HoRNet SAMP *Spatial Audio Master Processor*

HoRNet SAMP is a processor designed by the Italian engineer **Antonio Porcelli** to work with Dolby Atmos mixes and resolves one of "problems" that engineers find when approaching this new format: there is no master bus. This big difference with conventional stereo mixes makes it impossible do to any of the classic adjustment you do on a stereo mix.

Imagine you have 128 different tracks and you want to lift the "highs", you should go one by one and adjust the EQ, SAMP resolves this issue by providing four classic master bus processors that are linked together, so a change on an instance of SAMP is reflected on every other instance, simultaneously.

Packaged into a single plugin you can find: an **equalizer** (using the same technology found in our TotalEQ) a **compressor** (using the same technology of our SyncPressor) a **clipper** and a **limiter** (using the same technology of our Magnus MK2). These processors take the audio from each track and apply the same settings to every track, simulating the effect of a master bus processor.

The order of the effect can be rearranged dragging their position on the right part of the plugin interface and their meters and side chain are fed with the sum of the input of every instance of the plugin, mimicking the behavior of a master bus processor.

In the right part of the interface you can find the list of instances of the plugin available (up to 160 different instances of the plugin are supported) each of the instances has a peak meter built in showing you the level of that track and you can decide if you want to have them all participate to the "virtual" master bus or just some of them by clicking on each one.

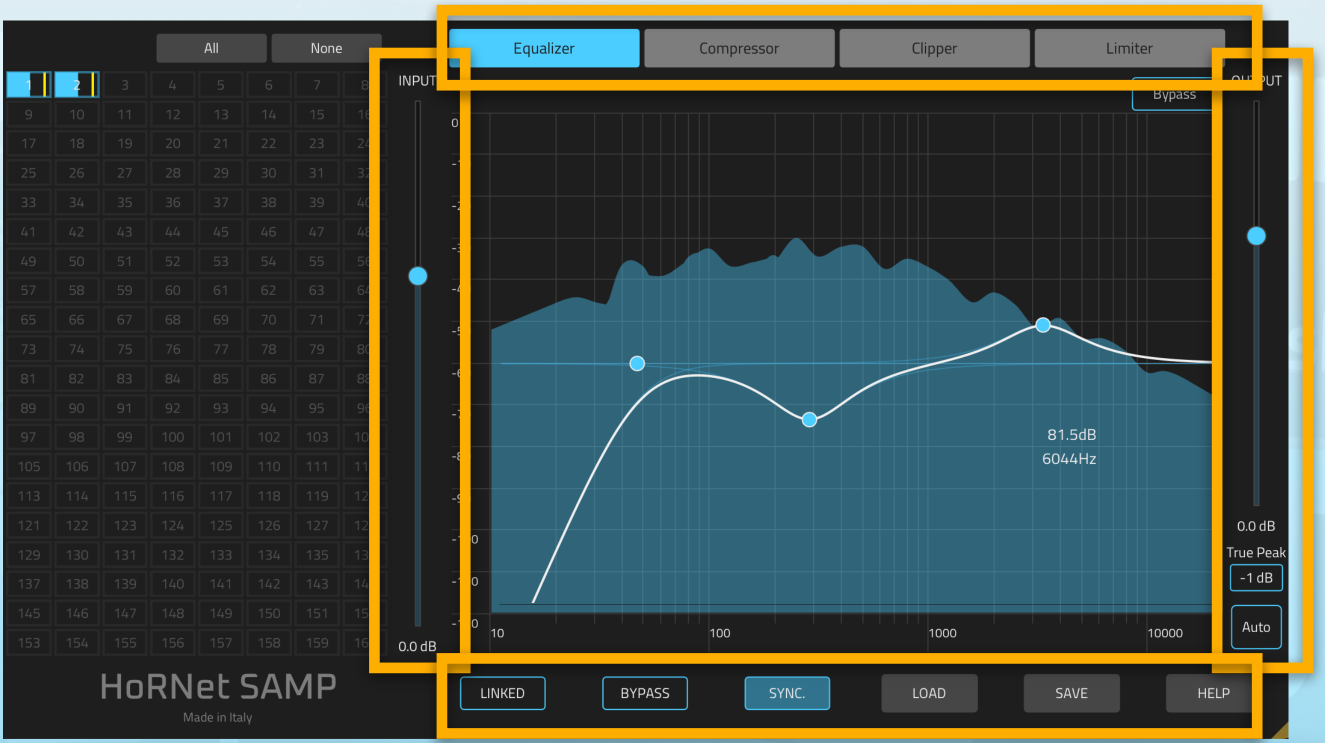
Of course SAMP reads the name of track on which it is used (except for the VST version) and hovering with the mouse on any of the 160 level indicators on the left will show you the track name of the indicator.

SAMP is a mono only plugin but it can work on any DAW, even if it doesn't support Dolby Atmos directly.

Track Selection

The highlighted area contains the 160 small peak meters that show the level of each instance available, clicking on each of them disable or enable the specific instance for the side chain sum.
The “All” button selects all the instances for the side chain while the “None” button disables all. Moving the mouse on any of the boxes will display the track name on the top left corner (only for VST3, AU and AAX)

General controls



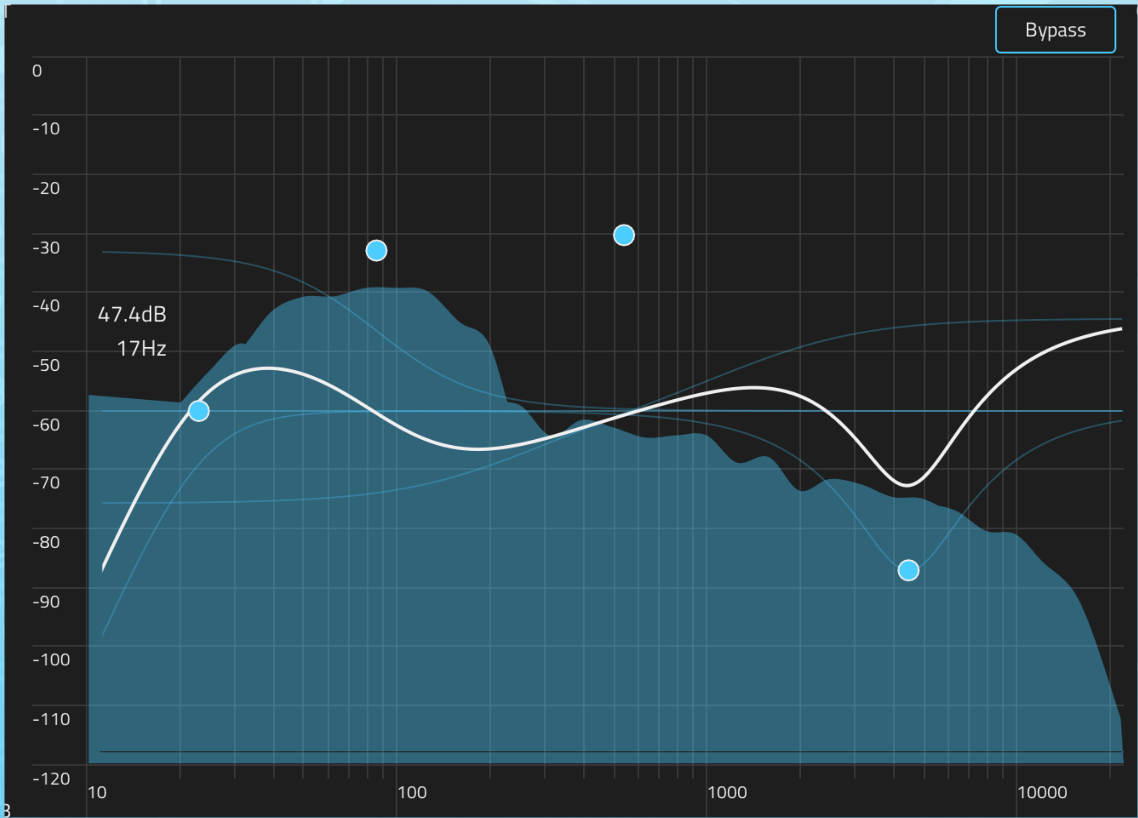
These areas contains the general controls.
On the left of the main plugin view we find the **input slider** and on the right the **output slider**. Adjusting those two controls allows you to change the input and output levels for every instance.
Right below the output slider we placed the True Peak target level and the automatic output gain button. Enabling the auto output will adjust the output gain slider to sit the higher peak at the chosen true peak level.
On the upper part of the interface you can find the **tabs for the module selection**. Clicking on any of the tabs allows you to edit the parameters for the specified module. You can drag the tabs to change their order.
On the bottom part of the interface we find (left to right) :

- **Global bypass button:** this button completely bypasses the processing for every instance of the plugin.
- **Sync button:** this button allows you to chose the type of attack and release times found in the compressor an limiter modules, when enabled you can chose attack and release times using musical time divisions such as “1/4” or “1/8”, of

course relative to the DAW tempo. When this button is turned off you will be presented standard attack and release options in milliseconds.

- **Load button:** this button allows you to load a custom preset for the plugin
- **Save button:** this button allows you to save the current plugin setting s in a custom preset.
- **Help button**: when this button is clicked the online help is enabled. Moving the mouse on every control will show a tooltip with a short description of the function.

Equalizer



The main part of the equalizer, which is based on our TotalEQ plugin, is occupied by the spectrum analyzer, in this case it's a smoothed 1/3 octave bandwidth analyzer with an integration time of 1000ms. We have chosen these values to give you a spectrum analyzer that offers you a general overview of your "virtual master bus", in fact the signal shown here is the sum of all the instances selected for the side chain.

To change the equalizer curve you can create up to six different control point, to create one just double click on the white eq curve, double click a control point again to delete it.
Each eq band, represented by a control point offers a large selection of filter types and different controls for every type.

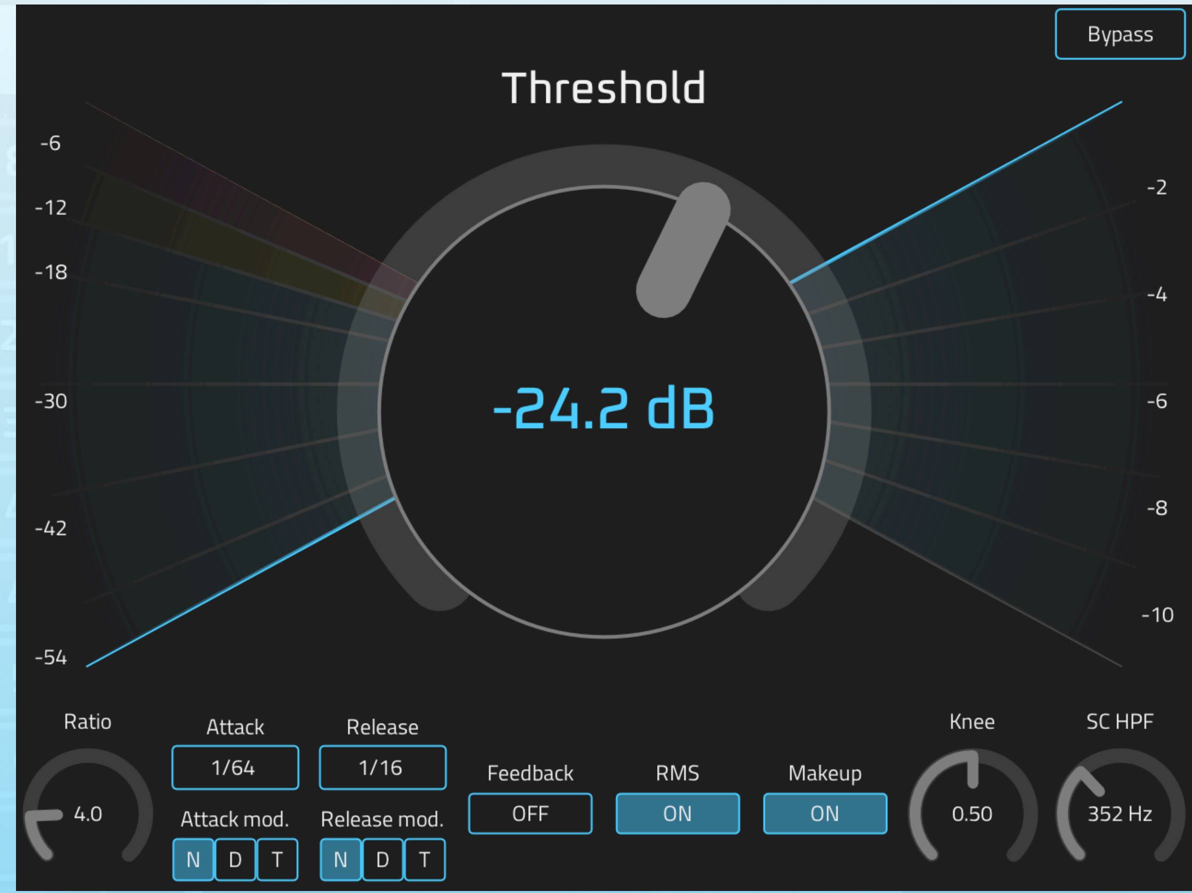
Equalizer controls

Pictured in the image above are all the controls of an equalizer band, not every control is available for every filter type, the bell one is the most complete.

The top row contains the control buttons:

- **D** : is for "dynamic", clicking this button will enable the dynamic mode of the band, threshold, attack time and release time are automatic, just move the dot around until you see the right movement in the curve
- **M** : is for "mute". Once clicked the processing of the band is bypassed.
- L : is for "listen". Once clicked you will only hear the processed sound from this band isolating it from the rest of the equalizer
- **A**: is for "analog", this button enables the analog behavior of the band, this translates in a slight different response curve and dynamic frequency gain and q changes when the signs is louder.
- **S**: is for "saturation", clicking this button will reveal a saturation slider that will allow you to add some distortion to the EQ band, keep in mind that bands are processed in series, so if you add distortion to the first band this will propagate to the next one un til the last.

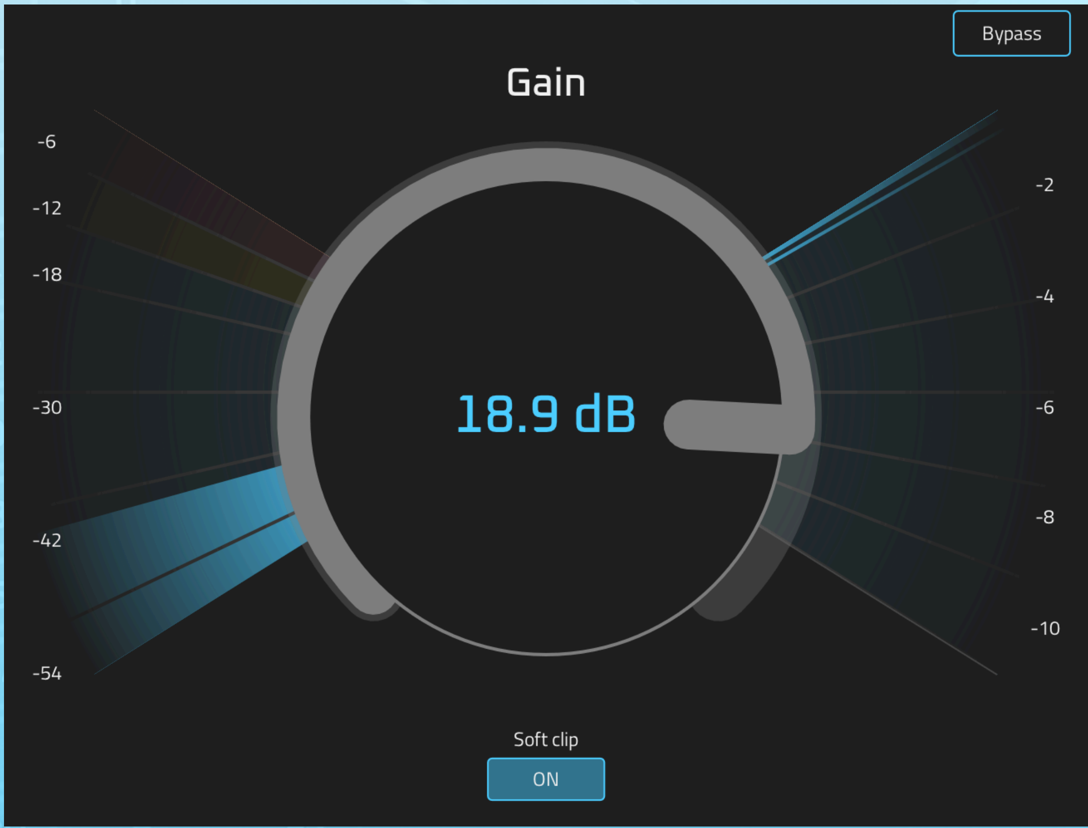
Compressor



The compressor module provides you the standard compressor controls and is based on the code of our SyncPressor, which is a clean digital compressor.
The main part of the interface is occupied by the threshold knob, the side chain meter, the input meter and the gain reduction meter.
On the left we find the input meter that displays the summed level of the instances selected for the side chain, the side chain level (which is different from the input level since it has passed though the attack and release stages) is shown as a blue line around the threshold knob, then that line crosses the grey indicator compression is applied. The amount of the applied compression is shown by the gain reduction meter on the right.
The bottom part of the interface contains the other controls such as the "ratio" attack and release times, a button to choose if the compressor must work as a feedforward one (default) or in feedback mode. Also we have provided a button to enable the RMS mode of the detector (so it reacts less to short transients) and a

knee knob that changes the softness of the transition area around the threshold level. In this area we acan also find the auto makeup gain switch and a sidechain high pass filter.
When "sync" is enabled in the general control area, attack and release times are shown as musical divisions like 1/4 or 1/8, also a "modifier" control is shown next to them, these buttons allows you to chose if you want "natural" length (**N**) "dot" length (**D**) or "triplet" length (**T**).

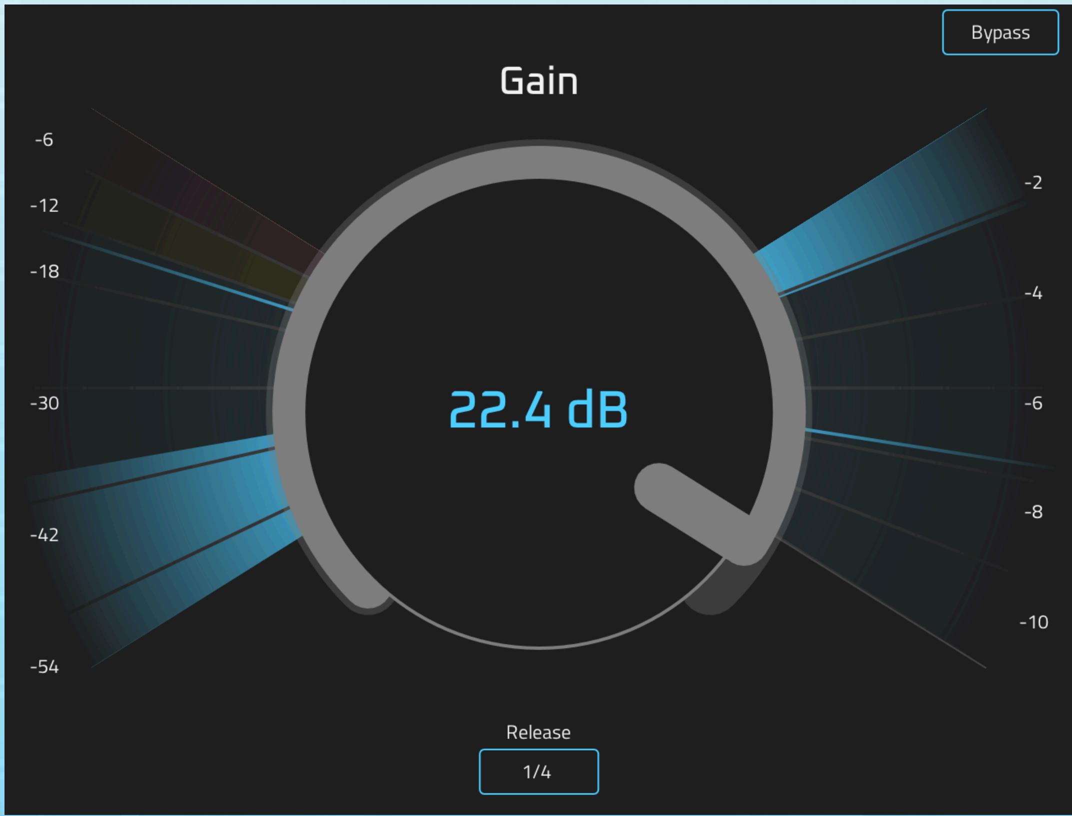
Clipper



The clipper module allows you to smooth out transients in the signal. It works by allowing you to increase the gain until it hit a hard or soft ceiling that adds harmonics to the signal.
On the left you have the input meter that shows the level of the summed instances selected for the side chain. On the right there is a gain reduction meter that show the amount of signal that is being shaved off by the clipper.

The button below the main knob lets you change between a soft and a hard clipper. The soft one has a nice analog behavior while the hard one is a digital clipper.

Limiter



The limiter module is based on our Magnus MK2 plugin and is a brickwall type since it has instantaneous attack time. The release time is handled by three different envelopes that reacts with different signal types, this envelope arrangement allows the limiter module to be as transparent as possible.
On the left of the interface we have the input meter that shows the level of the signal of all the instances selected for the sidechain, in the center we have the large gain knob that increases the signal inside the limiter. On the right we have the gain reduction meter showing you the amount of reduction being applied by the processor. Right below the gain knob we can find the release time control, when the “sync” button in the general control section is turned on, this control show musical time divisions (like 1/4 and 1/8), when the button is turned off you can enter the release time in milliseconds.