LISA

Plugin Manual

Developed by Brainworx in partnership with TOMO Audiolabs and distributed by Plugin Alliance

PLUGIN ALLIANCE TOMO AUDIOLABS LISA BRAINWORX

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About the LISA Hardware and Plugin

Assembled from the highest quality discrete components, LISA's parts are handpicked specifically for each of their applications. Only ELMA switches, Mundorf capacitors and Lundahl transfomer are used to build each LISA. Additionally, we do not embed potentiometers into our audio circuitry. We here at TOMO Audiolabs do this so that each new LISA owner is guaranteed the highest sonic quality available today!

Numerous listening sessions with our development team and various audio professionals certified the hope our extensive research and measurements instilled in us... LISA sounds extraordinary!! Proving that LISA is not only a masterpiece by technical measures.

Although LISA was designed with the needs of a working audio professional in mind, its sonic range can help you realize your own audiophile visions as well. If we had intended for this product to be loved solely by working studio pros we might have named it the TAL-100 but with all music lovers in mind...we named it LISA.

As a new owner of this marvelous product, we invite you to discover all of the creative possibilities LISA offers. Be sure to experiment with this wonderful tool and you will unlock sonic possibilities you never knew existed!!

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LISA is a dynamic equalizer with a unique design philosophy, making it different from any other equalizer you may be familiar with already.

The signal runs through top-notch input amplifier stages equipped with hand-wound high quality transformers. The signal is then split into six parallel channels that feed the six independent equalizer bands. After each signal is processed independently, all six channels are then summed up and amplified. The output stage is also transformer coupled and gives the user the ability to mix the processed signal with the unprocessed or dry signal.

The parallel topology allows parallel processing (two bands at nearly the same frequency) without the issues of interacting frequency curves. This keeps the processed signal very transparent, consistent and controllable. The results stay predictable, helping you to work very quickly and with an experienced touch.

Traditionally designed dynamic equalizers utilize a dynamic processor that is arranged before or after the filter circuit. Part of what makes the design of LISA's dynamic processor so unique is that it is built into the filter circuit. This allows the user to listen to only the processed signal without any of the dry signal.

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Plugin only additions including the new “TX Drive”

As always, the engineers at Brainworx have added new digital-only features that would be impossible in the analog domain. In addition to regular favorites like the Mono Maker, Stereo Width, Auto Listen, Headroom and the ground- breaking TMT (TMT, US Patent No. 10,725,727), offering an unrivaled analog experience in the box. The LISA adds a new parameter for the first time: **TX Drive**. This new parameter allows you to dial in more or less transformer saturation and character, separate from the other analog characteristics of the device. And the newly developed transformer model accurately reproduces both saturation and hysteresis distortions of the specific transformer, making it sound and behave exactly like its analogue counterpart. With all the additional features, this Plugin offers everything you need in a modern digital environment.

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Features

- Precise emulation of the unique TOMO Audiolabs LISA
- Independent opto-compression based dynamic section for each of its 6 bands per channel, switchable between compression and expansion
- Includes two parametric mid bands, parametric low and high bands with a shelving mode, and two dedicated boost bands - plus two highpass filter
- Parallel circuits allows for greater changes in frequency with more transparency
- Six EQ bands with a broad frequency overlap provide optimum processing flexibility
- Scalable User Interface
- Artist Presets to get you started on a variety of sources
- New TX Drive feature acts like a „Headroom“ control for the transformer model only. A new Brainworx innovation!
- Plugin only features include Brainworx' Mono Maker, Stereo Width, Headroom, Auto Listen and patented TMT technology
- Includes a separate mono version for individual tracks such bass, kick, snare and vocals (see left side on this page)

Input Section

❶ Input Gain

A discrete stepped input level adjustment, positioned after the input transformer and before the HPF.

❷ HPF Frequency

A variable Q, 3rd order HPF. At 20Hz, the filter exhibits a “near-Butterworth” perfect 3rd order response with 18dB/oct steepness. The filter’s slope gets progressively milder when moving up the frequency and at the upper setting of 180Hz, it starts at 6dB/oct, steepens to around 12dB/oct in the mid-100’s and finally goes to 18dB/oct at around 50Hz. The result is a more “musical feeling” at higher frequencies with the same effectiveness at the lowest frequencies.

❸ HPF ON

Turns the HPF on or off completely.

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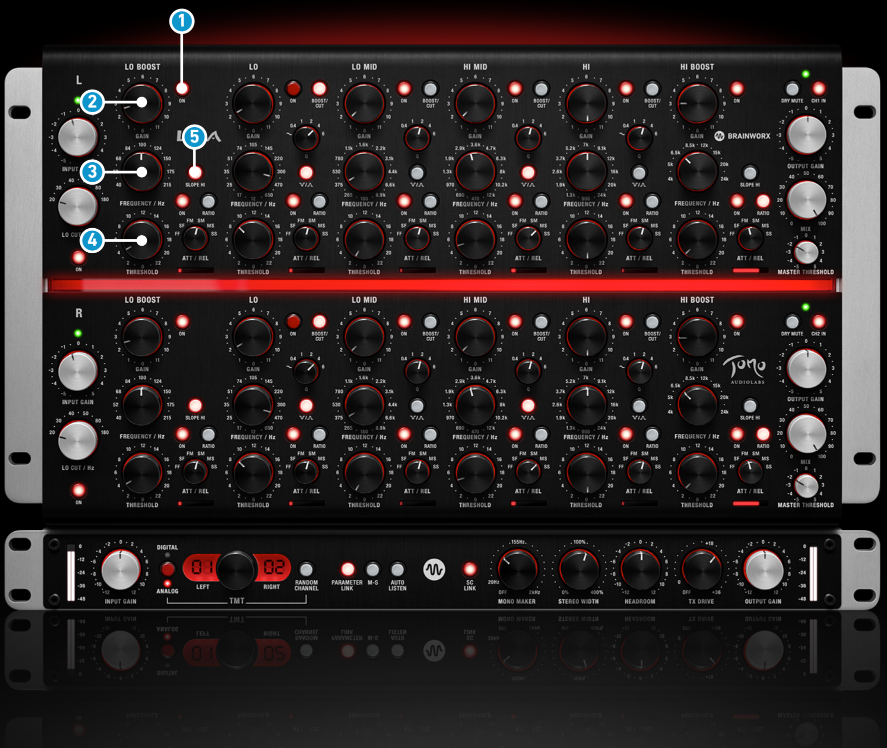


Band 1 & 6 - LO BOOST / HI BOOST

1 Band ON – BAND 1 - 6

Enables the filter band and routs its output to the summing amplifier.

2 Gain – LO BOOST / HI BOOST

Adjust the amount of level boost in the LO BOOST and HI BOOST band.

3 Frequency – LO BOOST / HI BOOST

Sets the centre frequency of the LO BOOST and HI BOOST filter.

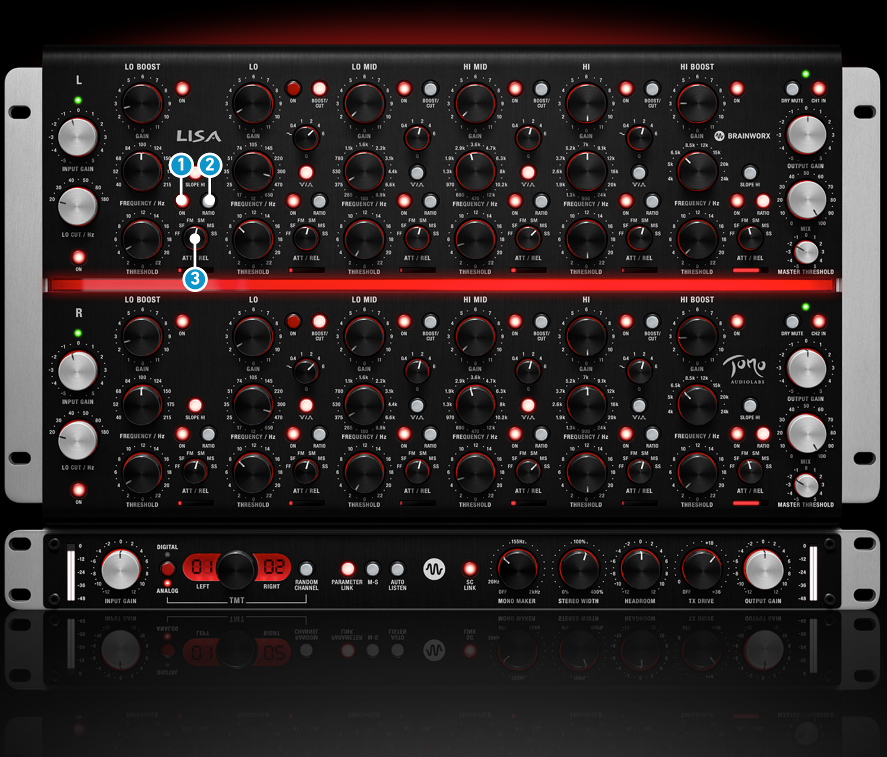
4 Threshold – BAND 1 – 6

Defines the level at which dynamic processing is applied to an input signal. The range is not specified in any meaningful units, but simply as steps from 0 to 23, where a higher number means more compression or expansion.

5 Slope – LO BOOST / HI BOOST

The Slope switch introduces another pole into the BOOST bands' response. In normal mode, those are 2nd order shelf filters, but with the Slope enabled, they become a 3rd order shelf filters with a little bit of resonance after the transition part of the response. This can be usefull for example to augment the kick or snare drum's fundamental frequency while simultaneously also cutting a little bit of the mids out right above it. Pultec trick, with far more precise control.

Only available on LO/HI BOOST bands.



❶ Dynamics ON – BAND 1 – 6

Engages or disengages the dynamics section completely.

❷ Ratio – BAND 1 – 6

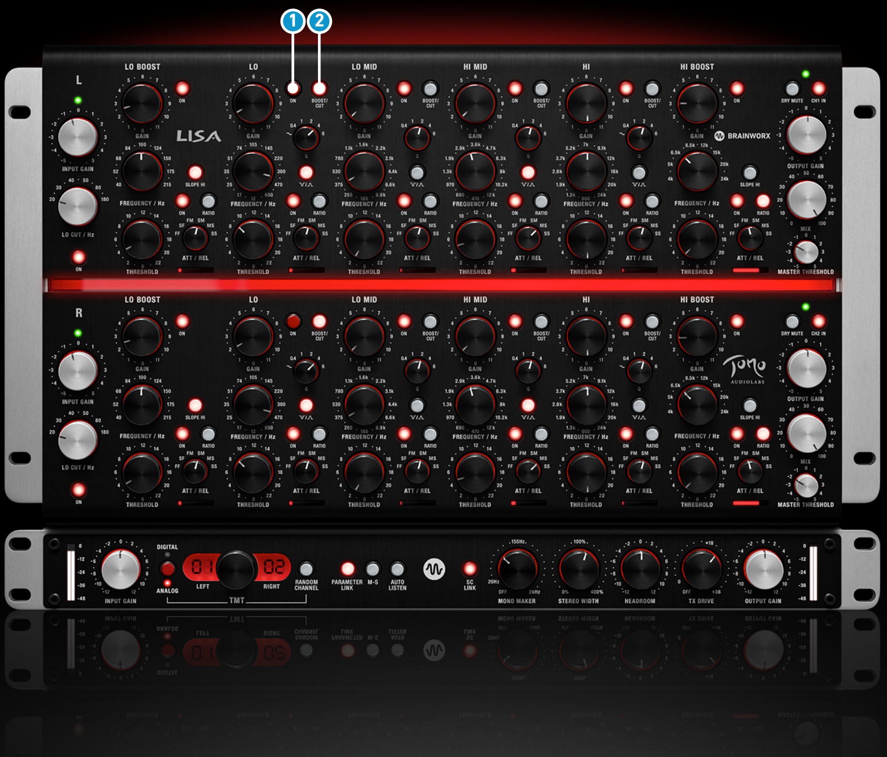
The ratio switch could also be labeled “compressor/limiter”. It changes the strength with which the dynamics section influences the filter band for every input signal volume increment over the threshold value. With Ratio set to “off”, the dynamics influence is gentle and hard to overdo whilst with the Ratio turned on, the dynamics sidechain becomes very sensitive and will trigger heavy filter change when the signal passes the set threshold level. The ratio parameter works slightly different to the ratio setting on a standard compressor you might be familiar with. Try different settings to get to know the differences.

❸ Att/Rel – BAND 1 – 6

Controls the timing settings for the dynamic processing. This selector switch lets you choose between 6 different combinations of preset attack and release times. As the actual timing contants are “dual” in nature, combining the actual timing circuit and the opto-cell’s own (slightly program dependent) timing behaviour, they have no accurate millisecond designations. From my experience, the attack sounds something like 0.1/1/3ms and the release something like 80/200/800ms for F/M/S, respectively.

F – fast **M – medium** **S – slow**

The first letter stands for the attack value and the second letter for the release value.



Band 2 to 5 - LO, LO MID, HI MID, HI

3 Band ON – BAND 2 to 5

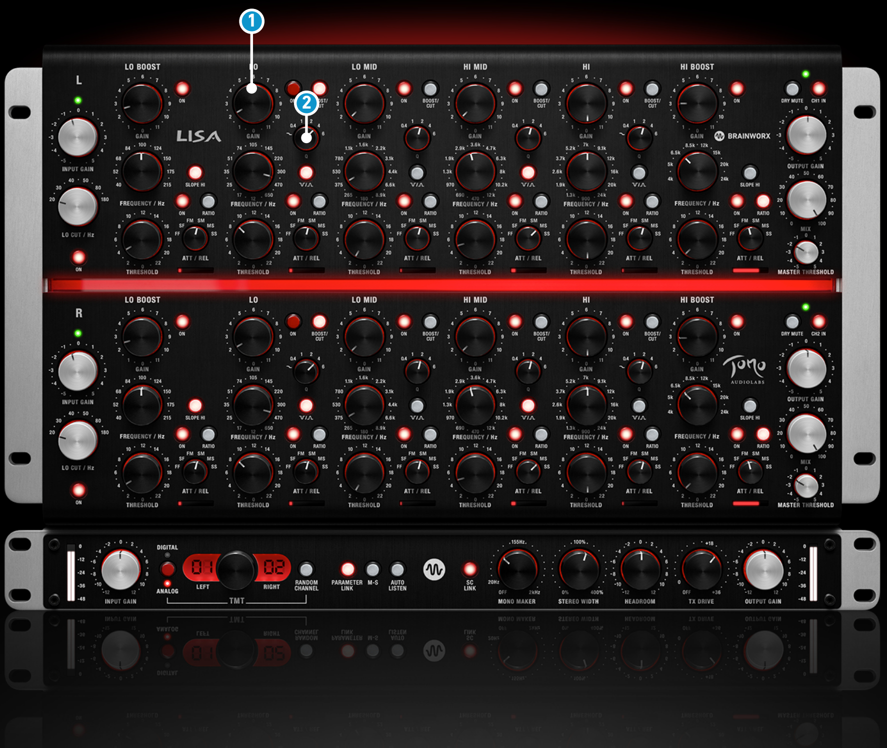
Enables the band and routs its output to the summing amplifier.

2 Boost/Cut - BAND 2 to 5

Switches the band's mode between boost and cut. It achieves that by changing the input to the filter section from neutral polarity input signal to the inverted polarity summing amplifier's output signal.

Pressed (LED ON): Boost

un-pressed (LED OFF): Cut



❶ Gain – BAND 2 to 5

Adjusts the level of the boost or cut. It doesn't have a dB valued scale, as the actual maximum level depends on the type of the filter (boost or normal) and its shape (boost slope, shelf or bell).

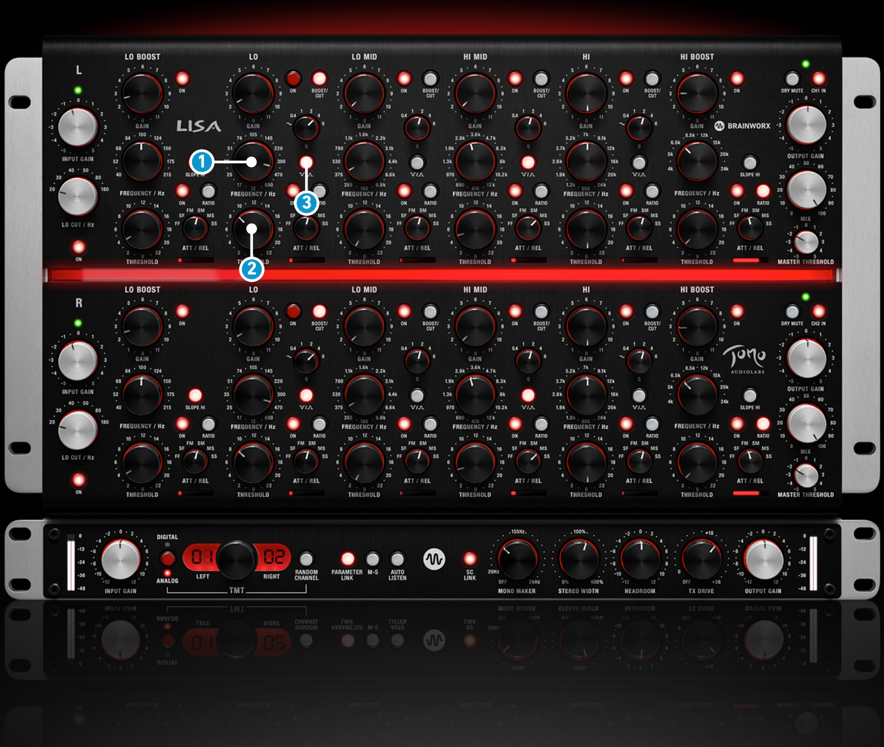
Note: Settings between 0 and 6 are stepped in 0.5 dB steps. Above 6 dB the steps are getting broader. You can find a stepping list in the table shown below, applicable with 100% dry signal only:

0,5 - 6	0.5 dB per step	12 steps
6,5 - 9	0.6 dB per step	6 steps
9,5 - 10	1 dB per step	2 steps
10,5	1.2 dB per step	1 steps
11 - 11,5	1.5 dB per step	2 steps

❷ Q – BAND 2 - 5

Decides the width of the bell filter. Higher Q means a sharper filter and a lower Q means a broader filter. The LO and HI bands also offer a sixth setting – shelf. This opens up the filter all the way up or down the spectrum (for LO and HI, respectively), to achieve a shelf-like response.

Shelf mode is only available on the LO and HI bands.



❶ Frequency – BAND 2 - 5

Sets the centre frequency of the bell filter or the “second knee frequency” in shelf or boost slope mode. To say it differently, the second knee is the one that is already boosted or cut, where the response levels out again.

❷ Threshold – BAND 2 – 5

Use this control to adjust the dynamic action of each individual band. The range is not specified in any meaningful units, but simply as steps from 0 to 23, where a higher number means more compression or expansion.

❸ Compressor/Expander BAND 2- 5

Working with a filter that can be set to either boost or cut, the compression and expansion can’t be simply defined as “down and up” in overall level. The design paradigm in this particular box is such that for any filter setting, be it boost or cut, compression means movement towards the unity-gain/flat response line, and expansion means movement away from it. In other words, compression is “less of it” and expansion means “more of it”, where “it” is either boost or cut.

Because of the shape-shifting nature (see notes), the dynamics section is probably the most unique feature of this product. It can do both cold-blooded problem solving and dancing with the music, and as far as I can tell, there’s no other device on the planet that does that the way Lisa does it. And contrary to initial instinct, the dynamically moving Q value doesn’t sound like “wobble”, but instead produces a very musical result, unobtainable by any other hardware or software device.

The resulting shape of the dynamically modified filter is also non-trivial. Instead of keeping the shape of the filter and changing the level of boost or cut (like moving the Gain knob), the dynamics in Lisa work so that they change the gain and shape/Q of the filter, which results in level change either at the peak (expansion) or around the peak (compression).

Compression applied to the filter will initially keep the peak gain and sharpen the Q, lowering the filter's gain around the set frequency. As the gain reduction increases, the peak will also start to dip in level, but will completely disappear only at extreme gain reduction. Expansion action will, conversely, increase the gain of the filter only at the peak, leaving the "skirt" of the filter intact. It's like augmenting what's important in boost mode and attenuating only the resonant frequency the filter is tuned to in cut mode, without pulling down the surroundings as well like a regular dynamic EQ.

Note: The dynamic section is integrated into the filter circuit and thus can only affect the filtered signal. If there is no gain boost or cut in a band there will not be any dynamic processing either. This is unlike a normal multiband compressor where the signal is always affected even when the output gain of the band has not been boost or cut.

Output Section

1 Dry Mute

The architecture of this particular EQ is parallel, achieving the desired curve by adding a band-passed signal to the raw input signal for each band. Dry Mute thus allows the user to mute the dry signal input to the filter section's summing amplifier, and consequently only listen to what's being added to the dry signal - a sum of all the parallel bandpass filters. It's like going into auto-solo mode for all the filters (in boost mode) at once.

Filters switched into cut mode use the summing amplifier as a polarity inverter. Muting the dry signal thus mutes their source signal as well, so their output is silence. This is a „quirk" of the hardware as well, but had to be retained in the digital model as well to make the dynamics levels behave the same way as on the hardware.

2 Channel IN

Turns the whole channel processing on or off. This includes the trans-formers, HPF, filters, both input and output gain, and parallel mix.

3 Output Gain

A discrete stepped output level adjustment, positioned after the filters and before the output transformer.

1 Mix

This control allows the user to mix the raw, dry signal with the processed signal. The dry line is accurately latency-compensated, so the mix never produces comb filtering.

2 Master Threshold

Master Threshold lowers the input level to all the bands' sidechains, This feature allows the user to control the general dynamic action of all the filter bands of one channel, extending/supporting the functionality of the Headroom parameter to balance the dynamics action for the mid and side channels separately in unlinked mode.

The proportions between the thresholds of the individual bands remain untouched. The Master Threshold function can be especially useful when the input gain setting must be changed but all threshold settings are already adjusted.

BRAINWORX Module

1 Master Input Level

A simple level trim right at the input of the signal chain. Works on all channels simultaneously and equally.

TMT Inside

The LISA offers 20 different eq channels, made possible by TMT. TMT is Brainworx's (TMT, US Patent No. 10,725,727) "Tolerance Modeling Technology", originally found in the bx_console line of plugins. It takes the real-world tolerances of audio components found in audio circuits into account, and offers various channels of analog audio which have realistic variances in frequency response, time constants in dynamic sections, etc. The result is digital audio that sounds as analog as possible, whereas even the L/R channels of a stereo instance will react slightly different. For more information please check www.brainworx.audio

2 TMT Mode

Sets the TMT channel mode. The Analog mode more closely resembles a real-world unit, while the Digital mode allows for perfect stereo matching in both frequency response and dynamics timing constants.

3 TMT Channel

This parameter lets you choose from the variety of subtle variations resulting from modeling electrical component tolerances.

In a Stereo instance, two adjacent channel numbers will be displayed. Each channel has its own, different character!

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1 Random channel

Whenever you instantiate a LISA plugin on a channel, it will start with the Default setup, which is Channel 1 in a flat setting. You can now randomize a channel by clicking the Random Channel button. Only the plugin instance you click on will switch to any unused channel number in that session randomly. The plugin will remember which channel numbers are already used in a session and activate an unused channel number, unless you engage more than 20 channels. At that point the plugin obviously would have to use a channel number that has already been used.

2 Parameter Link

When switched on, this feature links the channel parameters of both channels to allow for easier and faster work on stereo material.

When both parameters have different values and link is engaged, both parameter values remain unless one of them is touched and any control offsets between channels are lost.

3 M/S

Enables or disables the stereo-to-mid/side encoding at the start of the signal chain and decoding at the end of it. The labels on the GUI will change to reflect that mode of operation and designate the upper channel as „M“ and the lower channel as „S“.

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❶ Auto Listen

Enables the Auto-listen functionality for all relevant channel parameters (frequency, Q, Threshold, and Att/Rel). If enabled, the user will hear the soloed raw filter band signal whenever they click and hold one of the mentioned parameters. This is very useful to accurately position the filter in the frequency domain and to clearly hear the effect of the dynamics.

The same functionality can also be accessed by holding the (DAW specific) modifier key while tweaking the parameters. This is usually either Alt, Ctrl/Command, or Shift.

❷ SC Link

When this control is turned on, the dynamics' sidechains of each band are linked. This is especially useful in L/R stereo operation because it prevents the shifting of the stereo image in case a prominent element jumps in volume on only one side of the stereo field. The sidechains are linked only „per-band", so the Low band will still not affect the mid bands for example.

This feature only links the sidechains, not the settings of each channel's dynamics section! It is therefor possible to make each channel's dynamics section react completely differently even in SC-Linked mode by setting the timing and threshold controls without Parameter Link engaged!

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❶ Mono Maker

This tool is a critical component to several Brainworx processors, and it is an invaluable tool when mastering or tightening up a mix. Sweepable from 20Hz to 2kHz, this parameter folds the processed sound to mono at and below the frequency set. The most common setting is between 50-200 Hz, which ensures minimal loss of width while improving translation for lossy codecs and vinyl cutting.

It is switched off completely in the fully-CCW position.

❷ Stereo Width

Make your mix wider than it originally was by increasing the Stereo Width without losing the center of your recordings! You will not lose bass drum power or vocals by making your mix wider this way... and it will not sound different played back in mono at all. If you notice your Correlation Meter (e.g. bx_meter) showing less than 90°, dial up the Mono Maker a bit to tighten up the low-end until acceptable levels are shown.

❸ Headroom

Applies gain or attenuation to the filters' outputs before the dynamic sidechains. This is useful to try more or less dynamic filter action throughout the whole device or quickly adjust the dynamic effect of a preset to better fit the material.

Another use case is to control the dynamics action for the whole device when working in the unlinked, MS mode. In this case, you can use the Master Thresholds to account for the inherent level difference between the M and the S channel, then use the Headroom control to adjust the whole unit's dynamic action without loosing the ratio between the master thresholds.

Only affects dynamics, not the harmonic distortion amount.

1 TX Drive

This is the „Headroom" parameter for the transformer model only. The value is the 0dBFS reference level if the actual unit was connected in an AD/DA converter loop. Raising this value results in more harmonic distortion in low/low-mid frequencies, and lowering it results in less harmonic distortion. A small portion of the harmonics stays the same, as they result from the hysteresis phenomena that is also a part of our new transformer model.

The default at +18dBu is the reference level we use for all our hardware models. This means that if both Headroom and TX Drive are left at default, the plugin will react the same way as a hardware unit connected to a converter with that reference level.

2 Master Output Level

A simple level trim right at the output of the signal chain. Works on all channels simultaneously and equally.

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Meters

1 Input level LED

This indicator serves as a guideline for gainstaging the filters/dynamics section, so the circuits operate in sensible ranges.

2 Output level LED

Indicates the filter section's output level and is, as such, a guideline for output transformer level at callibrated setting of TX Drive.

3 Input Meter

Input Meter before all processing per channel.

4 Input Clip Indicator

Installed in the Input Meter:

Turns on if input signal is >0dBFS per channel, stays on if there have been overshoots before, click resets all clipping Indicators.

5 Output Meter

Output Meter after all processing per channel.

6 Output Clip Indicator

Installed in the Output Meter:

Turns on if input signal is >0dBFS per channel, stays on if there have been overshoots before, click resets all clipping Indicators

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1 Dynamics action bar meters

Indicates the dynamics sidechain control voltage level. This translates to a level of either compression or expansion, depending on the settings.

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Top Toolbar

1 UI Scaling

The UI size can be scaled using this menu options from 50% to 150% of its normal size.

Adapt the graphical user interface to the size and resolution of your screen. The plugin will check if the user interface will fit your screen before enlarging it. So if you accidentally chose a zoom factor which is too big for your current settings, the plugin will automatically stay at the maximum possible zoom factor.

2 Undo / Redo

You can undo and redo changes you made to the controls of the LISA plugin at any time. The Undo / Redo will work for as many as 32 steps. This makes experimenting and tweaking knobs easy. If you don´t like what you did... just undo it.

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1 Settings (A/B/C/D)

The Plugin offers four internal settings (A/B/C/D) which will be stored with every preset. So, one preset can contain up to four settings. You may use similar settings with more or less equalization in one setup / preset.

Now, the SETTINGS can be automated in your DAW! This way it´s possible to use different sounds for your lead vocals or drums in various sections of the song. Automate the A/B/C/D settings, and you can still tweak knobs of the individual settings without overriding multiple parameters in your DAW, which would be time-consuming.

2 Copy / Paste

To set up variations of similar sounds you don´t have to dial in the settings several times. Let´s say you like your setting A and want to use the same sound, just with less Stereo Width, as setting B.

Simply press Copy while you are in setting A.
Switch to setting B by pressing ‘B’ in the settings section.
Press PASTE, now setting B is identical to setting A.
Reduce the Stereo Width on the B setting.

Now you can switch between A & B and decide which one sounds best or automate different settings for various sections of your session.

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❶ M/S Monitoring (for Stereo Channels only)

Solo M: Solos the Mid (Sum) signal being processed by the plugin.

Solo S: Solos the Side (Difference) signal processed by the plugin.

❷ LED

Dims the red lighting. Adopted from the original hardware. In the Info Screen window which can be accessed by clicking on the Brainworx and LISA logos, the currently selected value can be applied as system default, so new instances will open up with your favorite setting first.

❸ Icon

The icon closes and opens the bottom panel containing the Brainworx´ plugin only features.

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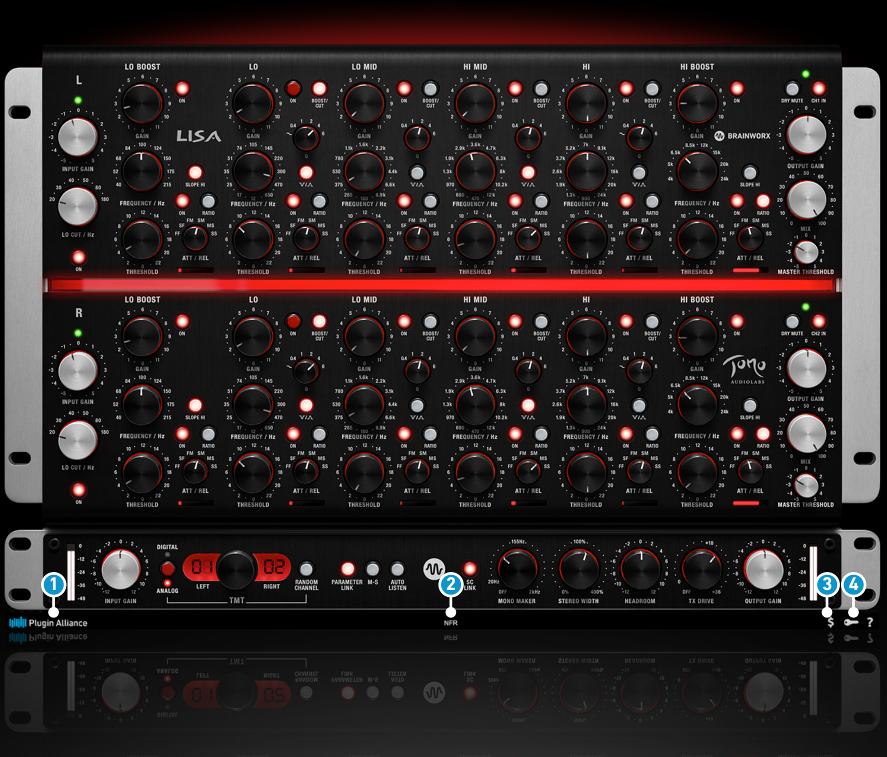

Bottom Toolbar

1 PA Logo

Clicking the Plugin Alliance logo takes you to the Plugin Alliance website via your web browser, that´s if your computer is online.

2 License Type

The toolbar displays information about the type of license you're running: Trial licenses will be displayed along with the number of days until expiration; there is no note for full licenses as these are unlimited.

3 $ (Icon)

If you are using a demo / trial version of our products, you can always click this icon to open a browser that redirects you to the respective product page in the Plugin Alliance store. This is where you can easily purchase a product without having to look it up on our website.

4 Key (Icon)

Clicking on the key icon brings up the activation dialog, allowing you to manually reauthorize a device in the event of a license upgrade or addition. You can also use this feature to activate additional computers or USB ash drives.

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1 ? (Icon)

Clicking the ? icon opens up a context menu that links to the product manual PDF, as well as other helpful links, e.g. to check for product updates online. You must have a PDF reader installed on your computer to be able to read the manual.

System Requirements & FAQ (Links)

For latest System Requirements & Supported Platforms

https://www.plugin-alliance.com/en/systemrequirements.html

Particular details for your product

https://www.plugin-alliance.com/en/products.html

Installation, Activation, Authorisation and FAQ´s

https://www.plugin-alliance.com/en/support.html

Modifier Keys

Tested with Logic Pro X, Protools, Cubase, and Presonus Studio One Mac/Win.

Plugin Format	Jump between Default / Last Setting	Fine Control	Auto Listen
Modifier Keys Assignment			
AU	Option	Shift	Command
AAX	Option (Mac), Alt (Win)	Command (Mac), Ctrl (Win)	Shift
VST + VST3	Command (Mac), Ctrl (Win)	Shift	Option (Mac), Alt (Win)

TIPS and TRICKS

Classic equalizer

In order to acquaint you with LISA's sound, we recommend starting with a classic equalizer setting before moving on to dynamic processing. As stated previously, LISA's parallel design topology allows you to listen to the filter output without any of the dry, unprocessed signal. This is impossible with common serial circuit designs.

Now, let´s listen to LISA and its equalizing possibilities:

- Put some music on the output of your converter Press the **DRY MUTE** button
- Activate band **HI MID**
- Switch the band to **BOOST** mode
- As a starting point, turn the **GAIN** control to its 6 dB value You are now listening to the filter band only
- Try different settings of the **Q** and **FREQUENCY** controls to get a feeling for how they affect the sound
- Try out the other filter bands
- Note: If you switch a band to **CUT** mode you will hear – nothing. As long as there is no dry signal of course there can not be any reduction.
- Next turn all **GAIN** controls back to 0 Deactivate **DRY MUTE**
- Now you can hear how the filter interacts with the added sum signal. Repeat the above procedure with some added dry signal

Simple compression

Now that you have an idea of LISA's sonic characteristics and equalizing possibilities we can proceed with adding the dynamic processing circuit. However, one must first understand three major concepts about LISA's dynamics processor. It will help you find the best settings fast and reliably.

Note: The dynamic section is integrated into the filter circuit and thus can only affect the filtered signal. If there is no gain boost or cut in a band there will not be any dynamic processing either.

This is unlike a normal multiband compressor where the signal is always affected even when the output gain of the band has not been boost or cut.

Note: Compression is always reducing the amount of boost or cut. This means: **BOOST** mode: Louder signals will get less frequency boost than quieter signals. **CUT** mode: Louder signals will get less frequency cut than quieter signals.

Note: The dynamics detection circuit is located before the **GAIN** control. Thus the gain reduction LED indicator can show gain change while the GAIN control is closed.

Put some music containing a very dynamic bass signal.

- Press the **DRY MUTE** button
- Activate band **HI MID**
- Switch the band to **BOOST** mode
- As a starting point, turn the **GAIN** control to its 6 dB value
- Use the **Q** and **FREQUENCY** controls to isolate the main frequencies of the bass instrument
- Listen to how the filter reacts in its static setting
- Toggle the dynamics function to **COMP** mode
- Choose the **SM** timing setting (slow attack and medium release)
- Activate the band’s dynamic processing by pressing its ON button
- Lower the **THRESHOLD** step-by-step and listen to how the processing changes and reduces the influence of the filter
- Toggle the **RATIO** switch to its higher setting (pressed) and listen to the changes

A higher ratio setting will let you hear the influence of the compressor on the timing of the input signal more clearly. A slower attack increases the likeliness of an altered signal transient – but only in the selected frequency range.

- Set the compressor values so that you can clearly hear it working
- Return the **GAIN** controls of the filter bands to 0
- Deactivate the **DRY MUTE** button
- Add the dynamic filters to the audible dry sum signal by re-engaging the filter **GAIN** controls

Simple expansion

The dynamic processor of each filter band can be toggled between compression and expansion. This uses UPWARD-EXPANSION and does not work like a gate, which would utilize downward-expansion.

The result is increased filter activity as the signal reaches the threshold.

Note: Expansion is always increasing the amount of boost or cut. This means:

BOOST mode:

Louder signals will get more frequency boost than quieter signals.

CUT mode:

Louder signals will get more frequency cut than quieter signals.

Attention: The dynamics detection circuit is located before the GAIN control. Thus the gain reduction LED indicator can show gain change while the GAIN control is closed.

Put some music containing very dynamic level peaks on. We recommend music with a very dynamic drum set.
Let´s assume our example contains the signal of a very dominant snare drum...

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- Press the **DRY MUTE** button
- Activate band **LO MID**
- Switch the band to **BOOST** mode
- As a starting point, turn the **GAIN** control to its 6 dB value
- Use the **Q** and **FREQUENCY** controls to isolate the fundamentals of the snare drum
- Listen to how the filter reacts on this frequency band and the instrument
- Toggle the dynamics function to **EXP** mode
- Choose the **FM** timing setting (fast attack and medium release)
- Activate the band´s dynamic processor by pressing its **ON** button
- Lower the **THRESHOLD** step-by-step and listen to how the processing increases the influence of the filter
- Toggle the **RATIO** switch to its higher setting (pressed) and listen to the changes

A higher ratio setting will let you hear the influence of the expander on the timing of the input signal. A faster attack increases the likeliness of an altered signal transient – but only in the selected frequency range.

- Set the expander to values where you can clearly hear it working
- Return the **GAIN** controls of the filter bands to 0
- Deactivate the **DRY MUTE** button
- Now add the dynamic filters to the audible dry sum signal by re-engaging the filter **GAIN** controls. Listen to how the frequency’s peaks stand out from the consistent signal
- Switch the filter to **CUT** mode
- Notice how the snare remains unaffected during quiet parts and how it gets reduced during louder sections

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Multiband compressor-simulation

Although we must admit this is a more daring setting, we'd like you to see how well LISA can emulate a ‚real' multiband (two bands for getting started) compressor. Try this out first and then try it out with more bands. Have fun with it!

- Put some music on Press the **DRY MUTE** button
- Activate the **LO** and **HI** band
- Switch the band to **BOOST** mode
- As a starting point, turn the **GAIN** control to its 6 dB value Switch both **Q** controls to their **SHELVING** setting
- Set the **LO** band **FREQUENCY** control to 650 Hz
- Set the **HI** band **FREQUENCY** control to 3.7 kHz
- The output of both filters will sum to a relatively linear frequency response Switch both dynamic sections to their **COMP** mode
- Start experimenting with different compressor settings. Even try switching the **FREQUENCY** switches to more overlapping values to reduce the ‚gap' at the crossover frequency of both bands
- If the compression is too heavy on the first step of the **THRESHOLD** control, try reducing the overall threshold with the **MASTER THRESHOLD** control in the output section.
- Deactivate the **DRY MUTE** button
- Now add a little direct signal with the **MIX** control in the output section. This is a way you can achieve parallel compression
- You will notice that by restoring the transients of the signal, the very heavy compression sounds more natural.

Artist Presets

Conan Manchester

Conan started his studio journey at the very young age of 4, appearing as a guest performer on his parent's recordings in his uncle's studio in London, England. By 11 years old he was a Tape Op in the summer holidays at the Steamrooms in London, learning everything he could along the way. By the late 80s he was producing and mixing his own music as well as mixing music for clients and bands.

By 1996 he started practicing the dark arts of Mastering and continued to produce his own music, signing to New York's ground breaking label Strictly Rhythm. Over the next 15 years he continued to produce his own music for many independent and major record labels as well as becoming a well-known dance music remix producer for bands like Blondie, Talking Heads, Talk Talk, Grace Jones, Radiohead, INXS and many more.

In 2010 he moved to Dubai, UAE and started his own independent remote and mobile mastering and mixing studio and has continued to master and mix continuously, embracing the changes in modern studio technology and practices to the point where he has been mastering and mixing 100% in-the-box, on headphones, for the last 4 years.

He continues to master and mix for a huge clientele worldwide and with the aim to "pay it forward", he now teaches mastering and mixing techniques and consults on production and music technology.

Eric Racy

Eric Racy is a platinum selling mixer, producer, engineer and remixer with credits and clients that include Katy Perry, Pharrell, Tyga, Jonathan Davis, Akon, 2NE1, Ministry, Lil Wayne, Big Sean, Busta Rhymes, Photek and many more.

Matt Weiss

Matt Weiss is a Grammy-nominated, platinum recording and mixing engineer who's worked with artists like Akon, Swae Lee, Jeremih, Chris Brown, Sonny Digital, Nicky Minaj, Becky G, Anitta, Rick Ross, Farruko, and Ozuna.

Peter Riese - Singer/Songwriter/Producer.

Recorded and produced from 1989 till today hundreds of german TV/Radio productions and records. Worked for artist like Propaganda, Blue Weaver (BeeGees), Frank Kirchner (Herbert Grönemeyer), Danny Gottlieb (Pat Metheny) as well as for big clients like Jim Henson, Deutsche Bank, Henkel or VW. Nominated for the german songriting award 2022.

Sam Sherbin

Sam Sherbin is a mixing engineer based in Los Angeles, California. First cutting his teeth at various studios in New York City, he went on to develop his skills under the mentorship of the late Bruce Swedien at his home studio in Florida. Since then, Sam has scored cuts with international stars like CL, Stray Kids, Rachel Platten, Ebony Obsidian and more. He collaborates closely with the likes of multi-platinum producer and composer Nick Lee, expanding Sam's footprint in both music and network television shows like NBC's Grand Crew.

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