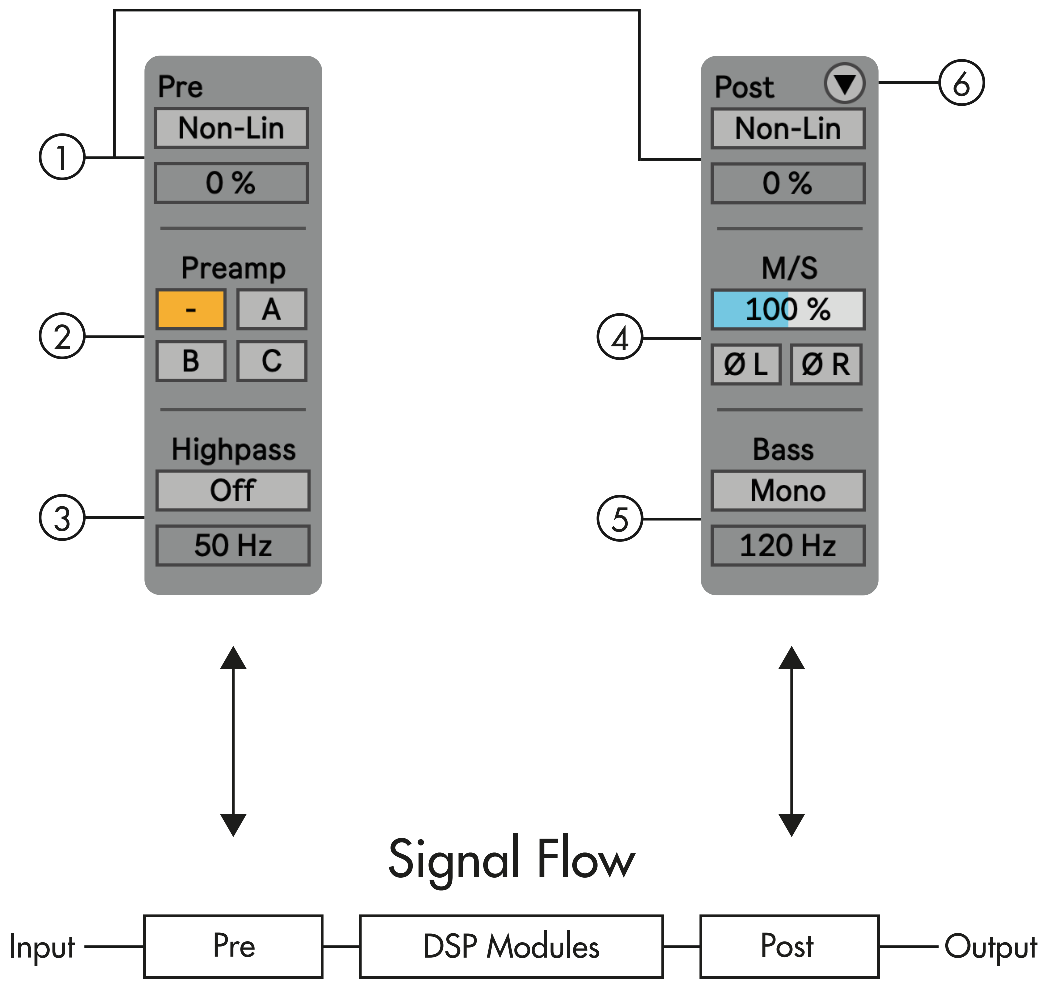
AS Console

by Sabroi

Developed using Max for Live
Ableton 11+ Required

[1] DSP stands for "Digital Signal Processing", which refers to the systems that do the processing of sound. The guide will be using the term when talking about the modules and how some features relate to their systems.

Pre and Post



① Non-Lin Enables a non-linear circuitry on the pre or/and post-stage of the DSP modules, in short, it adds saturation to the signal either before, after the DSP modules, or both. The type of saturation can be selected in the Advanced settings[1]. By default, the DSP system is not enabled which saves CPU processing, once enabled it will be loaded (hence first time enabled there is a slight loading time that happens) and use CPU processing. When toggled off, it won't use any CPU power but the system will remain loaded resulting in no loading time when re-enabled.

② “A”, “B” and “C” are different spectral preamp colors, based on a couple of the abbey roads box units. Having the “-” selected will bypass the preamp circuitry. The amount of preamp modeling can be selected in the Advanced settings1.

[1] Advanced Settings will be shown in page.23 *Advanced: Settings*

③ When the Highpass toggle is engaged, audio will be passed through a Butterworth filter (This happens before the audio passes to the preamps!)

A butterworth filter is designed to have a frequency response that is as flat as possible in the passband.

④ "M/S" portion allows for control of the stereo and mono balance,

0 % = only mono, 100 % = both equally and 200 % = only stereo.

Ø L and Ø R toggle inverts the polarity of either the corresponding left or right signal.

⑤ Bass portion controls the stereo information in the low end, when Mono is engaged everything below the number box parameter underneath will be mono.

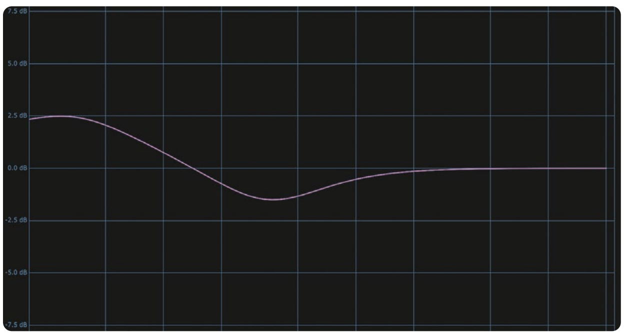
⑥ Toggle when engaged will show the Routing/Advanced panel. In page.20 will go thorgh all its paramters.

EQ

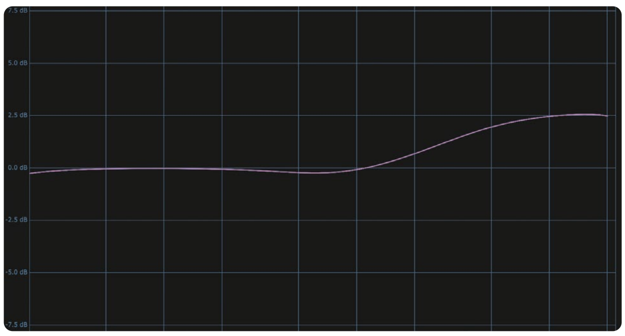
Overview:

EQ is the first DSP module tab in the main panel. It is based on a British class design which can be defined as any equalizer circuit that is designed and built in a way that emulates the classic EQ circuits from the legendary mixers Neve, Trident, Helios, and such.

The general characteristics of such an EQ are how it sounds when boosted. Rather than sounding surgical it has a musical quality. When boosting shelves, there are certain attenuation curves that appear (Side product of the analog hardware EQ units)



Ex 1: 2.5dB low-shelf boost



Ex 2: 2.5dB high-shelf boost

During the 60s and the 70s, music from Britain was characterized as having a more “raw” sound than the American counterpart. A characterization would be that British music had more mids while the Americans had more low-end and high-end, making it sound more “high-fidelity”

Parameter

Setting Gain:

The first row of knobs will set the boost or attenuation gain in dB,

By default, the Low and High gain are shelves while Mid gain

is a bell.

Change EQ Shape:

When holding down shift, there will be selectable bell shapes

for Low and High gain.

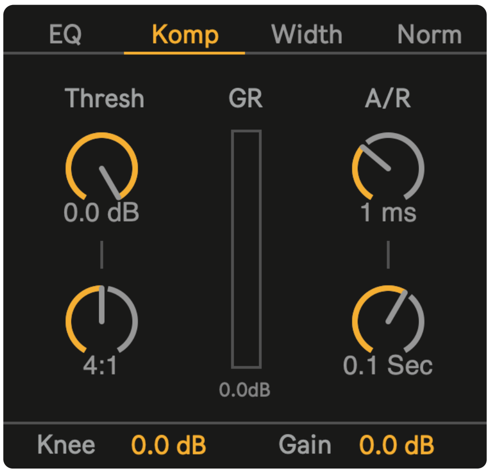
Setting Focus for Gain or/and Freq:

The sliders presented in Focus tab will deviate

from the value set by the knobs in Equalizer tab.

This allows for more surgical dialing in precision if required.

Komp (Boradband Mode)



Overview:

The Komp DSP module is the second tab on the main panel. There are both a single-band and multi-band option(can be toggled between in Advanced settings shown in ID).

Compressors can be used in a vast variety of ways, ranging from dynamic perception in larger time scales, to quick response time affecting the perception of sound itself as something different/distorted.

It can be used transparently to gain ride and limiting(limiter is a compressor with inf ratio and fast response) to apparent and stylistic

Design:

The compressor algorithm is based on

Josh Reiss "Under the Hood of a Dynamic Range Compressor" from Centre for Digital Music
Queen Mary, University of London[1]

and Mathworks audio document reference for "Compressor System Object"[1]

Translated to code by user TMHGLND(_t.mo)[1] and slightly modified calculation for obtaining

the smoothing coefficient for a millisecond time value.

[1] Links to the papers and articles mentioned here are at the bottom of "www.sabroi.com/console"

Parameter for Broadband Mode

Threshold:

Thresh parameter sets the threshold level above which gain reduction starts. Any signal exceeding the threshold will be reduced in gain relative to the ratio of the signal.

(Gain relativeness comes from the value set by ratio parameter)

Ratio:

Sets the ratio of input change to output change when the signal level is above the threshold.

For example, 2:1 means that for an input level increase of 2 dB the output will only go up 1 dB, or for a rise of 4 dB, the output will only goes up 2 dB.

(Scaling will be greater when set to higher values... 4:1, 8:1, and so on)

Attack and Release:

The attack (top parameter) determines how quickly gain reduction rises,

while release (bottom parameter) determines how quickly gain reduction falls.

Knee:

knee refers to how the compressor transitions between non-compressed and compressed states. The transition happens gradually from unity gain to the set ratio.

(Soft-knee for more transparent compression.

Hard-knee for more effect)

Makeup Gain:

When utilizing a compressor, the main objective is to make louder sounds softer and the softer sound louder. When the compressor is working it's only doing the first part (making the louder sounds softer)

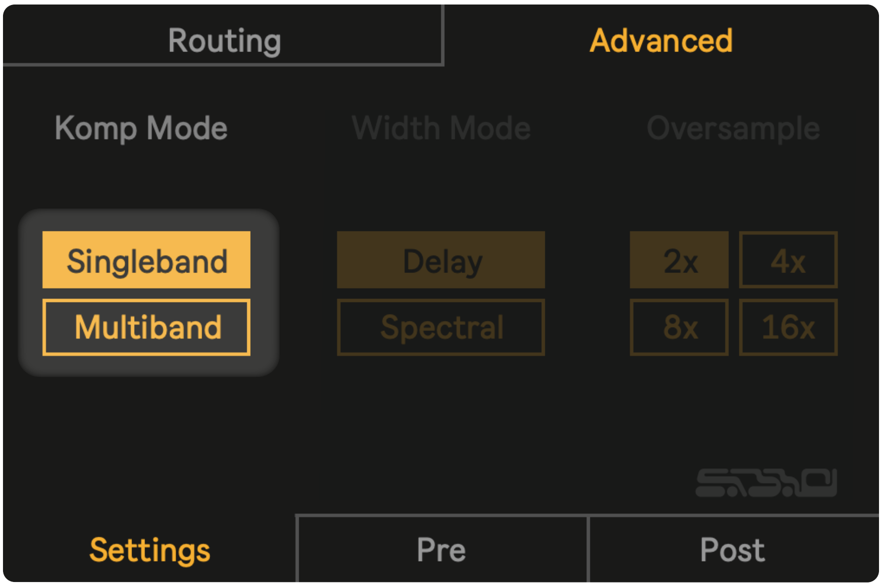
To compensate for this, the makeup gain control simply boosts the level of the output signal by a set number of dBs.

Gain Reduction Meter:

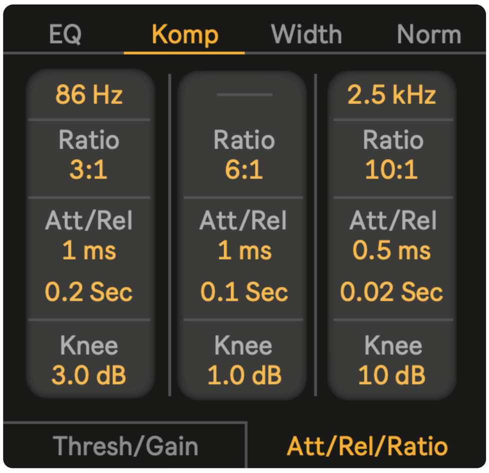
The GR meter shows the applied amount of gain reduction

It is a good practice to listen rather than look at the meters, but hearing the subtleties in compressor action takes experience, and GR meter can often help.

Komp (Multiband Mode)



Once the Routing/Advanced panel is unfolded by clicking . In the Advanced tab and then in Settings sub-tab there is an option to enable Multiband mode. It will change the UI of the Komp main panel, where the syntax will be for each column of the parameters from left to right, corresponding to low-end to high-end



A multiband compressor refers to a compressor applied to specific frequency ranges split by a crossover filter. In the case of AS Console, there are 3 frequency ranges corresponding to lows, mids, and highs. By default, the settings of the compressors are set to mimic the native Multiband Dynamics OTT preset except for the threshold and gain parameters.

Parameter for Multiband Mode

Threshold:

Threshold parameters set the threshold level above which gain reduction starts for individually defined frequency ranges.

(See “Parameter for Broadband Mode” for technical definitions)

Makeup Gain:

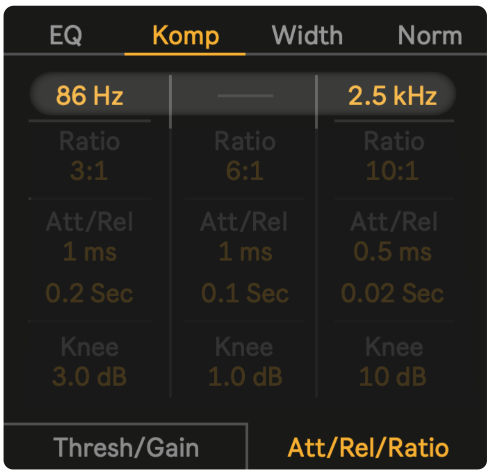
The makeup gain control simply boosts the level of the output signal for individually defined frequency ranges by a set number of dBs.

(See “Parameter for Broadband Mode” for technical definitions)

Crossover Filter:

The crossover filter has two parameters, the one to the left defines the low frequency range (everything up to the specified frequency falls under the low frequency range).

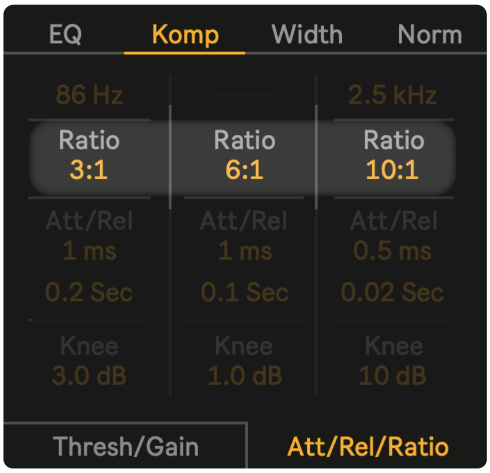
The right parameter defines the high frequency range (everything above the specified frequency falls in the high frequency range). The in between falls into the middle range!

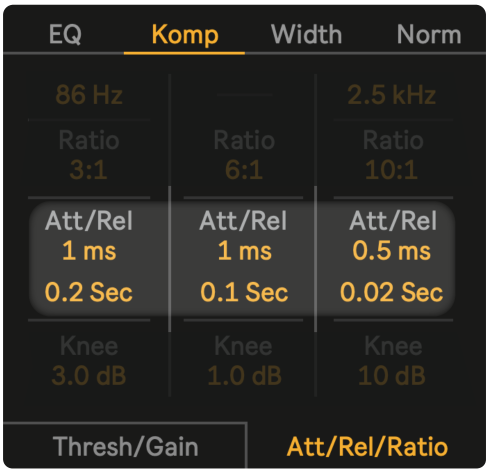


Ratio:

Sets the ratio of input change to output change when the signal the level is above the threshold for individually defined frequency ranges.

(See “Parameter for Broadband Mode” for technical definitions)





Attack and Release:

The attack (top parameter) determines how quickly gain reduction rises,

while release (bottom parameter) determines how quickly gain reduction falls.

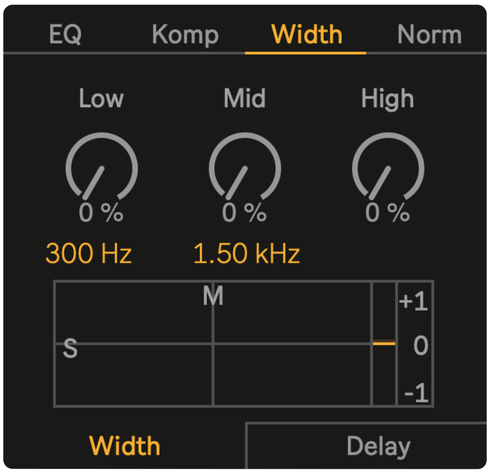
Knee:

knee refers to how the compressor transitions between non-compressed and compressed states. The transition happens gradually from unity gain to the set ratio.

(Soft-knee for more transparent compression.

Hard-knee for more effect)

Width (Delay-Based)



Overview:

Width is the third DSP module tab in the main panel. This module wouldn't be something you would find in a studio's console. For modern applications, there are lots of benefits to having stereo-generating tools in one mixing/sound design arsenal.

There is a crossover filter splitting the signal into three different frequency ranges. The parameters sets how much side signal will be generated from the incoming signal (it will generate a new side signal)

There are two different algorithms, we will start with the delay-based mode first. Its skeleton is based on the “Known Decoration Solutions” section from “Synthetic Ambience in Parametric Stereo Coding” AES Convention Paper[1].

The newly generated side signal is going through various processes to make it non-linear as well. The results makes it sound more natural blending it with the original source signal. Each band has slight variations on processes variable. One of which there are controls for in the Delay sub-tab, within the Width main tab.

[1] Links to the papers and articles mentioned here are at the bottom of “www.sabroi.com/console”

Parameter

Side Signal:

The first row of knobs will generate side signal within the defined frequency range.

The side signal gets added to the input signal.

Crossover Filter:

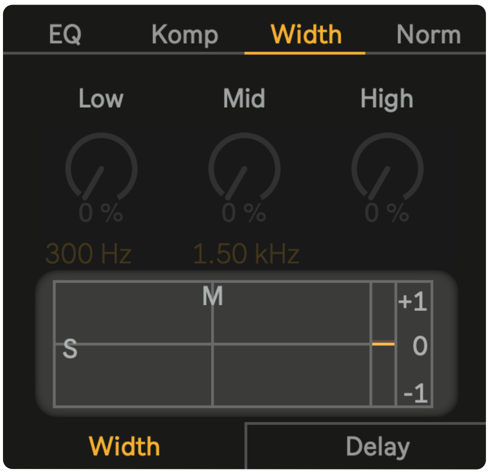
The crossover filter has two parameters, the one to the left defines the low frequency range (everything up to the specified frequency falls under the low frequency range).

The right parameter defines the high frequency range (everything above the specified frequency falls in the high frequency range). The in between falls into the middle range

Vectorscope and Phase Correlation Meters:

The center display is a Vectorscope, the mono signal drives the horizontal deflection of the display, and the side signal drives the vertical deflection. Phase correlation meter shows the overall amount of total phase offset between channels,

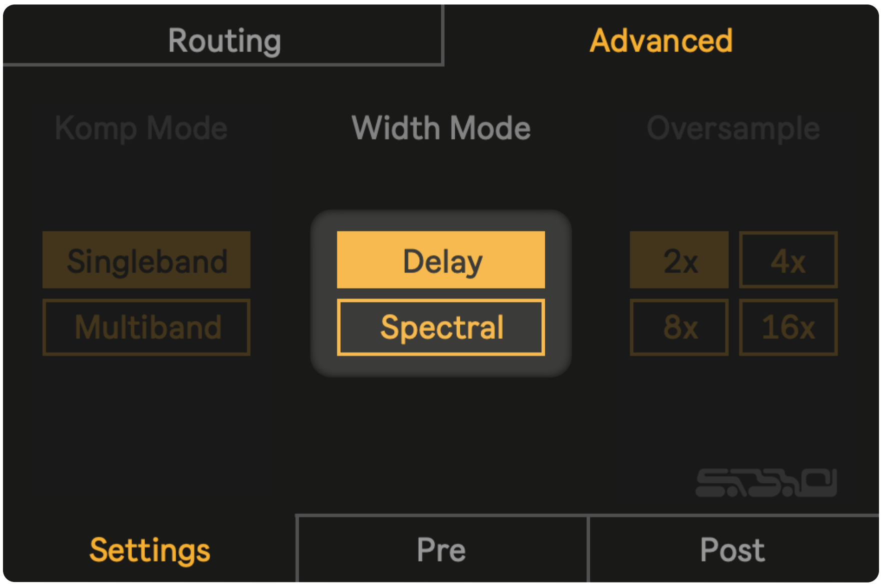
on a scale of -1 to +1.



Delay Length Control:

The sliders enable fine-tuning of each delay length for each corresponding frequency range to vary the spatialization. By default, delay lengths are optimized to each frequency range to ensure perceivable transients get masked.

Width (Spectral-Based)



Once the Routing/Advanced panel is unfolded by clicking . In the Advanced tab and then in Settings sub-tab there is an option to enable the Spectral algorithm. It will change the UI of the Width main panel, where the knobs will be replaced by a parametric EQ made out of sliders.

Instead of a crossfilter, there's a filterbank with 12 bands, each slider represents a band. The amount at which the slider is set to determine the level of the newly generated side signal within that frequency band.

Controls for these bands include their bandwidth and focus (deviation from a frequency)

The center display stays the same (*Width Delay-Based: Parameter* p.13 for technical details about the display)

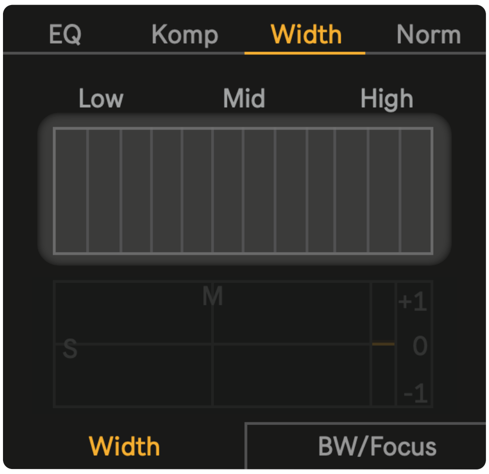
Parameter

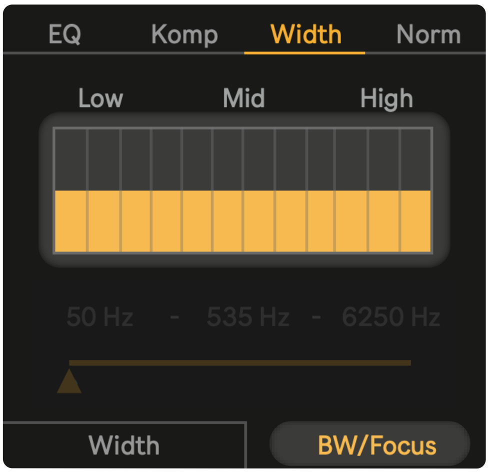
Side Signal:

Each slider corresponds to a frequency band, where the

the amount is set by the slider's value.

Holding Shift while clicking and dragging will move all the sliders simultaneously while maintaining their relative values.





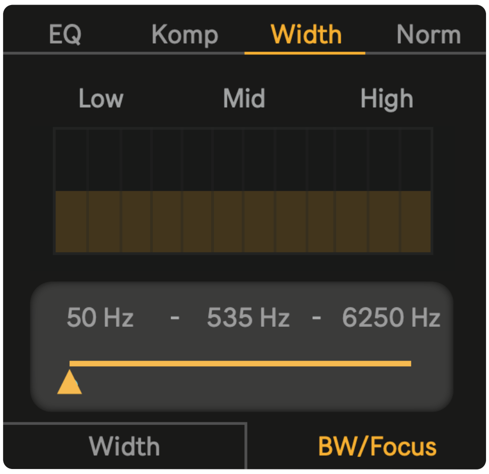
Bandwidth:

Sliders inside BW/Focus sub-tab represent the width of the corresponding band. A low value will be a wider band, while a higher value will narrow the band.

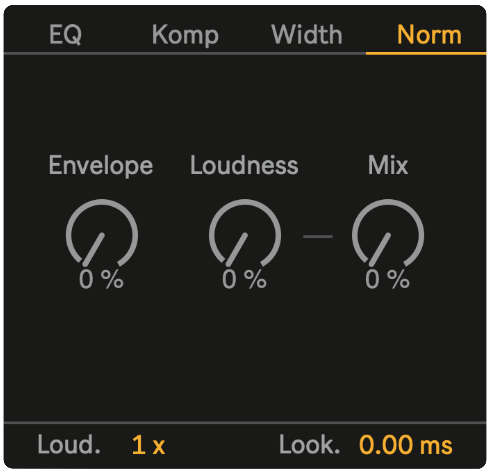
Holding Shift while clicking and dragging will move all the sliders simultaneously while maintaining their relative values.

Filterbank Frequncy Range:

The grey numbers specified in hertz indicate the lowest, middle and the highest banks. The vertical slider underneath shifts the range of all the banks and will be displayed above



Norm



Overview:

Norm is the last DSP module tab in the main panel. Similar to the Width module, it wouldn't be something you would find in a studio's console. It's a fairly unique effect that comes from "Electronic Music and Sound Design Volume 2" by Alessandro Cipriani , Maurizio Gir.

Described as a "Live Normalizer" and "It is not to be confused with the normalizer that is found in hard disk recording software, which increases a sound's amplitude by bringing its peak amplitude value to 0 dB, proportionally increasing the amplitude of all other samples".

In other words, the common normalizer will bring the loudest part of the whole audio to 0 dB, bringing everything else up in amplitude proportionally.

Live Normalizer brings both signals above -120 dB toward 0 dB as well as signals above 0 dB down to 0 dB, which can be stated as the Loudness parameter sets the mean amplitude

Parameter

Envelope:

Similar to Attack and Release parameters from *Komp Broadband: Parameter* p.7. Envelope paramter functions as one knob reaction time for how slow/fast the normalizer will bring up and/or down the amplitude. Low value will be a slower reaction time while a higher value will be a faster reaction time, this applies to both attack and decay time.

Loudness:

The Loudness parameter will more aggressively push the signals above and below 0 dB closer to the mean (0 dB)

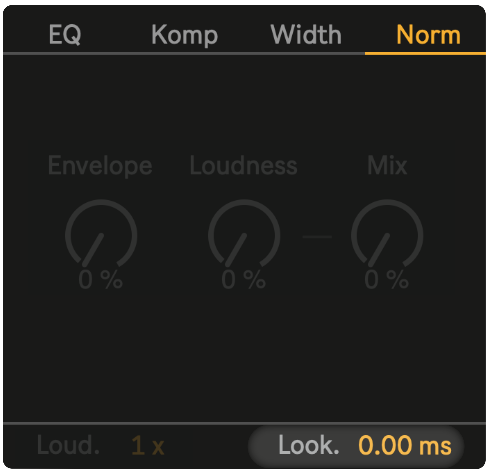
Mix:

Blends the non-dynamicly altered signal with the normalized signal.

This allows for parallel processing, a technique commonly done with compressors and pioneered by Bob Katz.

Loudness Scaling:

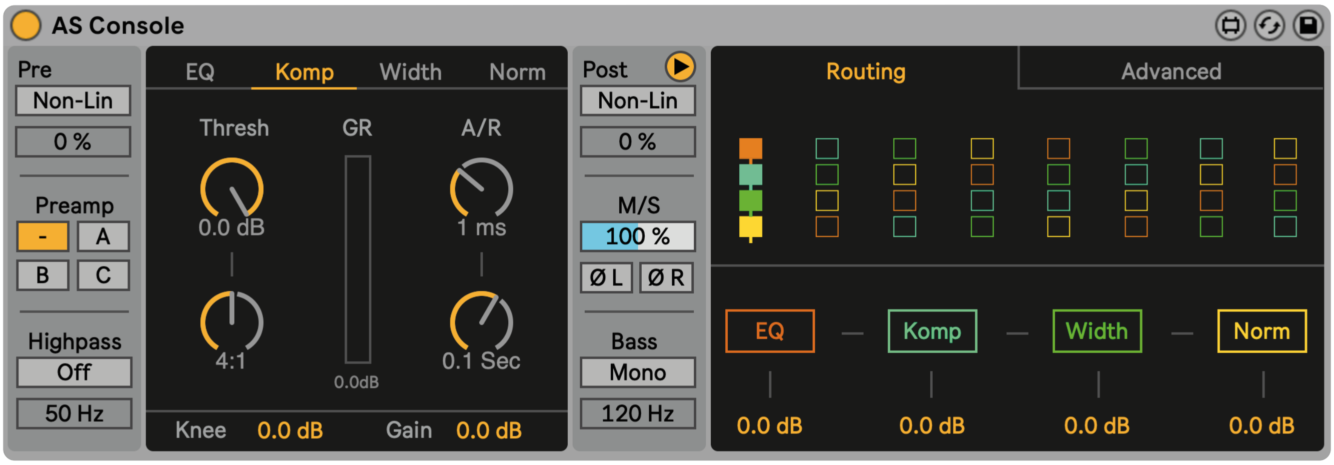
Loud. number box sets the scaling factor for the Loudness parameter. The effect of the Loudness parameter will be greater when Loud. number box is set to 2 x rather then 1 x



Look-ahead:

Look. number box will simply delay the signal before the gain computer applies its process to the signal. The results of this will be that the reaction time of the gain computer will be ahead of the signal to be processed, thus allowing it to catch transients in advance.

Routing



Overview:

AS Console has an additional panel for more advanced features such as routing, oversampling and etc.

By clicking , the panel will unfold. The starting tab shows routing configurations and an overview of the DSP modules states and input gain.

The graphics are color highlighted to make it clear before clicking the routing configuration how the arrangement of DSP modules looks like. When changing the routing the main panel and the overview of DSP modules will arrange themself accordingly.

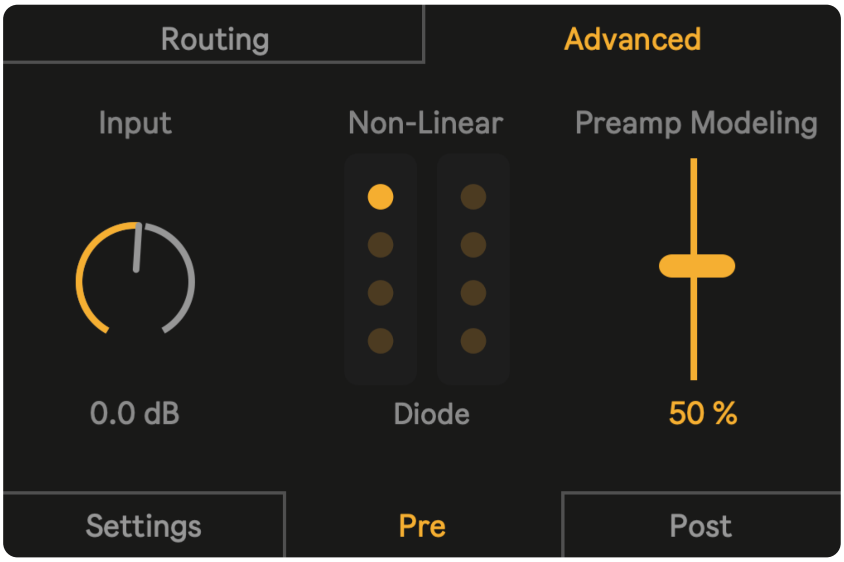
Mixer/States:

The gain introduced by the number boxes is the input to the module itself, ex. +3.0 dB gain boost on Komp will mean at the end of EQ there will be a boost of +3.0 dB. (if EQ precedes Komp in the routing)

Each DSP module has a toggle box that reveals the state of the DSP either on or off. They can be manually toggled or, by changing a parameter value within the corresponding module to a non-initial value.

When the state of a module is off, the DSP within that module will stop running, resulting in saving CPU process. (except for the norm module, because it's already very light. The signal will not go through the module if the the state is off)

Advanced Pre



Overview:

Within the Advanced tab, there are three sub tabs which are Settings, Pre, and Post. We will start with Pre, then Post, and last Settings.

There are three parameters in this tab (both with Pre Non-Lin and Preamp toggled on as well, as to enable the latter two parameters)

From left to right:

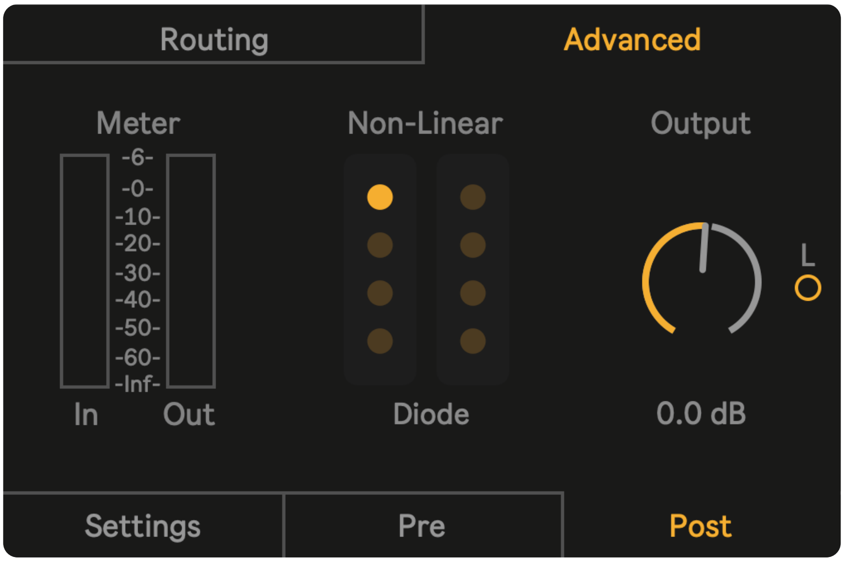
1. First knob is Input, this attenuates the incoming signal to AS Console.

2. Non-Linear icon tablature allows for different algorithms to be chosen for the Pre drive stage. Each orange dots represents a different algorithm. When the Non-lin circuitry is enabled and driven, the characteristics of the harmonics caused by the drive will be determined by the chosen algorithm, which are as follows:

- Diode:	Waveguide Modeling (Analog Modeling)
- Salley:	Non-Linearity modal from a Sallen Key Filter
- Var S:	Tangent Function
- Sat:	Tangent Function
- Sin:	Sine Function (Waveshapping)
- Jos:	Cubic Nonlinear Distortion
- Tom Szilagyi:	Sigmoid
- Gloubi-boulga:	Waveshaping

3. Preamp Modeling slider sets the amount of Preamp color to be applied by one of the A, B, or C modes which are spectrally based on the abbey roads box units, which were simply known as “Brilliance” or ”Presence” boxes at the time. These were simple passive equalizers that were portable versions of the grey RS127s.

Advanced Post



Overview:

There are three parameters and a read-out meter in this tab (with Post Non-Lin toggled on, as to enable the latter two parameters)

From left to right:

1. First thing is the Meter, which is a peak level indicator. It shows the peak level reached. Within the range of -70dB <-> +6dB.

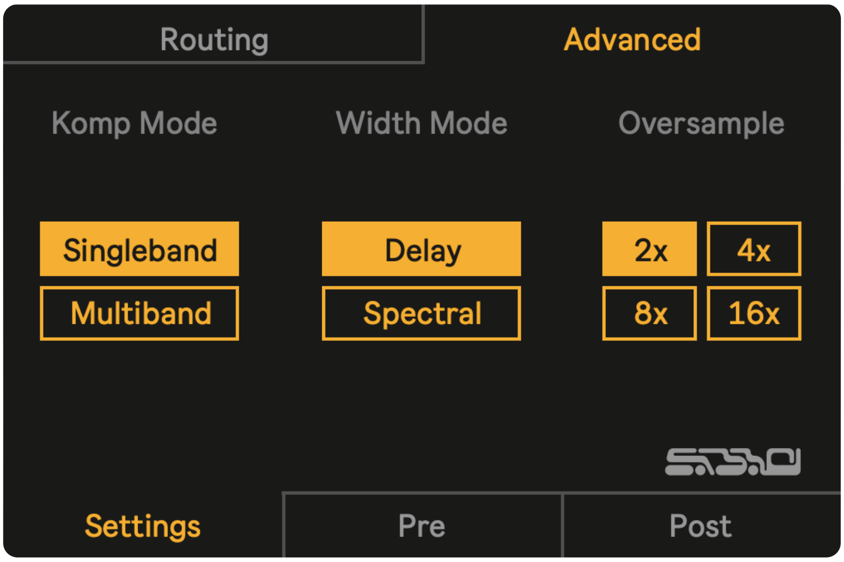
2. Non-Linear icon tablature allows for different algorithms to be chosen for the Post drive stage. Each orange dots represents a different algorithm. When the Non-lin circuitry is enabled and driven, the characteristics of the harmonics caused by the drive will be determined by the chosen algorithm, which can be referenced in Advanced: Pre[1]

3. The output knob attenuates the outgoing signal. Note that this gain attenuation stage happens before the Post Non-Lin circuit and the optional Limiter.

4. The small toggle with an L above enables a limiter to the very end of the signal chain. This can assure that the signal doesn't cause any unexpected peaks.

[1] The algorithms will be shown on *Page 21. Advanced: Pre*

Advanced Settings



Overview:

There are three tablatures, which affect the individual modules or overall algorithms.

From left to right:

1. First tablature changes the Komp mode from a single compressor to a multiband compressor.

2. Second tablature changes the Width mode from a delay-based algorithm to a spectral algorithm.

3. The last tablature sets the oversampling for the Non-Lin circuits. Oversampling is common when there are any processes introducing harmonics, which can cause aliasing when their frequencies are high enough causing foldover (these are the ailising frequncies)

I want to thank all people that have been giving thoughts and feedback while in develpment in all forms.

It wouldn't be the same device it is today if it wasn't for people like

daywaiter
@codmart

ALEPH
@ALEPH_Sound

Desembra
@DesembraMusic

Christian Kleine
@KleineMusic

nathan
@nthnblair

shrey
@tallbrowndude

And all the countless emails and messages white all sorts of helpful debugging and feature ideas.