

ALEXB A25 MANUAL

ALEXB AUDIO ENGINEERING

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ABOUT ME

AlexB is a one-man company, so:

Hi ! I'm Alex :)

I have been a member of the Acustica Audio community since the 2007, and started Beta-Testing in 2009. I released my first commercial program libraries for Nebula Pro in 2009 because I wasn't satisfied by the sound of the plugins.

What I looked for was a good emulation of the console to improve my music - I have composed a lot of songs for Café del Mar in that years. My first Café del Mar recording was done with a Korg CR-4 only, then the following years I have moved to PC world and Nebula has been found as the plugin of my dreams.

Sincerely at the first test I wasn't satisfied at all by the sound. The libraries was very poorly sampled and the plugin was a little cloudy and flat.

After being in touch with Giancarlo (the genius behind Acustica Audio) and to have said him about my thoughts about what to improve in Nebula, he has promptly given me a new improved release of the plugin. We have continued for the whole afternoon and after some exchanges of test and new releases, finally Nebula became dynamic, open, deep and with life. Thank you Giancarlo !

So, pushed by this experience I've made some of the most highly sought after and rare hardware devices available for use in the digital world while maintaining virtually all of the analog character that makes recording a true art-form. Every sampled hardware piece has been refurbished and modified to improve the sonic characteristics, thanks to my 30+ years of experience in electronics and audio engineering. With hyper-realistic samplings of pristine mastering equalizers, top class consoles, the most sought after compressors, and the rarest vintage devices, I'm proving to the audio community that Acustica Audio sets the standard for the finest sound quality in the digital realm by facilitating a true analog experience with programs that make full use of the VVKT technology.

Please visit the website for more information: www.alex.eu

Thank you !

AlexB... Audio Renaissance.

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Please do not illegally share the program libraries, your financial support allow me to continue in developing. **Be aware: there isn't any authorized reseller of my programs.**

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Thank you

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1. Documentation, Installation and Support

1.1 - Introduction

Thank you for purchasing the AlexB library programs for Nebula.

Now you have one of the best professional high quality audio software. I have spent countless hours to develop these no-compromise programs to give you only the best sound and the most realistic “feel” as possible to the real hardware. I'm confident that this plugin will help you make better and more professional mixes (while enjoying yourself even more)... Because: Sound First !

If you have any trouble with the software please do not hesitate to contact me at:

support@alex.eu

1.2 - Overview

Despite the digital revolution in the pro audio industry, many of today's top albums are still mixed on analog consoles and with analog outboard gear. Mixing into an analog desk just sounds better. Everything sits better in the mix, there is more weight to the bottom, and the overall sound is more three dimensional.

Analog devices produce electrical artifacts that affect frequency response, add harmonics, cause signal clipping and increase noise. These artifacts, which audio engineers often consider the character of a particular device, result from a combination of factors such as component grade, technology type (i.e. vacuum tubes, ICs, transistors), power supply specifications, equipment casing and other variables.

Depending on the circuit characteristics, input signal frequency response varies. Some circuits cut frequencies, others boost them. This behaviour is part of the overall device character and should not be confused with user adjustable EQ.

Total harmonic distortion (THD) is based on the levels of the odd and even harmonics of an input signal, usually at a level much lower than the fundamental level. THD balance and decay are circuit dependent, and thus differ from device to device.

Cross-Talk and Noise are two elements which every designer tends to avoid to not affect the audio quality. Since in the analog world they can't be avoided, fortunately in digital domain with Volterra Technology I have reduced the noise at less of -120dBfs and completely avoided Cross-Talk during the sampling.

The result is an optimum full quality sound from a like-new working condition hardware.

I have recreated these non linearity characteristics into these programs by sampling the units in excellent condition. Your tracks will become more alive with the classic vibe of a real hardware and you may notice that your mixes may take on an almost magical quality with punch, glue, and dimension that you didn't hear with your other algorithmically based plugins.

1.3 - Sampling Process

I believe that "Vectorial Volterra Kernels Technology" is the path of the future and will enable analog sound to be implanted into digital DAW environments with real harmonic content and analog vibe. In my creation of these Nebula Programs, I use only top notch modern and vintage gear, precisely sampled by using my own proprietary technique with custom converters I have built specifically for NAT3 which outperforms top notch commercial converters. Ultra filtered and stable AC supply, high end cables, with particular care to the connections, levels and impedance matching were used to translate the sonic qualities of this priceless devices into the Nebula software technology. Every captured sample is analyzed and carefully listened. Every volume change, gain change, frequency change is tested and accurately programmed without destructive digital processing for optimized sound and then compared to the original device. The result is a virtually indistinguishable digital replication of this landmark device.

The hardware is sampled without introduction of noise or aliasing. The thinking behind this process is to provide the full quality of the analog behavior, which means placing all emphasis on quality over cpu resources. The process is extremely efficient and optimized to be used on current computer technology with a forward thinking to the future of more powerful systems, but this will be a more cpu-intensive device than your typical software. Consider the value in having even one instance of the original unit in your hardware rack and choose to see the true value in having the best sound that technology has to offer.

The preset doesn't sound processed, harsh or digital as many plugins do, but instead it sounds like a natural extension of the original audio, gluing your tracks in the mix with an analog vibe.

Some plugins make your recordings sound like digital.

Some plugins are supposed to make your recordings sound like analog.

THIS plugin helps make recordings sound like MUSIC !



1.4 – Ultimate Edition

Ultimate Edition is the last evolution in sampling and programming to answer at the customers demand for a program library which covers the whole sample rate range used in musical production: 44.1kHz, 48kHz, 88.2kHz and 96kHz.

To make this a new resampling of the units was necessary, so why not to improve the audio quality also ?

A revision of my custom-built self-made converters has been done, which employs a non commercial chip used in military satellites - thanks to my previous work as chief technician into electronics and telecommunication lab for over 15 years, where called me "the doctor"... (who?), I had access at that technology - and a new sampling and programming technique has been developed with custom template and build up process.

The new Ultimate Edition library allows better sound with less - unnecessary - kernels, i.e. less CPU/RAM load and less latency which doesn't make necessary anymore the HQ/LE version; new and useful set of presets to expand the possibilities by leaving room for creativeness.

1.5 – Compressor's Regeneration

My mastering studio is full analog and the reason is because I never liked digital compressors, neither vst or Nebula world. So I was reluctant to revise my old releases of compressor programs library for Nebula, BUT some beta-testers and a good friend have pushed me to try to improve the behavior and audio performance as much as possible.

As usual my wish was to release a good program library with the authentic behavior and sound, to give the opportunity at who can't own the hardware unit to have that expensive sound in the box, at a fraction of his cost.

So, I have tried several ways to make work better my "bad" compressors, after five new internal releases that I haven't liked, I have decided to start from zero.

A new complete multi-template has been developed, without the use of my obsolete ten years old NAT3 release.

Every single sample, program line, every parameter and digit, every compression curve, attack and release shapes have been programmed, edited and calibrated manually, then tested, compared and listened with the original unit.

When the program was ready, a fine tuning has been done with double check: analyzing and listening session. This has been done for every single input/record changed/edited bit.

A very long and hard work, the same I do when I modify/upgrade the hardware !

The whole development's process has required over 960 hours of work, sampling, programming, testing, thus converting into digital world over 6000 records manually (!!!).

The final result was a musical compressor with the reaction and behavior very close at the sampled unit, with the exact curve's shape of the compression, attack and release which follow the envelope-follower. The attack curve's shape gives the character by bringing the transient - it makes recognizable the classic compression sound of the hardware: snap, smooth puff, etc... - while the release curve's shape gives his breath at the music.

At the end of this fantastic adventure:

thanks to all many customers that have written me emails by showing their respect and support for my work, as well as the beta-testers and friends.

1.6 - System Requirements

- Intel or AMD CPU based PC or MAC computer
- Free space on Hard Disk or better SDD (library size depending)
- Nebula3 v1.3.903 or Nebula4 with installed commercial license
- HD or SDD for Programs Backup

1.7 - Installation and file BACKUP

After downloading, unpack the files and **make a safe backup** of the library. I recommend to use a Toshiba Canvio 2.5" HD as well to do a regular backup of your system with Acronis True Image.

Copy the files manually, *.N2P into \programs folder and *.N2V into \vectors folder.
Clean the \temp folder in the main root nebulatemprepository.

1.8 - The Skin

Skins for Nebula3 and Nebula4 have a cost but they are included in the libraries as **gift**.
Skins are developed by external 3rd party and I'm not responsible about them.

To install the skin into Nebula3:

- 1 - copy the *.N2S file into the root skin folder
- 2 - run your DAW and open Nebula
- 3 - go into MAST Page
- 4 - set the Skin to ALEXB_N3
- 5 - click on save and reload Nebula

To install the skin into Nebula4:

- 1 - copy the *.N2S file into the root skin folder
- 2 - copy the Properties files into Properties root folder (present only if special N4 skin is available)
- 3 - run your DAW and open Nebula
- 4 - load a preset

After installation it's recommended to clean the \nebulatemprepository\temp folder.
Now you are ready to go at the next step to read how to use your new Nebula library!

Remember:

Scientific studies have proven that the brain is influenced more by the visual stimuli than acoustics.
What you see is not what you hear. In mixing and mastering nobody can hear your screen.

2. General Use

2.1 - Parameter Settings

Some parameters must be set into Nebula to work correctly with AlexB Programs.

Nebula3

The best way is to make copy-and-paste of Nebula3.dll and Nebula3.xml (or whatever is the name of your installed Nebula plugin has) then rename both copies as AlexB-N3.dll and AlexB-N3.xml.

Now set the following parameters by editing the AlexB-N3.xml file:

```
<AHEADLENGTH> 6000 </AHEADLENGTH>  
<RATECONVERSION> 4500000 </RATECONVERSION>  
<OFREQD> 11 </OFREQD>  
<SKINNAME> ALEXB_N3.N2S </SKINNAME>  
<DSPBUFFER> 8192 </DSPBUFFER> (optional for better audio quality)
```

click on save and load the AlexB-N3 in your DAW.

Nebula4

The best way is to make copy-and-paste of N4.dll and N4.xml then rename both copies as AlexB-N4.dll and AlexB-N4.xml.

Now set the following parameters by editing the AlexB-N4.xml file:

```
<AHEADLENGTH> 5000 </AHEADLENGTH>  
<OFREQD> 11 </OFREQD>  
<OTIMED> 5 </OTIMED>  
<LEDSPEED> 3 </LEDSPEED>  
<DSPBUFFER> 8192 </DSPBUFFER> (optional for better audio quality)
```

click on save and load the AlexB-N4 in your DAW.

You can use "Setups" function inside Nebula4 by following this video instruction as example:
www.alexeb.eu/video/AlexB_SPQ.mp4

settings for this library:

METERTYPE 2 (compressor)

METERTYPE 7 (line out)

NOTE: MAC users find more info on the "MAC Nebula4 Setup Addendum" on the website:
https://www.alexeb.eu/docs/_AlexB_MAC_Nebula4_Setup_Addendum.pdf

2.2 - Off Line Process

If your DAW isn't powerful or you want/need to freeze or export processed audio tracks I strongly recommend the Free NEBULASETUP2 by Zabukowski: <http://zabukowski.com/software/>

2.3 – Web Tricks, Trolls, Haters, Rumors, Myths...

TIMED and Kernel Length

On the forums you will find many “tricks” which theoretically will improve sound and performance:

WRONG ! Please leave the libraries at their original conditions !

Results from these changes are widely varied and often lead to very undesirable results, as losing dynamics and details.

The presets are programmed to sound as close as possible to the original sampled hardware. **If you change any parameter than the sound changes and it will be very different from the original sampled unit.**

Don't care, I don't, about all these rumors, tricks, myths and a tons of misinformations from wannabe, trolls, haters & Co. which make things up about what someone has said by pretending to know everything about what he thinks, how he works, what he does... And what he have to do.

My programs are perfect as is, they don't need any fix or debug*.

I'm referring to MY programs only. I haven't the arrogant presumption to say how other 3rd party developers work, think, or to tell them what they should do.

I sample at the right level and matched impedance, my converters allow to set and fine tuning these parameters.

I use real 0dBVU = -18dBfs reference to capture the real spectral and harmonic contents. Sampling in this way gives the best and authentic sound compared to the sampled unit but also some minor and irrelevant side effects which will be described underneath.

Sampling with lower value and then rise it digitally will produce a processed lifeless sound.

A lot of confusions and misinformations has appeared with the event of the plugin analyzers. These tools are funny to use with plugins but if used with libraries they can show some good and some not so good things, in some case they are artifacts created by them self.

(I don't use these tools, they are amateur-funny but not reliable for professional use. They use single sweep/tone analysis, professional tools use calibrated multitone analysis and they cost 1000 times more).

Chirp... Chiorp, Chiurp !

Analyzing the library without the right gain staging it can show distortion and spike (someone called it "chirp" – but chirp is a sweep, as reported in the Audio Precision user manual) because the analyzer clips the library to 0dBfs while a library has the sweet spot at -18dBfs. So, no birds in the library. XD

At the right level, no audible artifact is perceived and in the worse case that spike lives in the -80dBfs region. very close at the noise floor of the majority of the hardware processors. So, don't care.

To go to fix something that is not present or evident, because I wouldn't like to see it on the screen, it damages and compromises the sound only by making the audio lifeless and flat. My test with the beta-team proves it.

Woobble !

Another side effect I saw at the beginning of the sampling era is the ripple, which someone call it wobble. Like the spike, ripple is a side product of the sampling process which lives in the bottom end of the frequency spectrum. With the right use of the library and the right gain staging and clean monitoring-room system it is almost inaudible or negligible. Reducing this artifact with programming tricks only it's not good because the sound become lifeless and flat.

Recently with the Golden Edition releases the new sampling-programming process allows to reduce the ripple at +/-0,1dB.

Curiously in over ten years nobody has complained about ripple and spike, before the event of these plugin analyzers, and a lot of records has been produced – as Grammy Awards, Cine Hollywood, Top Hits Records ... - .

But now yes: we hear all these artifacts with the eyes ! LOL

Another myth is the preamp. In the equalizer library.

Assuming that this hardware hasn't the preamp but a make-up gain depending by the circuit adopted, my equalizer libraries don't need that “preamp” preset because they has been developed to work and to sound in the same way as the sampled hardware. The interaction between the preset at 1K and 3k or 5K, as described in the program manual, it gives the exactly behavior and sound by filters and make-up stage. A different mode like filters + preamp has given poor and inferior results in terms of audio quality during beta-testing: lifeless, narrow and flat stereo image, unfocused details and less dynamics compared to my technology.

I have tried it in the 2009 and beta-tested it again and again during these years, so I still don't use filter + premp mode because the hardware is not built in this way and this system-preset doesn't sound good.

*At the end:

the demo audio and some demo programs are available to evaluate the quality and usefulness of my programs. If you don't like what you hear simply don't use/buy it.

Thanks

2.4 - Gain Staging

GUI's meters show the value in dBfs.

Take care with gain staging since the programs are close to the hardware, as reference 0dBVU on the hardware corresponds to -18dBFS on your DAW digital meter.

Normally the best sound is achieved with occasional maximum digital peaks to -10dBfs, i.e. kick or snare transients, pluck synth and other hits. On the mixbus the whole mix can hit an occasional maximum digital peaks between -8dBfs and -6dBfs. (imperative!)

When the signal is too high the sound will be congested and saturated/distorted in a bad way, too high peaks (and inter-sample peaks) overload Nebula which plays a BLIP as alert.

I recommend to mix with a good and precise VU Meter like this by Waves:

www.waves.com/plugins/vu-meter



It mimics the way our ears react to sound by giving you a more realistic representation of the way audio level changes are actually perceived.

In this way you can easily check the levels on every single track and for the whole mix by inserting the VUMeter as last instance on the mixbus and by setting the 0dBVU = -18dBfs on it (Headroom).

I suggest to deactivate or remove the VU Plugin when you export the mix to avoid any coloration. Yes, some plugins color the sound even if they are analyzers and/or bypassed.

NOTE: a console, limiter, equalizer, tape machine or compressor is not a guitar amp! If you drop the level back to where it would be using the real hardware, libraries can sound huge.

Useful video about to use the VU Meter and Gain Staging:

www.youtube.com/watch?v=2DVz_T48M-Q

www.youtube.com/watch?v=ECRx4WF3pcc

Great book about audio recording engineering:

<https://bobbyowsinski.com/recording-engineers-handbook/>

Another great book about music production with my contribution about console:

www.routledge.com/Producing-Music-1st-Edition/Hepworth-Sawyer-Hodgson-Marrington/p/book/9780415789226

2.5 - Common Controls

All programs have some common controls which are detailed below.

Input Gain

The Input Gain control sets the level at the input of the plugin.

The range is from $-\infty$ dB to +6 dB.

Output Gain

The Output Gain control sets the level at the output of the plugin.

The range is from $-\infty$ dB to +6 dB.

Bypass

This switch control sets the plugin operative or bypassed

Meters

Input and Output Meters display the levels at the input and output of the plugin in dBfs.

Compressors and Expander/Gate have a gain reduction meter also.

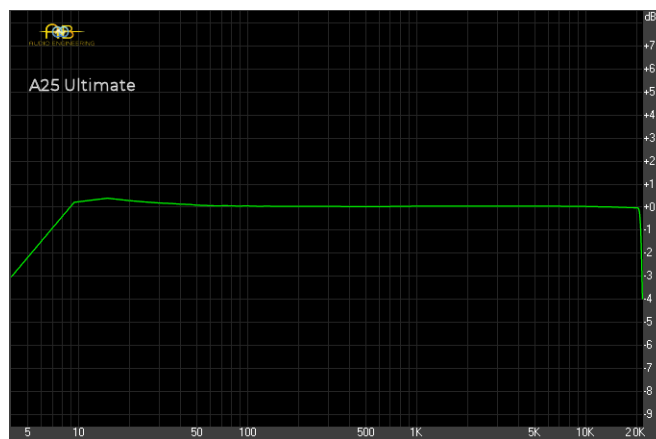
NOTE: *clicking on the controls while pressing "ctrl" on computer keyboard, the control returns to zero.*

3. American 25 Bus Compressor - Ultimate

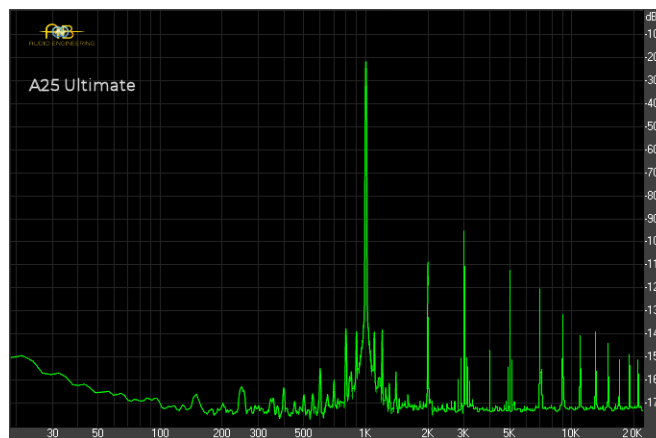
3.1 - About the original hardware

The American 25 Bus Compressor 50th Anniversary Edition is the American factory's dynamics flagship. Carefully designed to deliver American's signature analog sound, the A25 Ultimate delivers warmth, clarity and punch, whether used for subtle adjustments in mastering or heavy compression effects in mixing.

With its broad bandwidth the American 25 Bus Compressor has been engineered to deliver recordings at the best conceivable quality onto any format at any sample rate, maintaining all the energy, atmosphere and life of the original performance as perfectly as possible.



A25 frequency response (@ 0dBVU)



A25 THD (@ 0dBVU -10dB Gain Reduction)

3.2 - Session Setup

American 25 reproduces the characteristic sound of American Bus Compressor, this kind of compressor has been used in top records in the world. To faithfully reproduce this analog sound in the DAW, we recommend using the A25 in the track or bus-group where you need to control dynamically the sound. With subtle adjustments it could be used on stereo bus or in mastering.



To emulate the original unit sound you should use the presets in this way: Insert the Comp-Side Chain preset followed by the Line Out preset. Insert the TRST preset on the track used as side-chain control. These presets interact together to give the same behavior, harmonic content and overall sound as the original sampled unit.

The original internal side-chain of the A25 has been carefully sampled and reproduced into the TRST preset. If you want a classic linear side-chain control you can use the normal compressor preset followed by the Line Out preset only.



Compressor loves to be pushed hot, input peak signal between -10dBfs to -2dBfs are good for that typical softness transient sound of this unit. Input levels above -10dBfs allows more glue and musicality. Anyway: use your ears and enjoy ! :)

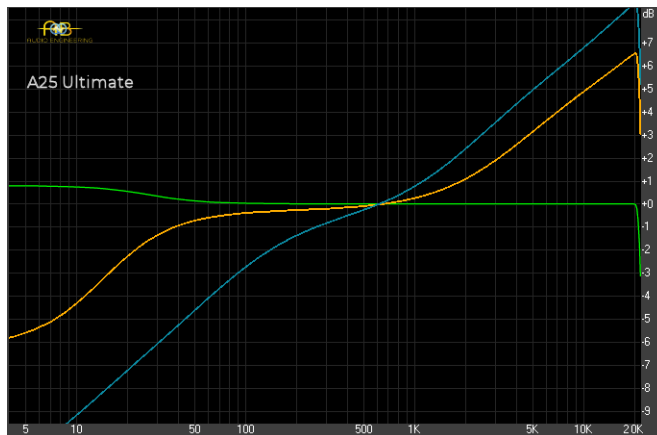
External SideChain

The Sc presets have to be used with the TRST preset to emulate the original hardware, however the Sc presets are useful to control the dynamic module of the Compressor via external side-chain for ducking, pumping or other effect.

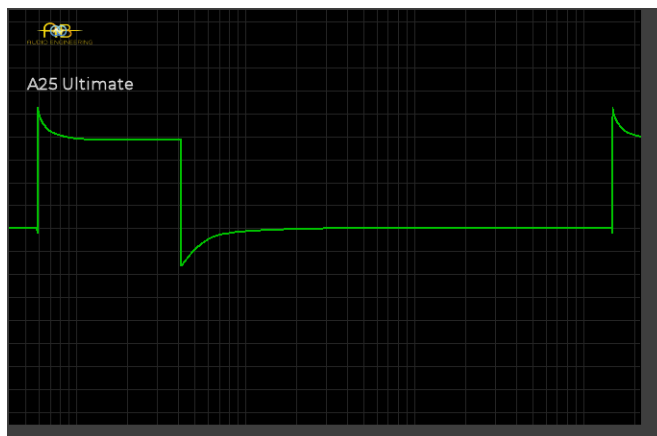
If you use several insert processing on the side-chain track, and you want to maintain the original behavior and sound of the A25 Bus Compressor, be sure to insert the TRST as the last insert processing.

To use side-chaining correctly please refer at the manual of your DAW.

This function works on Nebula4 only.



A25 TRST frequency response (@ 0dBVU)



Attack 10ms Release 100ms

NOTE: use the presets you need only. Stacking unnecessary presets is the wrong way to emulate the sampled hardware.

Remember: every digital processing introduces quantization errors/noise which degrade the sound, a smart use of the digital processing keeps the sound alive.

3.3 - Preset list:

The A25 library includes the following programs displayed into menu “A25” and subgrouped into 44.1kHz, 48kHz, 88.2, 96kHz.

A25 Comp New
A25 Comp New sc
A25 Comp Old
A25 Comp Old sc
A25 TRST
A25 Line Out

A25 Comp New

Feed Forward Compressor with attack, release, threshold, ratio, HPF, gain makeup and wet/dry controls.

A25 Comp New sc

The same as the Comp New preset but with external sidechain input and without HPF.

A25 Comp Old

Feed Back Compressor with attack, release, threshold, ratio, HPF, gain makeup and wet/dry controls.

A25 Comp Old sc

The same as the Comp Old preset but with external sidechain input and without HPF.

A25 TRST

Original side-chain circuit with Tone control.

A25 Line Out

Line Out stage with GDrive.

3.4 – Controls on Nebula GUI

Over the common and the above described controls, A25 has also:

Comp

- **ATT: Attack Control**

The “ATT” control defines the attack time of the Compressor.
The switchable values are 0.03, 0.1, 0.3, 1, 3, 10, 30ms.

- **REL: Release Control**

The “REL” control sets the amount of time it takes for processing to cease once the input signal drops below the threshold level.
The values are from 50ms to 3s continuous control.

- **THR: Threshold Control**

The “THR” control defines the input level at which limiting or compression begins. Signals that exceed this level are processed. Signals below the threshold are unaffected. The range is variable from 0 to -64.

- **RATIO: Ratio Control**

The “RATIO” defines the amount of gain reduction to be processed by the module. The switchable values are 1,5, 2, 3, 4, 6, 10, 12:1 (Limiter).

- **HPF: High Pass Filter Control**

The “HPF” control sets the cut-off point of a high-pass filter on the internal sidechain. The range is variable from 0Hz to 2000Hz.

- **MAKP: Gain Makeup Control**

The “MAKP” control sets the output level of gain makeup. The range is variable from 0dB to 25dB.

- **WET: Dry/Wet Control**

This control lets you balance between processed (Wet) and original signal (Dry) allowing you to do parallel compression without routing to buses/aux. The range is variable from 0% (Dry) to 100% (Wet).

TRST

- **TRST: external side-chain Control**

This control influences the tonal characteristics the compression applies to the audio by decreasing the reaction or the compressor to low frequency content and maintaining a uniformly compressed signal. The result is an increase of punch and low frequencies. Tone switch controls the TRST inverse filter: Normal, Medium, Loud.

Line Out

- **GDRV: GDrive Control**

The “GDRV” control is a unique feature not found in similar products from others brands that comes from Acustica Audio VVKT proprietary technology and sampling approach. It allows you to control the amount of harmonic distortion that is coming from the analog hardware. The “Input” control acts as the analog signal chain of the device, where reducing the volume also reduces the harmonic distortion in accordance. The “GDrive” function allows independent control of this harmonic content, so that the input level can be left alone while making adjustments to the harmonics. Reducing the harmonics leads to a cleaner signal with an already clean device. Increasing the harmonics should be done with moderation.

This type of effect is not truly representative of a real console, but it can be useful when you want more of the console’s nonlinear “vibe” without altering the channel’s levels. The available range is ± 12 dB.

Note that increasing the input signal the internal headroom will be reduced.

NOTE: clicking on the controls while pressing “ctrl” on computer keyboard, the control returns to zero.

END