** affects both the direct and effect signal. Usually you can just leave it at 0 dB, but if the output peak meter is going up in the reds it might be a good idea to lower the volume.

Input and Output Meters

The meters on both sides of the display shows the input (left hand side) and output (right hand side) peak levels for both the left and right channel. The meters have a red indicator at 0 dB.

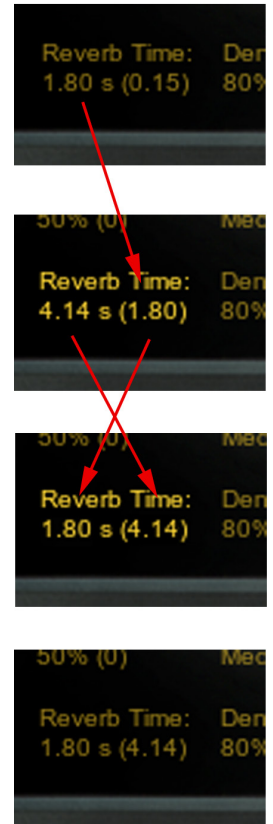
Parameter Display

The display shows the name of the parameter, the value of the parameter, as well as the previous value.

As soon as you start to change a parameter, the display will light up and the former setting will be displayed within the parentheses. While the display is lit up, the former setting will not be over-written.

By clicking on the parameter display, you revert the setting to the previous value.

After one second without changing parameters, the display will get dimmed again.



Presets

The presets are divided into two sections, the *modern* and the *vintage* section. The modern presets focus on application (drums, vocals, orchestral sounds, etc) and the vintage presets mimic sounds from vintage units.

Modern Presets

The modern presets have been designed with a -6 dB reference level, ie., with all faders set at 0 dB the send levels are set at -6 dB. This corresponds to a Mix level of 27% if the plug-in is used as an insert.

Ambience Dark

Opening up the low end is the most transparent way to add space to a source.

Drum Chamber

If you need a little more reverb on the snare or kit than what a room is giving you, but not a full blown plate, the chamber splits the difference and sits down nicely in the mix.

Drum Plate

Sometimes the snare or kit just needs a plate. Here it is.

Drum Room

Dryer recordings can still benefit from a little room sound on the snare or kit. Used in the right amount, it can make a good recording great, without adding any apparent reverb.

Guitar Hall

A small hall that is the perfect performance space for any acoustic string instrument. Works well with electrics too, when you want to hear the reverb.

Guitar Room

Widen up electric guitar tracks, or provide a little chorused ambience for acoustics.

Hall Dark

If your source has a lot of sharp transients that are jumping out of the track using the other halls, then this is the one to use. This one is very smooth.

Hall Large

This is the place for large orchestral works. The hall is large enough to really let the low end unroll but with a short enough reverb time to retain clarity. age units.

Hall Medium

Brighter and a little less reverb time than the larger hall, this one adds even more clarity to orchestral work. This is the hall to use for vocal ensembles, opera or spoken word.

Jazz Club

Perfect club size space for any performance. Just a hint of slap back to add a live feel.

Percussion Chamber

Highly diffused chamber designed for wood blocks, bells or hand slapped percussion sources.

Scoring Stage

The sound of an A-B mic'd scoring stage. Large and bright, this is the space to use for orchestral production of all sizes for film, or TV.

Studio A

A large studio tracking room big enough for the whole band. Well treated and very smooth, particularly in the high end.

Studio B

A smaller brighter tracking room ideal for the solo performer.

Synth Chapel

Long bright and clear with some sizzle. Huge size means huge bass as well in this one.

Synth Church

Similar to the chapel, but with a darker decay and more movement in the tail.

Synth Club

Short and full of sizzle. Use this when you want to highlight the source or excite the top end.

Vocal Chamber

If you need a little more reverb on the vocal than what a room is giving you, this chamber is a good alternative.

Vocal Church

A nice long bright church ideal for solo vocals or spoken word. The large predelay helps to retain clarity.

Vocal Plate

Bright and dense. This reverb is meant to be heard.

Vocal Whisper

When used in small amounts, this one brings the sizzle of the performance right up to your ear.

Vintage Presets

In this set of presets we have tried to mimic the sound from old vintage units. We chose to keep the brand names in the presets so that you see where we got the inspiration from.

224 Small Concert Hall

Spacious and expansive—low density initially then builds to a smooth reverb tail.

224 Large Concert Hall

A large empty concert hall with long rolling waves of reverberation.

224 Plate

Smooth and sweet, great on percussion.

224 Room

A medium room/chamber type ambience.

Ambience LRG

Ambience MED

Ambience SM

Carefully tuned and shaped, these add a subtle presence without sounding too wet.

AMS Ambience

Similar to the “Ambience” preset of the old AMS RMX16 reverb. Great on vocals.

Church

Dark Hall

Dark Stage

These are medium sized spaces with a warmer and more natural ambience than the vintage simulations. Great for spoken word.

Digital Snare

Rock Toms

Similar to a very rare and expensive SONY reverb from the early 1980s. Great for snare and toms.

EMT Hall

Similar to the EMT 250 digital reverb, great on anything.

EMT Plate

Similar to an early EMT mechanical plate reverb. Rich lows and sparkling highs.

Gate

Gate Kick

Gate Snare

Three gated reverb effects, one general purpose, one optimized for kick drum, and one for snare drum.

Plate Small

A tight plate type reverb, very dense and fairly bright.

TSAR-1R Reverb

THE TSAR-1R REVERB USES THE SAME algorithm as its big brother, but is controlled by just three reverb parameters.

Predelay

The **Predelay** has the same function as the TSAR-1 Reverb, but is limited to 200 ms maximum delay time (which is more than enough for natural sounding reverbs).

Time

Instead of just altering the Reverb Time parameter from TSAR-1, this Time parameter adjusts an internal predelay, the early reflections, diffusion, density and decay time. All these parameters have been meticulously fine-tuned to give as natural sounding result as possible for every setting.

Use the **Time** parameter to decide which type of space you want to set your instrument in. Don't be afraid of large halls. If you get too much reverb with a high **Time** setting, use the **Reverb Mix** or the send volume to decrease the amount of reverb.

Color

Set the overall tonal character of the reverb. **BRIGHT** is useful for strings and vocals, or when you want to give the perception of a highly reflective room, **NEUTRAL** for normal halls or studios, and **DARK** for smaller spaces with a lot of acoustic damping.



Automation

The TSAR-1 and TSAR-1R plug-ins fully support automation. But parts of the reverb becomes muted when parameters change in order to suppress noise artifacts (“zipper noise”). Do not use automation to modulate or gradually change parameters. A constantly moving parameter can lead to the reverb being muted.

Mono and Stereo Operation

The TSAR-1 Reverb plug-in is a true stereo reverb and for best performance you should always use it with a stereo output (even if you have a mono input). But it does work in both stereo and mono. How the different modes (stereo, mono and mono-to- stereo) are selected depends on your host software. In most hosts you can select these when you select the plug-in. In other cases, inserting one the plug-in on a mono track will usually make it use the MONO mode, while selecting it on a stereo track makes it use the STEREO mode.

Credits

Oscar Öberg, Niklas Odelholm, Torsten Gatu, and Arvid Rosén.

32 Tube Delay

Introduction

THE DELAY AUDIO EFFECT HAS A LONG history going back to the times when recording engineers would experiment with analog tape to get a decaying echo effect. Later on, standalone tape delay units were built and even later, delay created with digital technology became possible. Vacuum tubes, or valves, used in audio equipment are known for their unique sonic imprint on sound, something we perceive as “warmth” when the signal passing through it is overdriven.

Tube Delay is a marriage between analog and digital, combining the warmth and natural compression from tubes with the features and versatility of a digital delay. There are three tube sections in Tube Delay, one in the direct signal section, one in the delay feedback loop and one after the feedback loop. The tube sections are created with Softube’s patented modeling technology and gives a faithfully recreated sound of a “real” tube preamp.

So, think of Tube Delay as a tube preamp with an added delay section. The tubes in the delay section deteriorates the signal and you can even add more distortion after the delay before blending it with the direct signal passing through the preamp. The effect is perfect for fattening vocal tracks, adding slapback echo to drums or getting a gritty delay for guitars!





1. Delay time
2. Feedback
3. Dry/Wet mix
4. Treble
5. Bass
6. Direct (Dry) drive
7. Delay (Wet) drive
8. Output volume
9. Tempo sync

User Interface

Mix Controls the balance between the delay and the direct signal. Outputs only the original signal through the tube preamp section when set on **DRY**, and only the delay effect with added tube saturation when set to **WET**.

Set this control to blend the wanted amount of delay with the dry signal.

Remember! Since there are drive controls for both wet and dry signal paths that affects the volume, you might need to readjust the **Mix** after changing the **Drive** controls.

Direct Drive

(top knob) Controls the amount of tube drive in the dry signal path. Because the dry signal path is really a model of a tube preamp, you can use the dry

signal path to color or distort audio passing through it like you might use a “real” tube preamp.

Try moving the **Mix** control to full left and notice the coloring of the dry signal path when adjusting **Direct Drive**. This can be useful to just add some warmth to a single track or even a whole mix.

Delay Drive

(bottom knob) Controls the amount of tube drive in the wet signal path. Use **Delay Drive** to add color or distortion to the delay signal. This can make the delay signal stand out in a mix and give it a sound that ranges from organic to smashed.

Delay Drive affects the signal after the delay line feedback loop and will thus add distortion to all repetitions of the delay.

Delay Time and Tempo Sync

Controls the length of the delay in milliseconds or syncs to host tempo. The total length of delay after the original signal in Tube Delay is one second. In millisecond mode (**Tempo Sync OFF**), **Delay Time** will adjust the time from 1 to 1000 ms. The first half of the control goes from 1 to 100 ms, the second half goes from 101 to 1000 ms. This gives tweaking in the 1 to 100 ms range a greater resolution.

Common **Delay Time** settings are around 40 ms for a doubling effect and roughly 100 to 250 ms for a slapback echo.

When the **Tempo Sync** switch is ON (the “up” position), **Delay Time** is synced to the tempo setting of the host application. The **Delay Time** knob sets the length of the delay in these fractions of a measure: 1/16, 1/12, 1/8, 1/6, 1/4, 1/3 and 1/2. The maximum delay time is still one second though. To get to these values directly, simply click the numbers around the knob. It’s also possible to get values in between the fractions by adjusting the knob.

Some hosts don’t support reporting Tempo information to the plugin, in these cases a warning will be displayed.

Tempo Sync Here are some examples of the resulting delay time for a given host tempo when the **Tempo Sync** switch is set to ON (upwards position).

	60 BPM	80	100	120	140
1/16	250 ms	188	150	125	107
1/12	333	250	200	167	143
1/8	500	375	300	250	214
1/6	667	500	400	333	286
1/4	1 s	750	600	500	429
1/3	(1 s)	1 s	800	667	571
1/2	(1 s)	(1 s)	(1 s)	1 s	857

1/12 is the equivalent of an eighth note triplet.

1/6 is a quarter note triplet (a common reggae/dub delay rhythm).

1/3 is a half-note triplet.

Feedback Controls the amount of delay repetitions. The higher this setting is, the higher the amount of delay is returned to the input of the delay line.

Feedback goes from no repetitions (just a single delay) to a full feedback with never-ending repetitions.

Tone Settings - Bass

and Treble Controls equalization of the input of the effect. This affects both dry and wet signal paths and thus also the amount of distortion.

Output Volume Set the level of the output of the plug-in.

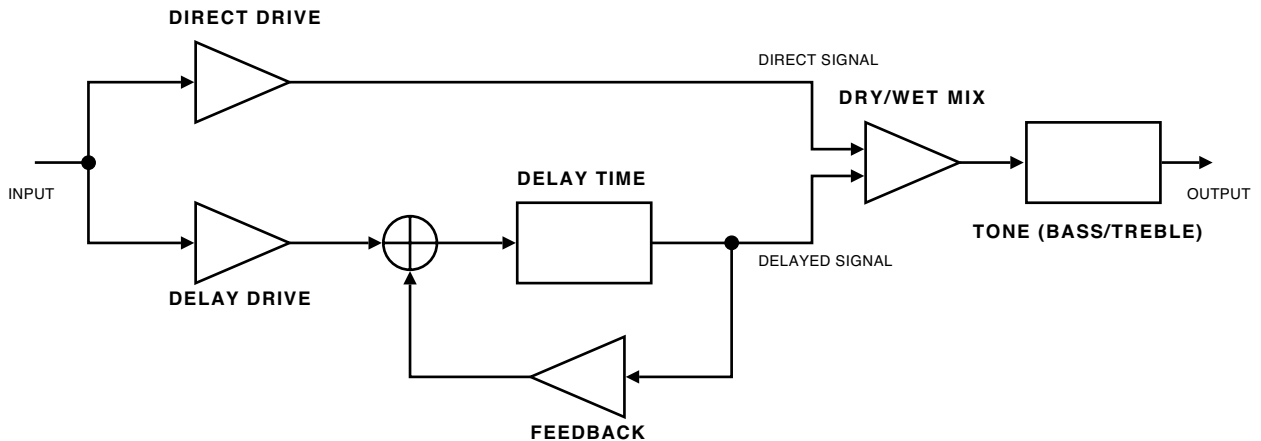
Block Diagram

Block diagram of the Tube Delay effect. As you can see, the dry signal (direct signal) isn't very dry, it is affected by both the tone stack and the tube circuits in the **Direct Drive** knob.

For simplicity, the **Tone** control has been depicted as the last stage after the **Mix** knob. In reality, the **Tone** control is embedded within all tube stages, and will thus affect all distortion.

Credits

Torsten Gatu – sound design and framework programming. **Oscar Öberg** – modeling and framework programming. **Niklas Odelholm** – framework programming and graphic design. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – 3D rendering and graphics. **Henrik Midtgaard** – concept.



33 Tube-Tech CL 1B & CL 1B Mk II Compressors

Foreword by John G. Petersen

After several years of consideration, we decided that the time had come to introduce a plugin of our TUBE-TECH CL 1B.

The development of the plugin was initialized in November 2006, as a cooperation between Lydkraft, Softube and TC Electronic. Hearing the result, we found that Softube was able to reach the difficult goal of making a plugin of the CL 1B which came extremely close to the original sound and gave the user all the characteristics of the hardware. After very serious testing of the software, the plugin was released late 2007.

To take this project one major step further, we decided in early spring 2009 to release a VST/AU/RTAS version of the plugin, and once again placed the developing task in the hands of Softube.

We are very impressed by the skills of these young guys from Sweden and we feel excited that the quality of the CL 1B is now available for all workstation users.

We hope you will enjoy the excellence of the TUBE-TECH CL 1B plugin.

Yours sincerely,



*John G. Petersen
President, Lydkraft/Tube-Tech*



About the CL 1B

The hardware CL 1B differs from many other compressors in that the gainreduction element is made from a nonsemiconductor component, which in itself has a very low harmonic distortion and none of the nonlinearity problems involved when using most semiconductor elements. All parts of these equally important design choices have of course been painstakingly modeled when creating the CL 1B plugin.

Another thing that is special about the CL 1B is the **Attack/Release Select** switch which allows the user to switch between a manual and a fixed attack/release setting, but also makes it possible for the user to combine both the fixed and manual settings. This gives a feature not normally obtained in other compressors: In the combined mode the attack and release controls makes it possible to obtain complex program dependent release-time slopes where a fast peak results in a fast release and vice versa.



1. Gain
2. Ratio
3. Threshold
4. Meter select
5. VU meter
6. Attack time
7. Release time
8. Attack/Release select
9. Sidechain bus select

User Interface

Gain The **Gain** control is used to “make up” for the gain loss, which takes place when the unit is compressing. It is placed after the gain reduction circuit and therefore has no influence on the threshold setting. The **Gain** control is continuously variable from off to +30 dB.

Ratio The **Ratio** control varies the ratio by which the input signal is compressed. If the ratio selected is 2:1, and the input signal increases 10 dB, the output signal is only increased by 5 dB. The **Ratio** control is continuously variable from 2:1 to 10:1.

Threshold The threshold is the point where the compressor begins its action. It is defined as the point where the gain is reduced by 1 dB. The **Threshold** control is continuously variable from +20 dB to -40 dB.

Meter Select Select what the VU meter should display:

INPUT: The meter shows the input level.

COMPRESSION: The VU meter is showing the gain reduction.

OUTPUT: The meter shows the output level.

Please note that it does not show peak or true RMS, it is a VU meter and behaves just like the original unit.

The meter and the plugin is calibrated so that a sine wave showing 0 VU at the output corresponds to a -18 dBFS output signal. Correspondingly, a 18 dBFS sine at the input will show 0 VU if the meter is set at showing the input signal.

Attack Time The **Attack** control chooses how fast/slow the compressor responds to an increase in the input signal. It is continuously variable from 0.5 to 300 milliseconds.

Release Time The **Release** control sets how fast/slow the compressor responds to a decrease in the input signal. It is continuously variable from 0.05 to 10 seconds.

Attack/Release Select This switch selects how the compressor reacts to an increase (attack) or a decrease (release) of the input signal.

There are three settings of this switch:

FIXED: Attack time: 1 millisecond Release time: 50 milliseconds.

MANUAL: Attack time: From 0.5 to 300 milliseconds Release time: From 0.05 seconds to 10 seconds.

FIX./MAN: This setting combines the release times of fixed and manual mode. The attack time is as it is in the fixed mode.

The **FIX./MAN.** mode always has a fast attack, but it is possible to obtain a release time that depends on the input signal, for example get a fast release when the peak disappears, then superseded shortly thereafter by the release time selected by the **Release** control.

The time the peak disappears to the point where the selected **Release** time takes over, is dependent upon the setting of the **Attack** control. That is, the **Attack** control changes function from a pure attack control to a control of the delayed release with the same time range.

Turn the **Attack** control clockwise to increase the time before the **Release** control takes over. Decrease the **Attack** control to shorten the time before the **Release** control takes over.

This function is valid only if the time of the peak is shorter than the setting of the **Attack** control. If the peak of the program is longer, or if the **Attack** control is set at its minimum position, it will respond just as in the **MANUAL** mode.

The **FIX./MAN.** mode acts as an automatic release function with a constant fast attack time and fast release time for short peaks and longer release times for longer peaks. This settings is mainly intended for use on program material (overall compression).

Sidechain Bus Select This knob has two settings:

INTERNAL: Normal mode. The compression depends on the same material as is being compressed.

EXTERNAL: Use an external side chain (if your hosts supports it) to control the gain reduction.

In many plugin formats, such as RTAS, VST3 and AU, it is possible to use an external signal as input to the detector. This is very common if you for instance want to compress a bass line using the bass drum as the trigger to the compressor. In that case, the bass line will be compressed whenever the bass drum is hit.

Monitor the Gain Reduction VU Meter when you set the threshold, since the Input VU Meter will show the original input signal. Just work with the **Threshold** knob until you get enough gain reduction.

Even if an external side chain signal is present, you will have to activate the external side chain by setting the **Sidechain Bus Select** to EXTERNAL.

Suggested Applications

Here you will find suggestions on various applications of the Tube-Tech CL 1B compressor plugin. They are given as a convenient guide that enables you to familiarize yourself with the different aspects of using the compressor. We have not mentioned specific settings of the **Gain** and **Threshold** as they are dependent on the input levels. Instead we have specified how much compression in dB we feel is needed for the various examples.

These examples were taken from the CL 1B hardware manual, and are of course just as valid for the plug-in as for the real unit.

Overall Compression (Final Mix)

Compression needed: 3-4 dB
 Attack/Release Select: FIX./MAN.
 Attack: 2 o'clock
 Release: 10 o'clock
 Ratio: 9 o'clock

Standard Compression (Bass, Piano, Guitar, Keys and Vocals)

Compression needed: 4-5 dB
 Attack/Release Select: MANUAL
 Attack: 2 o'clock
 Release: 10 o'clock
 Ratio: 10-2 o'clock

Heavy Compression on Instruments (Line Guitar and Piano)

Compression needed: 10 dB
 Attack/Release Select: MANUAL
 Attack: 7 o'clock
 Release: 1 o'clock
 Ratio: 3 o'clock

Compression of Drums (Snare and Bass Drum)

Compression needed: 2-3 dB
 Attack/Release Select: FIXED
 Attack: –
 Release: –
 Ratio: 9-12 o'clock

Mono and Stereo Operation

In stereo mode, the gain reduction of the left and right channel is always linked, in order to reduce stereo image shifting. The gain reduction used will be calculated from a combination of the two channels, just as if two hardware CL 1B had been linked together using a sidechain bus.



CL 1B Mk II

Mk II version is enhanced with features not found in the original hardware:

Sidechain Low Cut Filters the low end from the sidechain signal. Please note that the main signal path is not filtered.

Parallel compression Sets the amount of parallel processing. For more information on Parallel compression, please check Bob Katz' article on page 392.

Credits

Arvid Rosén – modeling. **Oscar Öberg** – modeling and DSP programming. **Torsten Gatu** – framework and DSP programming. **Niklas Odelholm** – GUI and framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Lydkraft ApS.

Credits – Mk II Version

Paul Shyrinskykh – product management. **Kim Larsson** – DSP and framework programming. **Björn Rödseth**, **Kim Larsson**, **Patrik Holmström** – framework programming. **Niklas Odelholm** – graphic design. **Ulf Ekelöf** – 3D rendering. **Daniel Delviken** – marketing. **Maxus Widarson** – quality assurance. **Igor Miná** – user manual layout.

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34 Tube-Tech Classic Channel & Classic Channel Mk II

THE TUBE-TECH CLASSIC CHANNEL contains three different Tube-Tech products, the opto-compressor CL 1B and two Pultec clones; the PE 1C program equalizer (the “classic” Pultec) and the ME 1B mid-range equalizer. On their own, these products immediately and easily produce a beautiful result, and combined they become an extremely powerful tool that lets you shape the sound of a track, bus or the whole mix.

As well as providing these beauties as separate plug-ins, we also chose to combine them into a single powerful plug-in—the Classic Channel. The Classic Channel lets you bypass or engage any of the units, as well as change their order. By bypassing them, they won’t take up any of your precious CPU, so you can might as well get used to always use the Classic Channel whenever you need just some CL 1B mojo or Pultec vibe. It makes it easier to add some eq or compression if necessary at a later stage...

Take the time to get to know the individual units and we assure you that you won’t get disappointed. There is something about these designs that just make your tracks sound just the way they should... And that’s just the reason to why we chose to emulate them.

A Note on Terminology

We have chosen to keep all the labels in the user interface faithful to the original units, even when discrepancies occur. The kilohertz label on the PE 1C is for example labeled “KCS” (kilocycles per second), while it’s labeled “kHz” on the ME 1B. Both the peak filters on the ME 1B are labeled as “*High Frequency*” and “*Low Frequency*”, while they are referred to in the manual as

“*High Mid Frequency*” and “*Low Mid Frequency*” in order to separate them from the real high and low frequency filters in the PE 1C.

User Interface

The user interfaces of the three individual units are the same as the individual plug-ins, so please see their respective chapter for more information.



PE 1C “Pultec” program equalizer

Primary use: Tame the top and low end. Sculpt the overall character of the sound. Use as a high frequency boost.



PE 1C “Pultec” Equalizer

ME 1B mid-range equalizer

Primary use: More exact sound sculpting than the PE 1C. Vocal and snare drum equalization in the 200Hz—7kHz range.



ME 1B Midrange Equalizer

CL 1B opto compressor

Primary use: It’s a compressor...



CL 1B Opto Compressor

Bypass and routing panel

The routing panel is used for engaging or bypassing the separate units, and also to select the processing order.

Program EQ In Engages the PE 1C “Pultec” program equalizer. Up position: **ENGAGED**. Down position: **BYPASS**.

Mid EQ In Engages the ME 1B mid range equalizer. Up position: **ENGAGED**.

Compressor In Engages the CL 1B compressor. Up position: **ENGAGED**.

EQ Before Compressor Compressor Before EQ

Selects if the equalizers should process the sound before the compressor (default position) or the other way around. Left position: **EQ BEFORE COMPRESSOR**. Right position: **COMPRESSOR BEFORE EQ**.

It is also possible to click the lamps of each unit to engage or bypass them.

1. Program EQ in
2. Compressor in
3. EQ before compressor/Compressor before EQ



Gain Staging

You might get confused by the three different gain controls on the units (PE 1C: **Output Gain**, ME 1B: **Output Gain** and CL 1B: **Gain**). All these volume controls are controlling the output volume of each unit.

If the equalizers are inserted before the compressor, their **Output Gain** controls will affect the **Threshold** and gain reduction of the compressor. If they are inserted after the compressor, they will of course not affect the operation of the compressor.

The best way to use these volume controls is to make sure the apparent audio level stays approximately the same when you bypass the unit. That way it will be easier to bypass and compare the audio, and also to switch order of the compressor and equalizers.

CL 1B Compressor Gain Staging

Just like the stand-alone CL 1B plug-in, the Classic Channel's CL 1B is calibrated so that a -18dBFS signal reads 0 VU.

Sidechain and the Equalizers

No, the equalizers does not affect the external sidechain of the compressor.



Tube-Tech Classic Channel Mk II

MK2 version is enhanced with features not found in the original hardware:

Mid/Side Enables the Mid/Side processing. Please note: when in MS mode, the color of the knobs will change to indicate the current parameter set.

Link Channels Links the parameters of both channels. Please note: when unlinked, it will be possible to equalize and compress LR or MS channels

separately. It's recommended to unlink the channels in Mid/Side mode.

Left/Mid Selects the correspondent Left or Middle channel.

Right/Side Selects the correspondent Right or Side channel.

Sidechain Low Cut Filters the low end in the sidechain signal. It's particularly useful in mastering applications to avoid "pumping" caused by low frequencies.

Parallel Compression Sets the amount of Dry and Wet signal. Please note: the parameter sets the amount of parallel processing for the compressor only, the EQs will not be affected.

Individual Bypass Each module can be bypassed by clicking on a red lamp. Shift-clicking on any bypass (red lamp) will bypass all 3 modules.

Credits

Niklas Odelholm – modeling, **Arvid Rosén** – modeling. **Torsten Gatu** – framework programming. **Oscar Öberg** – modeling. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Tube-Tech/Lydskraft ApS, Denmark.

Credits – MK II Version

Paul Shyrinskykh – product management. **Kim Larsson** – DSP and framework programming. **Björn Rödsseth**, **Kim Larsson**, **Patrik Holmström** – framework programming. **Niklas Odelholm** – graphic design. **Ulf Ekelöf** – 3D rendering. **Daniel Delviken** – marketing. **Maxus Widarson** – quality assurance. **Igor Miná** – user manual layout.

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35 Tube-Tech ME 1B Midrange Equalizer

Introduction

“Why do I need another eq?” Well... That is certainly not a new question. But for some reason the Pultec design of equalizers have managed to survive through the decades. The original Pultecs were built in the fifties and Tube-Tech has been making their Pultec clones since the mid eighties. Since the first Pultecs came out there have been tons of inventions that could have rendered this design useless: transistors, op-amps, circuit board designs, fully parametric designs, total recall, and the list goes on. But for some reason, the Pultec character is still desired in pro studios all over the world.

So do you need another eq? There are certainly stuff going on in the Tube-Tech ME 1B that is unique, but its biggest mojo is that it is so well-designed that almost whatever setting you use, it will sound good. And that is good product design, and probably the reason to why everybody needs a Pultec style equalizer.

About the ME 1B

Just like the PE 1C Pultec equalizer, the Tube-Tech ME 1B is a passive, tube-based equalizer. It was designed to accompany the PE 1C equalizer, and provide control over the frequencies that the PE 1C doesn't adjust. On its own it's a great tool to shape the mid-range of any audio material, and together with the PE 1C it becomes a versatile and full featured equalizer.



The focus of the PE 1C, “program equalizer”, is the shelving filters and the mid-range boost. It cannot attenuate the mids, and you don't have the ability to get bell shaped filter curves in the lows and highs. The ME 1B solves all these drawbacks by offering the user three bands of equalization: low frequency boost (200–1000Hz), an almost full range sweepable cut (200–7000Hz) and a high frequency boost (1.5–5kHz).

Together they are strong.

User Interface

The user interface of the ME 1B is pretty straight forward. You have three different sections, the **Low Frequency Peak**, **Mid Frequency Dip** and the **High Frequency Peak**. The only confusing thing is that the mid section never boosts, but always attenuate the selected frequency. The other two sections, low and high frequency, boost the frequencies. But as soon as you start to use the ME 1B in conjunction with the PE 1C you'll find that these sections will complement the PE 1C.

In addition to this you get an **Output Gain** control. It isn't included in the original hardware, but is useful in a plug-in.



Low Frequency Section

Mid Frequency Section

High Frequency Section

- | | |
|--------------|----------------|
| 1. Low freq. | 5. High freq. |
| 2. Low peak | 6. High peak |
| 3. Mid freq. | 7. Output gain |
| 4. Mid dip | |

Low Frequency Section

The low frequency section consists of a sweepable bell shaped peak filter and a gain knob.

Low Frequency Sets the center frequency of the peak filter. The frequency settings are 200 Hz, 300 Hz, 500 Hz, 700 Hz and 1 kHz. All these frequencies overlap with settings in the **Mid Frequency**.

Low Peak The right hand knob adjusts the amount of gain in the peak filter. Adjustable from 0 dB to about 10 dB.

Mid Frequency Section

The mid frequency section consists of a sweepable and attenuating bell shaped filter and a gain knob. With the gain set at 10 you get full attenuation (about -10 dB) from the mid filters.

Mid Frequency Sets the center frequency of the dip filter. The frequency settings are 200, 300, 500, 700, 1000 Hz (overlaps the low frequency section), 1.5, 2, 3, 4, 5 kHz (overlaps the high frequency section) and finally 7 kHz.

Mid Dip Sets the amount of attenuation at the selected frequency. Ranges from 0 dB to about -10 dB.

High Frequency Section

The high frequency section consists of a sweepable bell shaped peak filter and a gain knob.

High Frequency Sets the center frequency of the peak filter. The frequency settings are 1.5, 2, 3, 4, and 5 kHz.

High Peak Sets the amount of gain in the peak filter. Adjustable from 0 dB to about 8 dB. The exact gain depends on the frequency setting.

Output Section

Output Gain The output gain wasn't included in the original ME 1B, but it is pretty useful to have a gain control, especially when you want to A/B between bypassed and engaged mode. Ranges from -10 dB to +10 dB.

Filter Graphs

Here are some examples of the filter curves for different gain settings.

It is worth to note that the exact shape of these curves depend on three things:

1. What other knobs are set at (they depend on each other).
2. The frequency of the peak or dip filter.
3. The output impedance of the amplifier that is driving (inserted before) the real ME 1B.

In order to satisfy most setups, we chose to model the impedance so that you can get a little bit more gain or attenuation from the filters. That way you can always back it off a bit, if you need the plug-in to match exactly to your hardware setup.

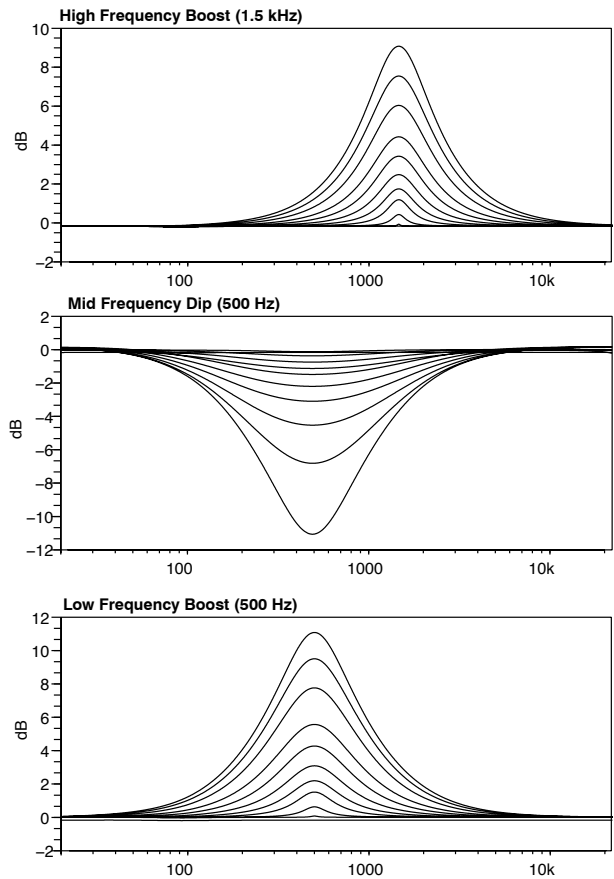
Buying Recommendations

The best way to get as close as possible to the original Pultec Midrange EQ sound in hardware is to get the ME 1B from Tube-Tech. They make great gear. Period. Every time we get to borrow Tube-Tech stuff we immediately fall in love and want to keep it.

Credits

Niklas Odelholm – modeling, **Arvid Rosén** – modeling. **Torsten Gatu** – framework programming. **Oscar Öberg** – framework programming. **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Tube-Tech/Lydkraft ApS, Denmark.

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36 Tube-Tech PE 1C “Pultec” Equalizer

Foreword by John G. Petersen

In the eighties I got really fascinated by the design and quality of classic tube processors such as Pultec and Teletronics. I felt there was a need to continue that tradition. TUBE-TECH Program Equalizer PE 1A (now PE 1C) was my first unit in production, and it started me off designing the whole TUBE-TECH range. We are excited to experience the PE 1C in the digital domain and impressed once again by the skills of Softube.

Yours sincerely,

John G. Petersen
President, Lydkraft/Tube-Tech

Introduction

“*Tube-Tech is Pultec.*” This is a quote from when the very first Tube-Tech product, the PE 1A, was promoted back in 1985. The vintage US made Pultec EQ 1P (designed and sold during the fifties) was the inspiration for Tube-Tech designer and founder John Petersen who was determined to produce a modern unit that could capture the magic of the original Pultec. The final testing, before shipping the PE 1A, was the EQ 1P and the PE 1A on different channels on a stereo track, making sure that each and every setting behaved exactly the same.

Today, decades later, the PE 1C (with minimal changes from the PE 1A) is still in production and the famous PE 1C has now entered the digital world with the PE 1C plug-in!

About the PE 1C

The Tube-Tech PE 1C is a passive, tube-based equalizer suitable for many different sources ranging from bass drums and guitars to vocals.

The equalizer features a Low Frequency section made up of two low shelf filters which can be combined to



attenuate and boost at the same time. These filters are actually bit apart in frequency even if they are controlled by the same frequency selector. The High Frequency section has a peak and shelving filter working in parallel, which provides a smooth top end while not getting too bright.

The ability to combine the different filters and sections is what gives the PE 1C its characteristic sound and is something which plays a crucial part in explaining the classic Pultec sound.

User Interface

The user interface of the PE 1C, or of a Pultec for that matter, can be confusing if you haven't used a Pultec before. First you have a low frequency section where you select the frequency with one knob, and boost and attenuate frequencies with two different knobs.

The high frequency section is made up of two subsections, the boost and the attenuation section. If you want to boost high frequencies, you select the frequency with the **High Frequency** knob, the width or "Q" of the bell filter with the **Bandwidth** knob, and then the amount of boosting with the **Boost** knob. You'll get a bell shaped boost filter for mid to high frequencies.

But for cutting high frequencies you only get three frequency choices selectable with the **Atten Sel** knob. The amount of high frequency attenuation is dialed in with the **Atten** knob. This section gives you a high shelving type filter.

In addition to this you get an **Output Gain** control. It isn't included in the original hardware, but we thought it might be useful in a plug-in.

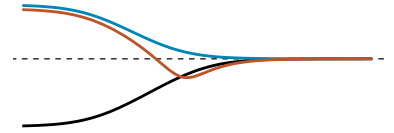
Low Frequency Section

Boost The **Boost** knob goes from 0 to 14 dB and controls a low shelf filter with a cut-off frequency that is actually a little lower than the ones that are printed on the panel. This gives the unit a unique frequency response when combined the the **Atten** setting.

Atten The **Atten** knob will attenuate the signal 0 to -18 dB using a low shelf filter.

Low Frequency Sets the cut-off frequency for the low frequency **Boost** and **Atten** controls. The frequency settings are 20 Hz, 30 Hz, 60 Hz and 100 Hz.

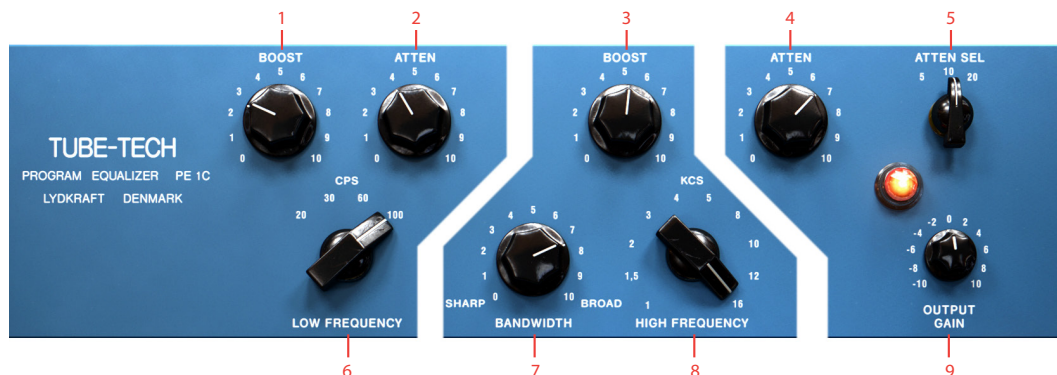
By increasing both the **Boost** (blue curve) and **Atten** (black curve) knobs at the same time, you'll get the very Pultec specific sound (red curve), sometimes referred to as the "Low End Trick", which is commonly used for equalizing bass drum. Tube-Tech has some nice instruction videos about this on their web site.



High Frequency Section

Boost Controls the gain for a boost peak filter that goes from 0 to 18 dB for sharp bandwidth and 0 to 10 dB for broad bandwidth.

Bandwidth Sets the width, or Q_c of the **Boost** peak filter. Goes from SHARP to BROAD.

*Low Frequency Section**High Frequency Section**High Frequency Attenuation*

- | | |
|---------------------------|-------------------|
| 1. Boost | 6. Low frequency |
| 2. Atten | 7. Bandwidth |
| 3. Boost | 8. High frequency |
| 4. Atten | 9. Output gain |
| 5. Atten select frequency | |

High Frequency Sets the cut-off frequency for the high frequency **Boost** control. The frequency settings are 1 kHz, 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, 5 kHz, 8 kHz, 10 kHz, 12 kHz and 16 kHz.

High Frequency Attenuation

Atten The Atten knob will attenuate the signal 0 to 18 dB using a high shelf filter.

Atten Sel Sets the cut-off frequency for the high frequency Atten control. The frequency settings are 5 kHz, 10 kHz and 20 kHz.

Output Gain Section

Output Gain The Output Gain knob controls the output volume from the plug-in. The range of the output volume is -10 dB to 10 dB.

Buying Recommendations

The best way to get as close as possible to the original Pultec sound in hardware is to get the PE 1C from Tube-Tech. As mentioned on Tube-Tech's web page, the sound of a PE 1C is exactly like the sound of a brand-new Pultec—not a dusty, old, and unserviced Pultec. The difference is that a meticulously serviced Pultec has a wide and open frequency response, without graininess or harshness. The PE 1C is not a dull and muddy old school equalizer, it's in fact pretty much the opposite!

Anyway, Tube-Tech makes great gear. Period. Every time we get to borrow Tube-Tech stuff we immediately fall in love and want to keep it.

Credits

Arvid Rosén – modeling, **Oscar Öberg** – DSP programming, **Torsten Gatu** – framework programming, **Niklas Odelholm** – DSP programming, **Ulf Ekelöf** – 3D rendering. Original hardware was designed by **John G. Petersen** at Tube-Tech/Lydkraft ApS, Denmark.

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37 Tube-Tech Equalizer Collection

Tube-Tech Equalizers Mk II is a combination of the legendary Tube-Tech equalizers PE 1C and ME 1B. We combined them in one unit for better ergonomics and workflow providing perfect tone shaping solution for mixing and mastering. For more information about PE 1C please go to page 216. For more information about ME 1B please go to page 220.

Credits – Mk II Version

Paul Shyrinskykh – product management. **Kim Larsson** – DSP and framework programming. **Björn Rödsseth**, **Kim Larsson**, **Patrik Holmström** – framework programming. **Niklas Odelholm** – graphic design. **Ulf Ekelöf** – 3D rendering. **Daniel Delviken** – marketing. **Maxus Widarson** – quality assurance. **Igor Miná** – user manual layout.



38 Valley People Dyna-mite

IF YOU HAVEN'T USED THE DYNA-MITE BEFORE, you will feel extremely confused.

DON'T PANIC!

As soon as you have acquainted yourself with the slightly weird layout and panel labels, you will learn how to master this powerful tool, and soon you will feel comfortable in knowing that you belong to the music industry's version of the "Trekkies", ie., the Valley People fanatics.

Take your time to look through this chapter to get acquainted with the normal modes of operation). During the learning period you will go through these steps:

1. Utter confusion. Desperation. Did I really pay for this?
Solution: *Read “Basic Limiting”, “Basic Expanding” and “User Interface”. It’s just three pages. Listen to the sound examples on the Softube web site.*
2. Pride. You have learned what the controls do. You think you master every aspect of it.
Solution: *Read the entire manual, as well as the manual for the original hardware (it’s included in the installer).*
3. Awe. You realize all the potential that is in this little thing, and start to think about all the cool things you can do with it.
Solution: *Do it!*

But don’t despair. It doesn’t take much time to learn how to use it, it’s just that the first 15 minutes can be a bit confusing.

Final word from the developers: *Expanding is the new black!*

About the Valley People Dyna-mite

The Dyna-mite was designed for two main purposes: **limiting** and **expanding**. You choose limiting or expanding by setting the **Mode** switch in either the **LIMIT** or **EXP** mode.

Threshold, Release and **Output** works as in any limiter or expander. **Range** is a nice addition that enables you to limit the maximum amount of gain reduction.

You can set the type of detector with the **Detector Type** switch (the switch closest to the **Range** knob). **AVG** is smoother and slower than **PEAK**. **GATE** is a mode that works best in the **EXP** mode.

With the **Det Source** switch you select if you want to use the normal, internal, source (**INT**) or use an external side-chain (**EXT**). With an external source you can do many fun things, including **keying** and **ducking**.

You will always get a brief explanation about the current mode in the “Modes Display” beneath the unit.

User interface



- | | |
|--------------|----------------------------|
| 1. Threshold | 5. Detector source buttons |
| 2. Release | 6. Gain reduction meter |
| 3. Range | 7. Mode buttons |
| 4. Output | 8. Detector type buttons |
| | 9. Clip meter |

Parameters

Threshold Adjusts the level at which the Dyna-mite starts to expand or limit.

Release Adjusts the time it takes to restore the gain after expansion/limiting.

Detector Source Set it to INT (normal operating mode) as a start. DS-FM and EXT are advanced modes.

Mode Selects main mode.
EXP=Expander/Gate, OUT=Bypass,
LIMIT=Limiter.

Detector Type Three different ways to detect the signal. Play around and try them out.

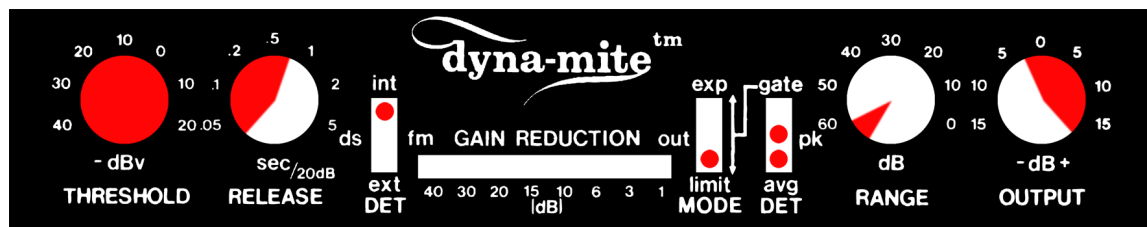
Range Sets maximum amount of gain reduction. Advanced. Leave at 60 dB as a start.

Output Sets output volume. Lower the volume if the output clips to much.

Gain Reduction Meter Displays the current amount of gain reduction.

Clip LED Lights up when the output signal is clipping.

For more detailed explanations of each knob, please see the following chapters.



Basic Limiting

Limiting is a hard Inf:1 ratio compression that can be used to increase the apparent volume, or to even out dynamic differences in a track. It works by reducing the gain for passages that are louder than the selected threshold. The limiting mode on the Dyna-mite is often used to squash drum tracks.

Mode: LIMIT (limiting)

Detector Source: INT (internal source)

Detector Type: AVG (slow attack) or PEAK (fast attack)

Threshold: Adjust to set amount of limiting (as read from the GAIN REDUCTION meter).

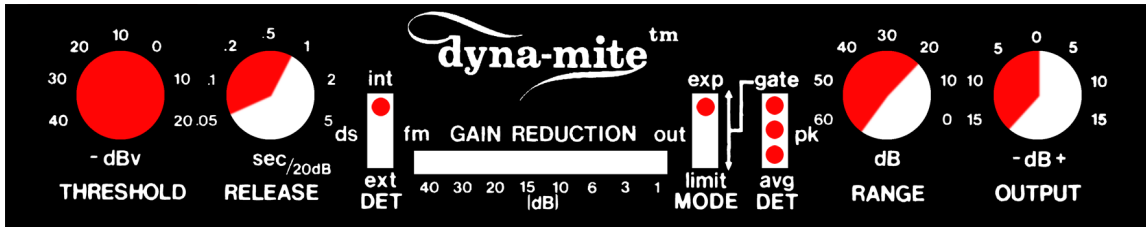
Release: Adjust to set release time

Range: 60 dB

Output: 0—15 dB

Procedure

1. Set **Threshold** until you get the desired amount of limiting, as red on the LED array.
2. Set the **Output** volume until you get the desired output volume.
3. Set the **Release** control by ear. If you get too much pumping, increase the release time.
4. Toggle the **Detector Type** between AVG and PEAK to get different attack times.



Basic Expanding

Expanding is when you let the Dyna-mite reduce the gain for passages that are below the select threshold. You can use it as a soft noise gate, use it more aggressively as a hard guitar/drum gate, or you can use to expand the total dynamics of a track.

Mode: EXP (expanding)

Detector Source: INT (internal source)

Detector Type: AVG (slow attack), PEAK (fast attack), or gate (hard noise gate, fast attack)

Threshold: Adjust to set the threshold of expansion

Release: Adjust to set release time

Range: 60—20 dB

Output: -15—0 dB

Procedure

1. Set the **Release** and **Range** to their min positions (CCW).
2. Adjust the **Threshold** so that the desired signals don't light up any LEDs (no gain reduction), while the undesired signals (noise) cause gain reduction (LED array lights up).
3. Set the **Release** control by ear: too fast release time can cause unwanted, abrupt, effects when the signal goes down below the **Threshold**.
4. Set the **Range** to get the desired amount of maximum attenuation.
5. Toggle the **Detector Type** (AVG, PEAK, GATE) to get different kinds of gating effects.

Switches

All three switches work independently of each other, so don't be alarmed. Once you know what each switch does, the Dyna-mite will be easy to operate.



Detector Source (INT, DS-FM, EXT)

Determines the source of the detector.

INT

Internal source, normal operating mode for **noise gating**, **expanding** or **limiting**. The gain reduction circuit acts upon the same signal that is fed to the detector.

DS-FM **Internal source**, but with a detector filter that **boosts the high frequencies**, thus making it easier for high frequencies to trig the detector.

This mode can be used for **de-essing**, or it can be used rather creatively when increased sensitivity to high frequencies is desirable.

EXT **External source**, or “**side-chaining**”. The detector is fed a separate signal (a side-chain) and calculates the gain reduction from this signal. This gain reduction is then applied to the main signal. This is used for **keying** (side-chain expanding) and **ducking** (side-chain compression) modes.



Mode (EXP, OUT, LIMIT)

Determines if the Dyna-mite should expand or limit the sound.

EXP **Expander mode**. Reduce gain for signals **below** Threshold.

OUT **Bypass**. Output volume knob and output stage clipping still active.

LIMIT **Limiting mode**. Reduce gain for signals increasing **above** Threshold.



Detector Type (GATE, PEAK, AVG)

This switch determines the ratio and attack times of the detector.

GATE **Fast and aggressive attack.** This is the most extreme mode. Fast attack time (about 50 μ s). In **LIMIT** mode, the ratio is negative (1:-20), which means that for every dB above threshold, the output signal will be reduced by 20 dB! In **EXP** mode, you'll have a hard noise gate with a 1:20 ratio. So for every dB below the threshold, the signal will be reduced by another 19 dBs.

PEAK Fast attack time (about 50 μ s), useful for hard limiting of transient material. Inf:1 ratio in **LIMIT** mode, and 1:2 ratio in **EXP** modes (for soft noise gating).

AVG Slow attack. A slower and more complex attack time (about 1–15 ms). Inf:1 ratio in **LIMIT** mode, and 1:2 ratio in **EXP** modes (for soft noise gating).

Detector Type=GATE can be confusing in the beginning, but you can see it this way: In **EXP** mode, the signal gets gated when the level drops below the threshold (that's normal). In **LIMIT** mode, the signal gets gated when the level goes above the threshold (that's weird).



Knobs

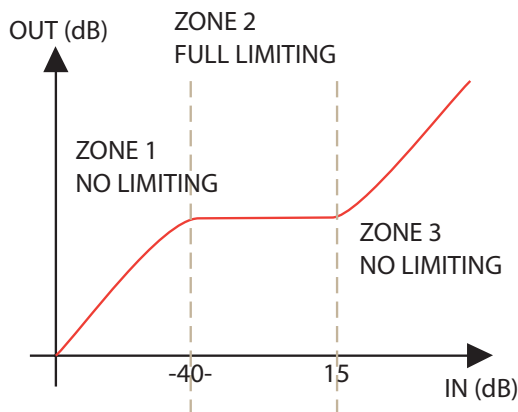
Threshold

Classic threshold control. Determines the signal level above which Limiting, or below which Expanding action begins.

Variable from -40 dBv to $+20$ dBv, which in a plug-in translates to about -60 dBFS to 0 dBFS. A full scale (0 dBFS) sine wave, with **Threshold** set at max ($+20$ dBv), will cause the Dyna-mite to just start to limit or expand. The exact values depends on the mode selected.

VCA Gain Coupling (a.k.a. Auto Make-Up Gain)

In internal limit and DS-FM modes, the Dyna-mite engages the VCA Gain Coupling which acts like an automatic make-up gain.



Range

This innovative control limits the **maximum gain reduction**. It varies from 0 dB (no gain reduction) to 60 dB **possible** gain reduction.

This control is very useful when expanding, gating or keying. If you use the Keying mode (EXT, EXP, AVG/PEAK) to, for example, duck a background music score whenever a narrator speaks, you can set the **Range** control to 15 dB to prevent the music to be reduced by more than 15 dB.

In the original hardware, the **Range** control was only available in the EXP mode and left out in the LIMIT mode. This was probably due to implementation difficulties, so we decided to introduce the **Range** knob to the other modes as well. (To get the plug-in to behave exactly like the hardware, just set the **Range** knob back to 60 dB in the limit modes.)

Advanced: Range Knob in Limiting Mode (Zones)

When limiting, the **Range** control gives you a very powerful tool to create a “zone” in which the limiting occurs. If you for example limit a drum track and the Gain Reduction meter reads 40 dB in the peaks, you can set the **Range** control to 25 dB and thus get three different zones:

1. No limiting when the drums are below -40 dB
2. Inf:1 limiting in the range -40 dB to -15 dB
3. No limiting for peaks exceeding -15 dB

Since the peaks in the third zone would cause a gain reduction above the limit of 25 dB, and the **Range** is set at 25 dB, no further gain reduction will occur. The effect is that you have a very effective limiting in zone 2, but with the transients of zone 3 being let through. It's different, but sounds somewhat similar to that of a parallel or “*New York style*” compression.

Release

Classic release time control. Determines the rate at which a gain is restored after Limiting or Ducking, as well as the rate at which gain is reduced after Gating, Keying or Expansion attacks.

Variable from 50 ms to 5 sec/20dB.

Anticipatory Release Computation. With fast release time settings and certain material, the gain reduction will follow the signal envelope too tightly, and “pumping” will occur. To escape these effects, yet still allow the user to select rapid release times, Valley International developed a proprietary circuit scheme known as Anticipatory Release Computation (ARC). It analyzes the program input and anticipate conditions which would cause either waveform gain modulation or excessively rapid pumping, and computes a different release envelope in order to prevent, or greatly diminish, these effects. Technical explanations apart, the ARC circuit is definitely a big part of the famous Dyna-mite sound.

Output

Output volume control. Ranges from -15 dBv to +15 dBv. In limiting modes (LIMIT and DS-FM), the output volume will be automatically corrected to fit with the chosen **Threshold** level. In all other modes, the **Output** volume will act as a simple gain control. You will soon find that this auto make-up feature is very handy, since it is extremely easy to make changes to the amount of limiting (with the **Threshold** control) without disturbing the balance of the mix.

Clip Indicator

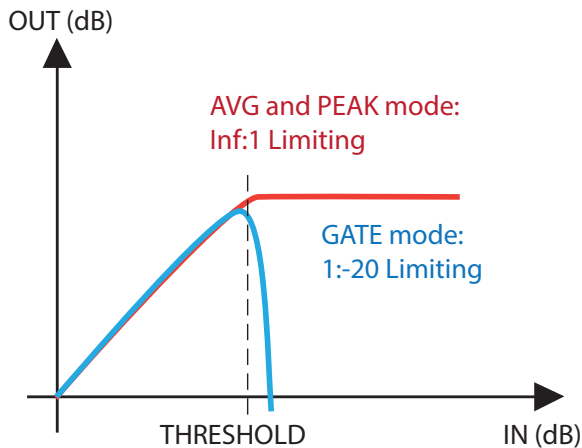
The plug-in features an accurate model of the output stage in the hardware. So whenever the hardware would clip, the plug-in will as well. This is indicated by a clip LED, and just as with the hardware, it is possible for the clip LED to flash, even with very heavy limiting, because the transients will in some cases slip through (for example in the AVG mode).

The original manual states: “*The clipping signal is [often] of exceedingly short duration, and is, in all probability, fully inaudible. Any decision to lower the output level because of sporadic flashing of the clip LED should be based upon what you hear. If the signal sounds “clean”, you are probably better off to leave the output level alone.*”

Times change, and today we often strive for a “dirtier” sound with distortion and clipping, so feel free to crank up the **Output** volume and experience the brutal distortion of the original Dyna-mite hardware...

Limiting Modes

There are two basic limiting modes: the **AVG** and **PEAK** detection. On top of this, you can use the built-in pre-emphasis high frequency filter (**Detector Source** = **DS-FM**) together with the internal signal. You can also use an external side chain to trigger the limiter (**Detector Source** = **EXP**), which is usually called **ducking**. Finally, you can set the **Detector Type** to **GATE**, which will give you **negative limiting**.



In theory, the difference between **AVG** and **PEAK** lies basically in the way the detector calculates the envelope of the signal. **AVG** uses *Linear Integration Detection*, a proprietary method that Valley International developed, while **PEAK** uses a more traditional “log-of-the-absolute-value” method.

In practice, the differences lies in the timing. **AVG** is a bit slower, has a more natural sound, but can be tricky to work with. **PEAK** is faster and behaves more like a traditional limiter.

You can always toggle between the **PEAK** and **AVG** modes and listen to the difference.

Classic Limiting

Mode: LIMIT

Detector Type: AVG/PEAK

Detector Source: INT

Threshold: Adjust to material

Release: Usually short, 0.05-0.5 sec

Range: Typically 60 dB (see section about “Zones” for usage)

Output: Adjust to material, typically above 0 dB.

These are the classic limiting modes. If the detector is in **PEAK** mode, you will be able to raise the total volume more than in **AVG** mode, but the waveforms will be more distorted than in **AVG** mode. In **AVG** mode it is easier to get a more transparent limiting, but it is also easier to get lots of punch in a drum track since the slower attack time will let the transients through. The original manual recommends the **AVG** position except when you have very transient sounds.

Level Limiting (INT, LIMIT, AVG)
Peak Limiting (INT, LIMIT, PEAK)

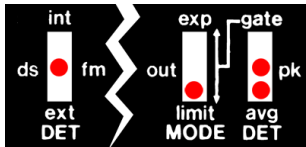
Ratio = Inf:1, Automatic

make-up gain during limiting. **Range** control forced to 60 dB on hardware unit, but active in the plug-in.

AVG mode is good for transparent limiting, or added punch to a bass line or drum track.

PEAK mode is good for hard limiting, for example adding length to a snare drum sound or to limit fast transients.





De-Essing and Classic Limiting With Pre-Emphasis Filtering

Mode: LIMIT

Detector Type: AVG/PEAK

Detector Source: DS-FM

Threshold: Adjust to material, normally about 6—10 dB of gain reduction.

Release: Usually short, 0.05-0.5 sec

Range: Usually 60 dB

Output: Adjust to material

This is a setting that's very useful for vocal de-essing, but also whenever you need to limit high frequency sounds (such as controlling cymbal levels).

The DS-FM filter only boosts the high frequencies on the signal that enters the detector. The main signal (the one you can listen to) is not affected by this filter.

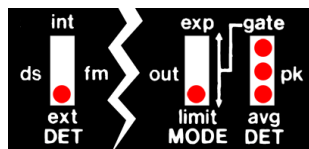
De-essing (DS-FM, LIMIT, AVG)

Limiting High Frequency Sounds (DS-FM, LIMIT, PEAK)

Ratio = Inf:1. Automatic make-up gain during limiting. High frequency EQ inserted in detector path. **Range** control forced to 60 dB on hardware unit.

AVG mode: Good for vocal de-essing. The original manual says: *"It is important to note that the use of Linear Integration Detection [ie. the AVG mode] provides a much more effective mechanism for de-essing than does the conventionally used peak detection."* The reason for this is that the AVG mode's slower response time won't limit transient high frequency sounds (like a 't'), but will limit longer high frequency sounds (like in 'sss').

PEAK mode: Good for drum track limiting, if you want to limit the cymbal sibilants. Setup a good limiting without using the DS-FM mode, and then toggle between INT and DS-FM to hear which one you like the best.



Side-Chain Limiting/Ducking

Mode: LIMIT

Detector Type: AVG/PEAK/GATE

Detector Source: EXT

Threshold: Setup the threshold level according to your side chain source level.

Release: Tune by ear. Classic voice-over usually needs longer release times than ducking strings ducking under a 4/4 bass drum.

Range: Use it! Set it to the desired maximum amount of ducking (in dB).

Output: –

A rather normal procedure is to let one track “duck” below another. For example ducking a synth pad by the strike of the bass drum (good use for the GATE mode). Another classic example is using it to duck background music below a voice-over (good use for the AVG mode).

Operating procedure (from the original manual): *“In setting up for the Ducking operation, one would normally start with the Range at maximum (CCW) the Release time at minimum (CCW), and the Threshold at maximum (CW). After applying both signal sources, the Threshold would be adjusted such that a reliable full duck were achieved on each external signal passage. The operator [that’s you!] would then adjust the Range control for the desired depth of duck, and the Release control for the desired recovery rate.”*

AVG/PEAK MODE: Inverse Envelope Follower (EXT, LIMIT, AVG/PEAK)

Signal gain is determined by the level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 1 dB decrease in signal gain.

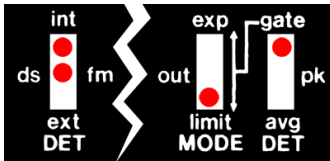
AVG mode gives the “nicest” and “friendliest” ducking mode.

PEAK mode gives a faster ducking that might crackle for some settings.

GATE MODE: Hard Ducking (EXT, LIMIT, GATE)

A 1 dB increase of the side chain signal level (over **Threshold**) causes a 20 dB decrease in signal gain. PEAK detection.

Extreme ducking. Each time the side chain signal goes just a teeny bit above **Threshold**, the main signal will be almost entirely gated. Can be quite effective together with a limited gain reduction, as set by the **Range** knob.



Weird Limiting

Mode: LIMIT

Detector Type: GATE

Detector Source: DS-FM OR INT

In the category “Weird Limiting” we can find the negative ratio stuff. The negative limiting mode is a very special case, and is normally used together with longer release times and the **Range** control. In some settings, specifically with long release times and a **Range** at about 10–15 dB, you can get a very crackling sound. Increase the **Range** or the **Release** time to avoid this.

Negative Limiting for “Organ Effects” (INT, LIMIT, GATE)
Modified Negative Limiting (DS-FM, LIMIT, GATE)

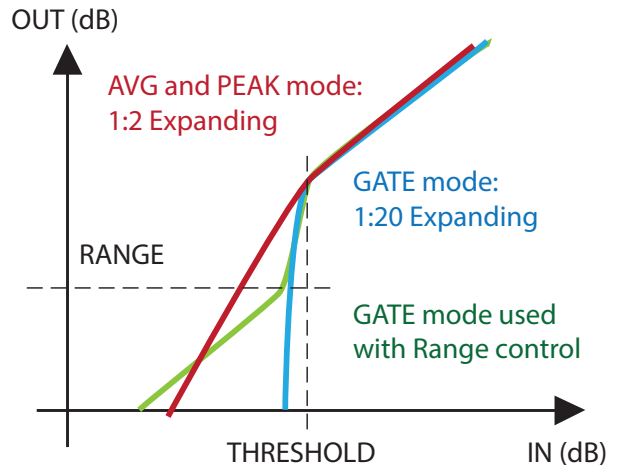
Ratio = 1:–20. As input signal exceeds **Threshold**, a 1 dB increase causes a 20 dB decrease in output level.

This is the mode that is the hardest to find some use for. It crackles and pops a lot, and one might suspect when reading the original manual that this is a mode that wasn't planned when designing the gear. More like “Wait, what happens if you use the gate together with the limiting mode? Ahh...”

Expanding Modes

The expansion modes is operated in a very similar way to the limiting modes, with the big exception that it is, well, expanding rather than limiting.

Another exception is that the **GATE** mode actually gates the signal in the **EXP** mode, while it in the **LIMIT** mode acts like a negative ratio limiter.



Classic Expansion/Noise Gating

Mode: EXP

Detector Type: AVG/PEAK/GATE

Detector Source: INT/DS-FM

Threshold: Adjust so that the desired sound levels extinguish all lights on the Gain Reduction LED array.

Release: Start with minimum setting, and then adjust “by ear”.

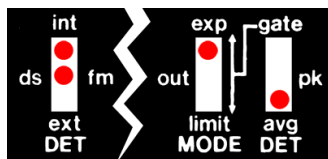
Range: Start with 60 dB, adjust to decrease the effect caused by the gating.

Output: Usually below 0 dB

In the classic noise gating mode you have three different settings of the **Detector Type**: **AVG**, **PEAK** and **GATE**.

The **GATE** mode is the most brutal mode, and is pretty efficient to use on drums or heavy metal guitar if you want a more aggressive sound, but it should be regarded more as a creative tool rather than a problem solver.

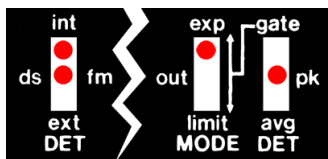
In all of these modes you can toggle between **INT** and **DS-FM** to get different weighting of the frequencies. Setting the **Detector Source** in **DS-FM** mode will make the expander trig more easily on high frequencies, which can be a good thing if you for example are gating a drum beat and want to avoid gating the high frequency cymbals. Since noise often has a lots of high frequency content, you should set it to int if you want to do actual noise gating. But you probably want to use it more creatively anyway...



AVG MODE: Soft Noise Gating (INT/DS-FM, EXP, AVG)

Ratio = 1:2. A signal x dB below **Threshold**, will be attenuated another x dBs.

The least obtrusive noise gate. Good for classic noise gating with sources that doesn't have strong transients (such as voice and strings). Use on slow to medium attack sounds.

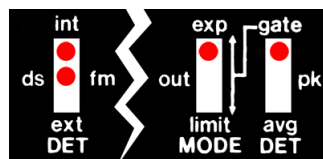


PEAK MODE: Soft Noise Gating with Faster Attack Time (INT/DS-FM, EXP, PEAK)

Ratio = 1:2. A signal x dB below **Threshold**, will be attenuated another x dBs.

If the attack of the instrument is moderate to fast, such as with drums or certain types of guitar, it is better to use this **PEAK** mode. The faster attack time in **PEAK** mode assures that the gain is fully restored when the transient strikes.

This mode can be pretty nice to use if you want to shorten the decay of a sound, such as a bass line or steel stringed acoustic guitar. With this mode you can get a standard Jazz Bass to sound like a damped Hofner string bass, or a steel stringed \$3000 Martin to sound like a \$30 banjo... Just set the **Release** time so that you get the right amount of decay.



GATE MODE: Hard Noise Gating (INT/DS-FM, EXP, GATE)

Ratio = 1:20, **PEAK** detection of input signal. A signal x dB below **Threshold**, will be attenuated another x dBs.

Use this setting as a traditional, boring hard noise gate, or... use it to create hard hitting beats from almost any drum track! Set a short release time and aggressive gating, and set the **Output** volume high to get some distortion. Stack up many Dyna-mites with different amounts of distortion and release times to get fat snare and bass drum sounds... Toggle the DS-FM switch to include/exclude cymbals.

Keying Modes

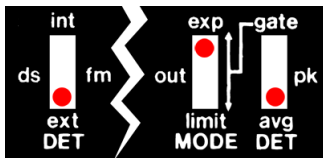
Mode: EXP

Detector Type: AVG/PEAK/GATE

Detector Source: EXT

While most people are accustomed to ducking, a.k.a. side chain compression—for example ducking a synth pad when the bass drum strikes, the art of keying is still somewhat of a hidden secret for most people. I mean, how many times have you heard Oprah talk about the benefits of keying?

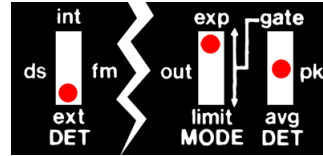
Keying is when you let your main track (say a bass line) follow the envelope of the side chain track (for example a bass drum). So with this example, the bass line will be gated whenever the bass drum is silent, and whenever the bass drum strikes, so will the bass line.



AVG MODE: Envelope Following Soft Keying (EXT, EXP, AVG)

Signal gain is determined by the level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 1 dB increase in signal gain.

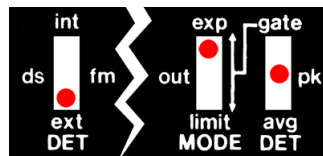
AVG mode has (as usual) a more natural sound, but will slow down the attack of the main signal. Use when you don't need a fast attack time when opening the gate.



PEAK MODE: Envelope Following Soft Keying with Faster Attack (EXT, EXP, PEAK)

Signal gain is determined by level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 1 dB increase in signal gain.

The extremely fast attack time will assure that the keyed signal has full gain when the attack of the side chain signal arrives, but this can sometimes cause the transient to “pop” or “crack” when the gate opens. This is usually not a problem if your mix consists of both the keyed and the side chain signal, since the real transient from the side chain signal will be audible at the same time as the (unwanted) “pop” sound.



GATE MODE: Hard Keying (EXT, EXP, GATE)

Signal gain is determined by level of the side chain signal. A 1 dB increase of the side chain signal level (over **Threshold**) causes a 20 dB increase in signal gain. **PEAK** detection.

Same fast attack times as the Soft Keying mode with **PEAK** detection, but a much more aggressive on/off type of gate. Before the age of the DAW, it was very useful when you needed to tighten up poorly performed background vocals or horn sections. Use the player with the best timing as key, and use the Hard Keying mode on the other players to assure that all off-time notes are being gated. Today, it's just a cool effect.

Mono and Stereo Operation

Inserting the Dyna-mite in **STEREO** mode makes it behave just as the real unit in “STEREO COUPLE” mode.

Buying Recommendations

The Valley People Dyna-mite is a very cool and affordable hardware limiter/expander. Many people don't know how to use it and say they don't like it just because they cannot operate it. Once you've learned the plug-in you will have no problem at all to operate the real hardware, so if you find a unit—make sure you pick it up!

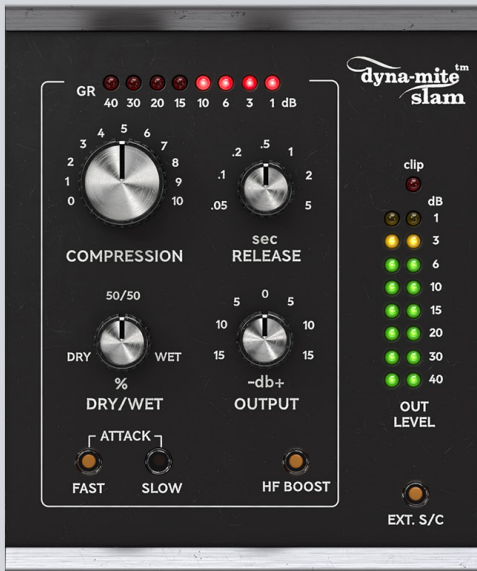
(The thing on top of the unit is a description of all the different modes, the same texts that are displayed in our nifty little display.)

Credits

Oscar Öberg – modeling, **Niklas Odelholm** – graphics programming, **Torsten Gatu** – framework programming, **Arvid Rosén** – modeling and framework programming, **Ulf Ekelöf** – 3D rendering. Original hardware was designed by Valley People Incorporated.

ALL VISUAL AND AURAL REFERENCES TO THE VALLEY PEOPLE DYNA-MITE ARE TRADEMARKS BEING MADE WITH WRITTEN PERMISSION FROM PMI AUDIO. THE VALLEY PEOPLE, DYNA-MITE AND ASSOCIATED LOGOS ARE TRADEMARKS OF PMI AUDIO GROUP, USED UNDER LICENSE. ALL SPECIFICATIONS SUBJECT TO CHANGE WITHOUT NOTICE. ALL RIGHTS RESERVED.





39 Valley People Dyna-mite Slam and Gate

THE ORIGINAL DYNA-MITE BY VALLEY PEOPLE IS ICONIC for both good and bad reasons. It was built to be a Swiss Army knife of dynamics, with a gate, expander, peak limiter and RMS limiter all built in to a single unit. It excelled at these tasks, being both the rawest and punchiest compressor out there, as well as the smoothest, most care-free gate/expander there is. That's one reason it's iconic.

The other reason is its terrible user interface. Engineers remember the Dyna-mite as "the thing that sounded awesome but was too scary to use on a session". Trying to cram that much functionality into a unit that was 1U high and half-rack wide made it very difficult to use. The combination of three 3-way switches also led to weird combinations, such as the -20:1 negative limiting setting. What's with that?

But that shouldn't take away the fact that the Valley People Dyna-mite sounds amazing. It can make a snare punchier than anything else, it can totally squash a drum kit, it's really (really!) a really good gate that you should use on your toms, and it can create tons of cool sounds. Too bad no-one dared to use it.

Fast forward to 2019. The futuristic AI-blockchain engineers work frantically with their 3D holograph "Minority Report" inspired Emacs terminals. They see the solution. A new user interface for the holy foundation of the Dyna-mite. It would not be sacrilegious, it won't change the essence of the divine circuits, merely it's face. They can do it, they will do it!

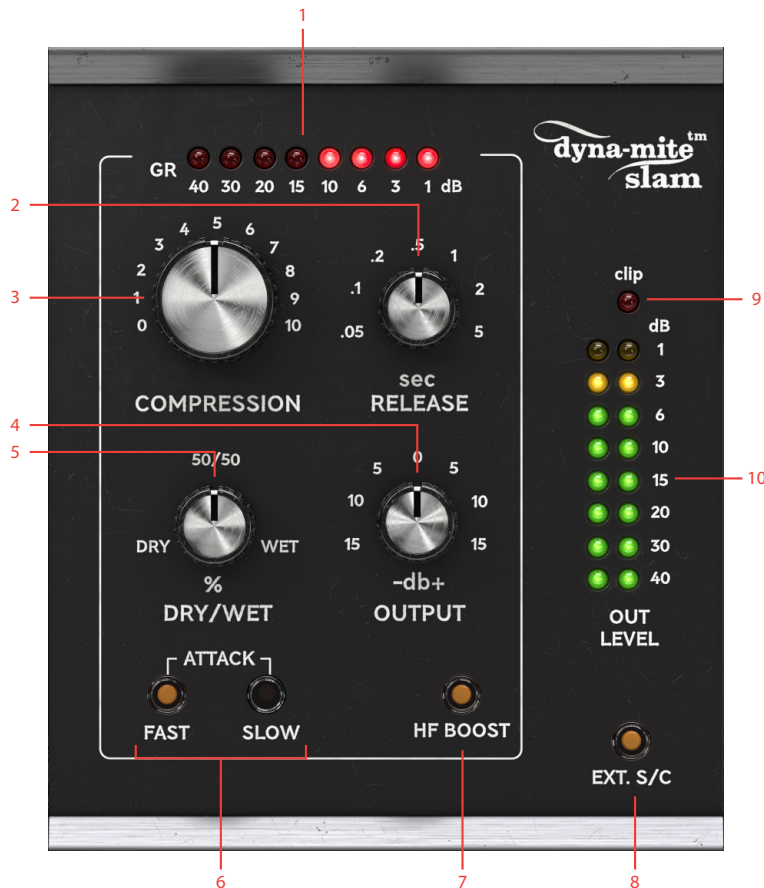
Dyna-mite Slam

Slam is the compressor/limiter part of the original Dyna-mite. The only thing that has been added is a parallel compression knob, because you know you want it. Away with the Range knob, and all the weird settings. This is as close to a one-knob compressor as Dyna-mite will ever become.

For more in-depth information on the Dyna-mite's operation, please see the Dyna-mite chapter.

User interface

There's one knob that's bigger than the others. That's the first one you should reach for.



1. Gain reduction meter
2. Release knob
3. Compression knob
4. Output knob
5. Dry/Wet knob
6. Attack Fast/Slow
7. HF Boost
8. Ext. S/C
9. Clip meter
10. Output meter

Parameters

Compression Increase Compression for more compression. This corresponds to the Threshold knob on the Dyna-mite, but backwards. Turn it up for more, down for less.

Release Sets the release time of the compressor. Shorter release means more aggressive pumping, a slow release makes it smoother.

Dry/Wet Decrease to blend in some of the unaffected dry signal. This is very useful when applying heavy compression, to mix some of the original transients back into the audio.

Output Sets the output volume. The Dyna-mite has automatic make-up gain, so you probably don't have to reach for this knob that much. Don't be afraid of overdriving the output, the built-in distortion is thick and funky, and very useful!

Attack Fast/Slow These two buttons adjust the attack time. They correspond to "pk" and "avg" on the original Dyna-mite.

HF Boost The HF Boost increases the high frequencies that the compressor's detector sees, which makes it trigger more easily for high frequen-

cies. If Dyna-mite Slam reacts too much on low frequency content, such as the kick drum on a drum bus, engage HF Boost to make it listen more to the top end. With HF Boost engaged, the compressor will react more naturally to the frequency content, since its internal detector will "hear" the sound more like our ear would hear it.

This corresponds to the "ds/fm" mode on the original Dyna-mite.

The HF Boost does not affect the audio path we listen to, just the audio that the detector uses to decide if the compressor should reduce gain or not.

Ext. S/C Engage the external sidechain (if available). The external sidechain disables the automatic make-up gain.

Gain Reduction Meter Shows the amount of gain reduction.

Output Meter LED meter showing the output volume.

Clip Meter The Clip Meter will turn on if the Dyna-mite is clipping the output signal. The output clipping has been carefully modeled from the original unit and can be used as a distortion effect.

In use

Slamming It

If you need to get something to happen with your drum bus, you might need to slam it. Set Attack to FAST, Release to .1 and increase Compression until you read 10-20 dB of gain reduction. Decrease Dry/Wet until you get enough of the original drums back into the audio.

Distorting It

The output will distort at roughly 0 dB, and with compression and auto make-up it might be difficult to hit that level. But there's a trick! If you enable "Ext. S/C", you will disable the auto make-up, and you can turn up Volume to get to that sweet distortion!

Punchy Drums

Punchy means a fat, energy-filled, attack with a lot of mass. That's what the Slow button does.

Click on Slow and increase Compression until you get 6-20 dB of gain reduction. You should hear how the first part of the sound (the transient) gets more attack while the rest of the sound gets lower in volume.

Use it on a snare drum to make it sound like the drummer is hitting the drum harder.

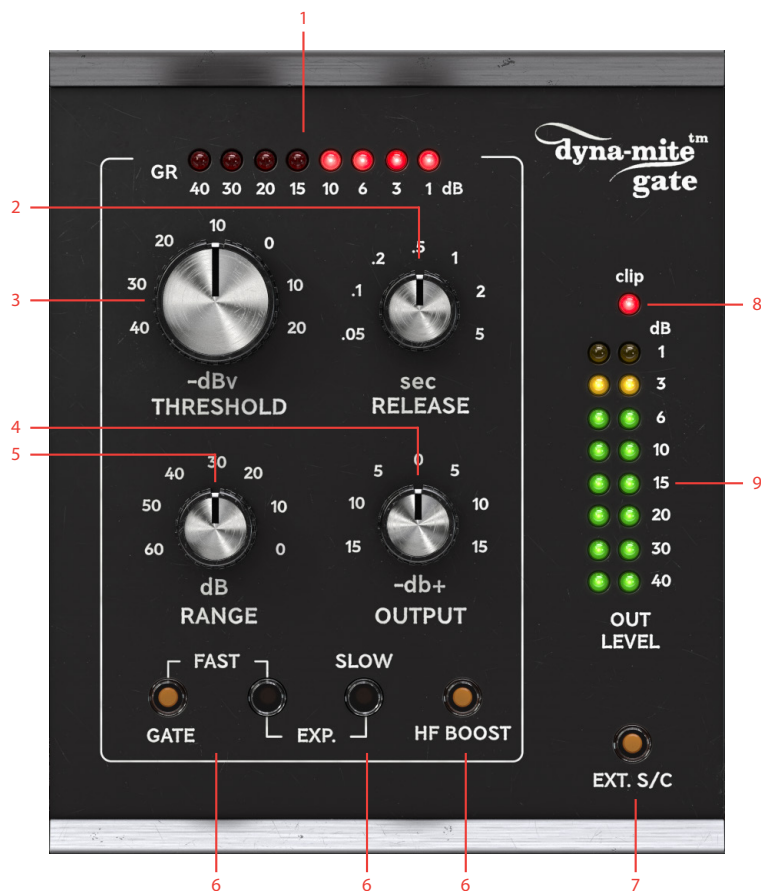
Dyna-mite Gate

The Dyna-mite Gate is a two-in-one expander and gate. Its "auto release circuit" makes it very musical, and it's as close to a set-and-forget gate as we've ever seen. The knobs should be pretty standard, but the sound is something more than just standard.

For more in-depth information on the Dyna-mite's operation, please see the Dyna-mite chapter.

User interface

As with Dyna-mite Slam, the Threshold knob is bigger than the other knobs because it's the main control. Decrease Threshold so that the gate opens up on your wanted sounds (for example a drum hit), but closes in the silences between hits. Make the release time short, but still slightly longer than the sustain of your sound. If you need total silence between hits, click the "FAST 1:20" button. If you need a smooth gate, click the "SLOW 1:2".



1. Gain reduction meter
2. Release knob
3. Threshold knob
4. Output knob
5. Range knob
6. HF Boost
7. Ext. S/C
8. Clip meter
9. Output meter

Parameters

Threshold Adjust the Threshold to change the sensitivity of the gate.

Release Sets the release time (the time it takes for the gate to close).

Range Sets the maxing amount of gain reduction. If you only need 10 dB of attenuation, set Range to 10 dB.

Output Sets the output volume.

Fast Gate Hard gating (1:20) with a fast attack (time it takes for the gate to open). Use for percussive sound where maximum attenuation is needed, with the risk of making the gate sound slightly artificial. This corresponds to the "gate" mode on the Dyna-mite.

Fast Exp. 1:2 expansion with a fast attack (time it takes for the gate to open). Use for most percussive sounds, such as drums, to achieve a natural sounding gate. This corresponds to the "pk" mode on the Dyna-mite.

Slow Exp. 1:2 expansion with a slow attack (time it takes for the gate to open). Use for sounds with a soft onset, such as strings, pads, etc. This corresponds to the "avg" mode on the Dyna-mite.

HF Boost The HF Boost increases the high frequencies that the gate's internal detector sees, which makes it react easier to high frequency sounds. If you need to gate a kick drum, it's a good bet to leave it off. If you need to gate a snare with high frequency content you can turn it on to achieve better separation between kick and snare.

This corresponds to the "ds/fm" mode on the original Dyna-mite.

The HF Boost does not affect the audio path we listen to, just the audio that the detector uses to decide if the gate should open or not.

Ext. S/C Engage the external sidechain (if available).

Gain Reduction Meter Shows the amount of gain reduction.

Output Meter LED meter showing the output volume.

Clip Meter The Clip Meter will turn on if the Dyna-mite is clipping the output signal. The output clipping has been carefully modeled from the original unit and can be used as a distortion effect.

In use

Standard toms gating

Set the switches to "Fast Exp.". Adjust Threshold to taste. (Short tip, but it really is that easy.)

Dyna-mite Overdrive

Set Threshold to minimum. Increase Output Volume until the clip LED turns on. Compensate for the loud output signal by turning down the fader in your DAW.

Sub-Synth Sidechaining

This is a neat trick from the past, that use the sidechain to key a synth so that the synth sounds whenever the sidechain has signal. You probably have plug-ins that do all this, but it is cool to learn how it's done.

Setup the Dyna-mite Gate on a synth track that plays an extended low note. Send the kick drum to the sidechain of the Dyna-mite Gate, so that the gate opens whenever the kick drum hits.

The synth will only sound when the kick drum sounds, and you can use the synth to play a low note that harmonize with the kick drum. Adjust the release time to set the sustain of the note.

Tighten Back-up Harmonies

Okay, another side-chaining trick, but keying is the new black. Setup the Dyna-mite Gate on the backing vocal bus. Sidechain it with the lead vocals. Let the switches be "Slow Exp." with a fairly fast Release time. Adjust Threshold so that the gate opens whenever the lead vocal sings.

Credits

Niklas Odelholm - DSP and UI programming, graphic design, user manual. **Oscar Öberg, Arvid Rosén** - modeling. **Ulf Ekelöf** - 3D rendering.

40 Vintage Amp Room

Introduction

STRAIGHTFORWARD AND SIMPLE, Vintage Amp Room emulates three great guitar amps in a complete studio setup with speaker cabinets and fully flexible microphone positioning. It's easy to use for live performances and recordings, as well as for reamping and lastminute tweaking.

Amp Room has none of the added gadgets or weird-sounding presets sometimes found in simulation software. Focusing on quality and accuracy, it simulates three amps, with sounds that are timeless and authentically raw. The goal is not to give you a preproduced, ultimate guitar sound, but to provide you with the ultimate tool to create your own. Because after all, only you know exactly what sound you want.

Product Philosophy

EVERYTHING ABOUT THIS PRODUCT is authentic. The raw and dynamic sound is an absolute replica of what you would get if you had these amps set up in a real studio. What you see on the screen are photorealistic 3D renderings of the complete setups, and you get to move an actual mic around when deciding which position is best, from near field offaxis to far field and back, continuously, without preset positions.

THE IDEA IS THAT using Amp Room should be like working directly with these amplifiers set up in a real studio, with a microphone in front of the cabinet, connected to your DAW. And we kept it simple. Amp Room has no knobs with dubious or unintuitive functionality, and no added gadgets or ridiculously fake-sounding effects.



The sound of the amps has not been tampered with, because we don't give you a pre produced, ultimate guitar sound. We simply provide you with the tools: three great, classic amplifiers, in digital form. Then it's up to you to add compressors, EQs, reverb, or any other effects, to get the sound you want.

In short, you need the same skills to master this software as you need when dealing with a real studio setup. You can get back to doing what you do best, because music production is about using your ears, not your computer skills.

Supernormalize

The “supernormalize” feature (from the beginning the internal name of a slightly magic component of our model building toolbox), makes sure that the output signal from Amp Room always has a reasonable volume. From a user perspective, this means that the Amp Room never clips the signal (unwanted digital distortion). Also, the output volume is normalized, so even with low settings on the Volume knob in Amp Room, the output volume will be within a reasonable level and ready for digital recordings. Compare this to the real amp, which easily can have a dynamic range of 100 dB.

From a technical perspective, this means that the signal path within Amp Room always has the highest dynamic range possible, and you won't lose any bits on the way.

User Interface

Simplicity has been the goal when we developed this product; the amps and mics should work just as they do in the real studio. Unfortunately, moving a mouse cursor on a 17” LCD screen isn't exactly like grabbing the real mic stand in a cozy studio. Nevertheless, we have tried to imitate that feeling

as far as it is possible, and if you have experience of working in real studios, you will notice that Amp Room handles and sounds the way you expect it to do.

Amp Panel (Top Area)

In the Amp Panel you can alter the settings of the amp by clicking the knob and dragging the mouse up and down. Switches will be switched either by clicking on them, or by click-and-dragging the mouse up and down.

Amp Panel



Room View

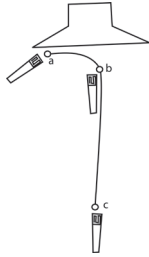


Room View (Bottom Area)

In the Room View you can do two things, select one of the other amps or change the mic position.

Mic Position

When the mouse is located over the mic stand, the mouse pointer changes to an “up-down” arrow. Click and drag the mouse up or down to change the position of the mic stand. The microphone moves along a predetermined path, so you only need to move the mouse up or down, just as if you were changing a knob.



Amp Selection

You can change amp by clicking on the background and dragging the mouse to the left or right. The mouse pointer becomes a “left-right” arrow when you are pointing at the back ground to indicate that it is possible to change amplifier.



White Amp

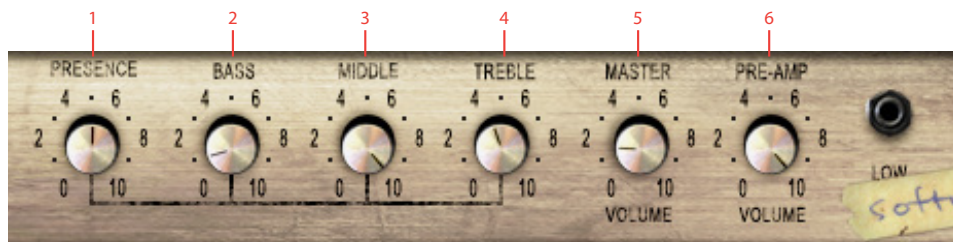
White is based on an allround great amp, a classic that set the industry standard for classic, distorted rock'n'roll sounds. The genius is its simplicity. It's stripped down to the bare essentials – all tubes, only one channel, and no reverb or any other built-in effects.

Soundwise, it goes from mad screaming preamp shred, to warm, speakersabouttobreak power amp distortion. Perfect when you need a characteristically distorted edge and a powerful roar.

First Use

Set all parameters in the middle (12 o'clock). Turn down the **Master Volume** a little bit (to 10 o'clock) and play your guitar while changing the **Pre Amp Volume**. The sound goes from almost HiFi clean on low **Pre Amp Volume** settings, to a crunchy distortion (**Pre Amp Volume** in the middle), and finally to a high-gain setting with **Pre Amp Volume** on max. If you want an even meaner distortion, turn down the **Middle** and turn up the **Master Volume**.

The sound will change quite dramatically when you start moving the mic away from the cabinet, since certain frequencies will cancel each other out due to interference between the four speaker drivers.



1. Presence
2. Bass
3. Middle
4. Treble
5. Master volume
6. Pre-amp volume

Knobs

The functionality of the knobs on the front panel of the plugin correspond exactly to the front panel of the real amplifier.

Presence The presence is used to control the feedback loop and thus the amount of treble in the power amp. Turn it up to get a high frequency boost.

Bass, Middle, and

Treble These knobs are the tone control of the amplifier. This circuit is located after the preamp distortion and is used to shape the sound of the preamp distortion.

Master Volume This knob controls the amount of power amplifier distortion. The power amp distortion is usually a little bit dirtier than the preamp distortion and gives a crunchier sound that sounds amazing for chords. Set the **Pre Amp Volume** to a low setting and turn up the **Master Volume** to the max to get that typical power amp distortion.

Pre Amp Volume This knob controls the amount of preamplifier distortion, but since a high output from the preamplifier also makes the power stage distort, this also affects the power amplifier distortion. The preamp distortion is typically much “tighter” than the crunchy power amp distortion. To get a tight preamp distortion, you’ll have to turn down the **Master Volume** while maxing the **Pre Amp Volume**.

Brown Amp

Brown is based on one of the most versatile classic amplifiers in music history. It’s been used in all kinds of genres, and with all kind of instruments, thanks to a sound that is clean, crisp and clear, but also warm, thick and punchy.

Brown can produce anything from the clearest countrystyle sound for steelguitars to a dirty, bluesy guitar sound. It works for both funky rhythms and mellow electric piano chords. It’s got a notoriously warm sound with a massive bass and a shimmering treble. Versatility embodied, it can do more or less whatever you want it to do. Except high gain heavy metal. And baking pizzas.

First Use

Set all knobs in the middle (at 12 o’clock). Set the microphone in a far field position (click the mic and drag the mouse downwards). Turn on the **Vibrato** and **Bright** switches. You should hear a bluesy sound with lots of room ambiance and a nice transition between distortion and a clean sound. Change the **Volume** to get more or less distortion and move the microphone closer to the cabinet to get a “tighter” sound with more bass frequencies.



- | | |
|-----------|----------------------|
| 1. Bright | 5. Bass |
| 2. Volume | 6. Vibrato On/Off |
| 3. Treble | 7. Vibrato speed |
| 4. Middle | 8. Vibrato intensity |

Knobs

The functionality of the knobs on the front panel of the plugin corresponds to the front panel of the real amplifier, except the spring reverb, which hasn't been included in this plug in. We have also added a “Vibrato” switch, which on the real amplifier is a foot switch. There is also a difference when all the tone controls are set at 0, see “Treble, Middle, and Bass” below.

Volume This amplifier doesn't have a Master Volume control, so the **Volume** knob controls both the preamplifier and power amplifier distortion.

Bright Turn on this switch to get a high frequency boost. Works only at low to moderate **Volume** settings.

Treble, Middle, and

Bass These knobs are the tone control of the amplifier, but they also control the amount of distortion. If you, for example, have too much distortion in the bass frequencies, try turning down the **Bass** knob. (Electrically speaking, the tone controls are located before the preamp distortion.)

The tone controls behave exactly like the real thing, except when all the controls are set fully counter clockwise. On the real amp, the amplifier would have become silent, as if you turned down the volume. A lot of reasons made us exclude this “feature,” so now when you turn the **Treble**, **Middle** and **Bass** knobs fully counter clockwise, that actually corresponds to the knobs being set to a little bit over nothing on the real amp.

Vibrato, Speed and

Intensity Turn on the vibrato (which is in fact a tremolo) by turning on the **Vibrato** switch. Use **Speed** to control the speed of the tremolo, and **Intensity** to control the amount of the tremolo effect.

Green Amp

Green is packed with character and produces a pleasant guitar sound with warm yet intense poweramp distortion. Its edginess makes it shine through in mixes, without drowning out other instruments.

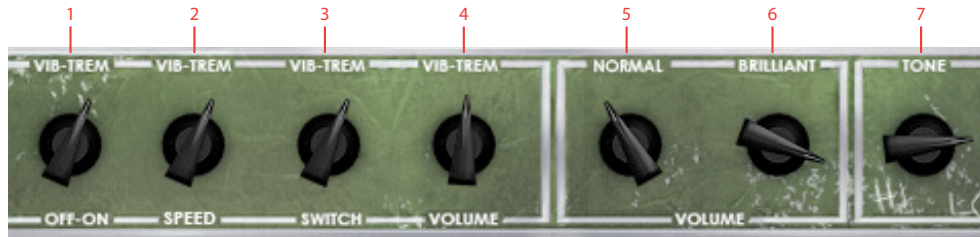
Green is based on a famous britpop amplifier from the middle of the 60's. It has three different channels ("Vib Trem", "Normal" and "Brilliant") with two inputs each, and normally a guitar player will connect the guitar to the "VibTrem" input and patch the signal to the other two channels. We have simulated the amplifier exactly this way; the guitar connected to the "VibTrem" input and patched to the other two channels.

To complicate things further, this amp doesn't have a normal tone stack (EQ) but a single tone control which attenuates high frequencies when turned clockwise (different to what one would expect of a tone control). The previous owner of the original amp was so kind as

to carve out "HI" and "LO" in the metal panel so that he would remember this. We have of course included this feature in Amp Room.

First Use

Turn on the vibrato/tremolo effect (**Vib-Trem Off-On**) and select the tremolo effect (**Vib-Trem Switch** pointing at "TREM"). Set the **Vib-Trem Volume** knob in the middle position and all other knobs in the minimum position. Now adjust the **Brilliant Volume** to get a good balance between the tremolo effect and a clean sound.



1. Vib/Trem Off/On
2. Vib/Trem speed
3. Vib/Trem switch
4. Vib/Trem Ch. volume
5. Normal Ch. volume
6. Brilliant Ch. volume
7. Tone

Knobs

The functionality of the knobs on the front panel of the plugin corresponds exactly to the front panel of the real amplifier, except that we have added the foot switch as a “Vib-Trem Off-On” switch.

Vib-Trem Off-On Turn on the vibrato or tremolo effect by moving this switch clockwise.

Vib-Trem Speed Three way switch that sets the speed of the vibrato or tremolo effect.

Vib-Trem Switch Select between a vibrato and a tremolo effect. Unlike the makers of the Brown amp, the guys who built this amp actually knew the difference between a tremolo effect (amplitude modulation of a signal) and a vibrato effect (frequency modulation of a signal). Although this vibrato effect doesn’t sound anything like a real-world vibrato (opera singerstyle vibrato), it’s a lovely effect that gives the sound that special touch.

Vib-Trem Volume Volume control of the Vib-Trem channel. This channel is basically the only one of the three channels that has any preamp distortion worth talking of.

Normal Volume Volume control of the Normal channel. Except for the fuzz-like poweramp distortion on higher volume settings, this channel is very clean and doesn’t change the guitar sound so much.

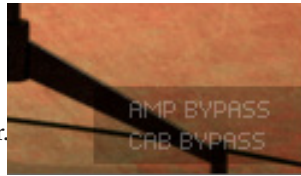
Brilliant Volume Volume control of the Brilliant channel, which has a much brighter sound than the Normal channel.

Tone The Tone knob cuts high frequencies when it’s turned clock wise.

Use the **Vib-Trem Volume** to get the distortion you want, and then use the **Normal** and **Brilliant Volume** as a bass and treble control. If you want a more fuzz-like distortion, use only the Normal and Brilliant channel.

Bypassing Amps or Cabs

You can choose to bypass the amp or the cabinet by selecting **AMP BYPASS** or **CAB BYPASS** from the small box in the lower right corner.



This is very useful if you want to use Vintage Amp Room with an external speaker (or speaker plugin) or if you just want to add a cabinet simulation to a track, such as a recorded line out from your amplifier.

By bypassing the cabinets in Vintage Amp Room, you can use the cabinets from the other Amp Room plugins. Just insert, for example, Bass Amp Room (with bypassed amplifier) on the track after Vintage Amp Room (with a bypassed cabinet).

Buying Recommendations

If you like the sound from Amp Room and would like to get that sound using the real deal (let's face it, a real vintage amp is always sexier than a plugin), here are some buying recommendations.

White

If you like the sound of the White amp, we recommend you to buy a Marshall JCM800 2203 with a 1960A 4x12" cabinet. Nothing beats the roar from a 100W valve amp.

Brown

Likewise, if Brown is your favorite, we strongly recommend a 1966 Fender Twin Reverb combo with a 2x12" cabinet.

Green

Nothing beats the real stuff, and that quirky vibrato calls for a Vox AC30/6 Treble from the middle of the 60's, perhaps suited with a pair of new Celestion Blue drivers.

Room

And finally, if you like the sound of the room in which the cabinets we're measured, you better call Soulmine Studios in Linköping and book some studio hours. Just tell them to set things up just as they did for the guys from Softube, and you'll be fine from there...

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Credits

Oscar Öberg – concept, preamp modeling, sound and DSP-programming. **Niklas Odelholm** – cabinet and microphone modeling and sound design. **Arvid Rosén** – power amp modeling and sound programming. **Torsten Gatu** – interface and DSP programming. **Per Connman** – amp selection and modeling. **Anders** – graphics programming. **Ulf Ekelöf** – graphics programming and 3D rendering. **Micko** – 3D rendering. **Papa Bear** – collages and graphic profiling.

41 Other Platforms

Reason Rack Extensions

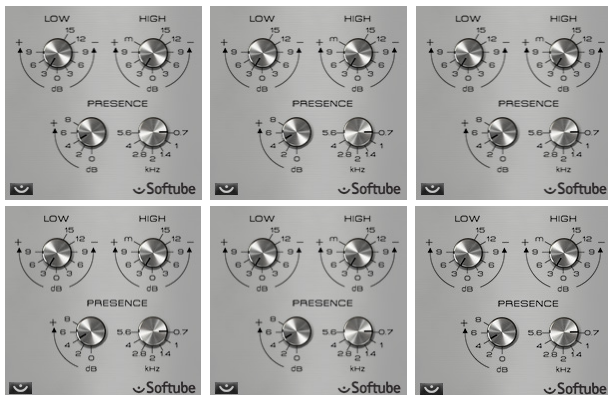
A lot of Softube products are also available as *Rack Extensions* (RE) for Propellerhead's host Reason. The sound quality and features are the same as in other plug-in formats. Rack Extensions are only sold through the Propellerhead web store.

The Rack Extensions licenses are completely separate from our VST/AU/RTAS/AAX licenses, are sold separately, and cannot be transferred.

For more information about Rack Extensions, please visit www.propellerheads.se.

Products Currently Available

Tube-Tech Classic Channel, Trident A-Range, Saturation Knob, FET Compressor, TSAR-1 Reverb, TSAR-1R Reverb, Valley People Dyna-mite, Spring Reverb, and Tube Delay are currently available in the Rack Extensions format.



Cakewalk Sonar ProChannel

ProChannel is an intramix interface for VST plug-ins, with easy accessible knobs directly in the mixer. ProChannel compatibility is included in the normal VST/AU license, and no separate installer is needed.

The ProChannel versions of the FET Compressor and Focusing Equalizer do not include all parameters, but the full VST versions are installed at the same time as

the ProChannel modules, and you can easily access them from the inserts menu.

Products Currently Available

The products included in the Mix Bundle: TSAR-1R Reverb, Passive Equalizer, Focusing Equalizer, Active Equalizer and FET Compressor are available with a ProChannel compatibility. Saturation Knob is included in Sonar X1 Producer Expanded.

UAD-2 Powered Plug-Ins

All Amp Room plug-ins (Vintage, Metal and Bass Amp Room), Tonelux Tilt and Tilt Live, Valley People Dynamite, Summit Audio TLA-100A, Tube-Tech CL 1B, PE 1C and ME 1B are currently available for the UAD-2 Powered Plug-Ins platform and Apollo High-Resolution Audio Interface. Marshall Legends, including Plexi Super Lead 1959, Silver Jubilee 2555, Bluesbreaker 1962 and JMP 2203 are available exclusively for UAD.

TC Electronic PowerCore

Vintage Amp Room has previously been available for PowerCore, but is no longer sold or supported.

42

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Nigel Stewart, Milan Ikits, Marcelo E. Magallon and Lev Povalahev

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Brian Paul

Mesa 3-D graphics library

Version: 7.0

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Baptiste Lepilleur

JsonCpp

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easywsclient

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This font was created by "30100flo".

This font has a homepage where this archive and other versions may be found:

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Christian Robertson

Roboto typeface

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