AlexB A16GE+ Manual

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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 theplugins.

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: http://www.alexb.eu

Thank you !

AlexB... Audio Renaissance.

NOTICES

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

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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@alexb.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 – Golden Edition Plus

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

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

1.7 – Installation and file BACKUP

After downloading, unpack the files and **make a safe backup** of the library. I reccommend 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**.

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

SK libraries have special functions like “preset change” which allow you to switch between the programs by retaining the settings. The use requires Nebula Setup. You can follow the video instruction in the product web page.

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 to 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.alexb.eu/video/AlexB_SPQ.mp4

Settings for this library:
METERTYPE 2

NOTE: *MAC users find more info on the "MAC Nebula4 Setup Addendum" on the website:*
https://www.alexb.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...

<u>TIMED and Kernel Length</u>
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).

<u>Chirp... Chiorp, Chiurp !</u>
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.

Wooobble !

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 Hawards, 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:
https://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:
https://www.youtube.com/watch?v=2DVz_T48M-Q
https://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:
https://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 -∞ dB to +6 dB.

Output Gain
The Output Gain control sets the level at the output of the plugin.
The range is from -∞ 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 16 Console Golden Edition Plus

3.1 - About the original hardware

This American Modern Studio Console gives you the legendary reliability and incredible sound that have characterized American consoles for a half century. The A16GE+ deserves its spot in 2520 Op-Amp's and circuitry, the superior results are quite audible with optimized low-frequency reproduction and crisper imaging, which delivers that sought-after, massive American “punch-in-the-gut” sound that jumps right out of the speakers.

With its broad bandwidth the American 16 Console has been engineered to deliver recordings at the best conceivable quality onto any format at any sample rate, capturing all the energy and atmosphere of the original performance as perfectly as possible.

3.2 - Session Setup

American 16 Console reproduces the sound of Modern American Recording Console using a library programs consisting of channels input, group bus, mixbus and pan-fader. To faithfully reproduce into the DAW the analog console signal chain and workflow, we recommend using the American 16 Console in one of two following session setup configurations.

> As a virtual summing box : Input Channel is inserted on the last insert of the DAW audio tracks, like a direct out routed to a summing box. The MixBus is placed on the first insert of the master track, just as the stereo return would be routed from the analog console back to the DAW.
>
> To simulate a console : Input Channel is inserted on the first insert of the DAW audio tracks, the MixBus is placed on the last insert of the master track. If you group channels in your DAW, i.e. drums elements, you can insert the GroupBus as last insert in the submix group bus to achieve the classic bus coloration. Optionally you can insert the pan-fader at the end of each channel track.

You should set the Pan Law in the DAW at -3dB. You might like to use the Pan-Fader (included in the library) on some stereo tracks and group bus instead of the DAW's panner, the A16GE+
Pan-Fader should be the last insert into DAW's track or group bus leaving the Pan Law in the DAW to 0dB.
Obviously the Panner doesn't work on mono tracks but you can still use the fader which gives better sound and precision compared to the DAW's counterpart.

TRICK: *to emulate the non linearity between the channels of the console, you can set the GDRV control slightly different on every track into a range of +/-3dB.*

3.3 - Preset list:

The American 16 GE+ library includes the following programs displayed into menu “A16” and subgrouped into 44.1kHz, 48kHz, 88.2kHz and 96kHz.

A16GE+ Input
A16GE+ G.Bus
A16GE+ M.Bus
A16GE+ Panner

A16GE+ Input
The A16GE+ Input in is the first stage of the console, normally it works as line amplifier and you should insert it in each track of the mix. With the Input preset you can choose between DI (1), Line (2) and Mic-Preamp (3) by selecting them with the Source control.

A16GE+ G.Bus
If you send some tracks to a submix bus group in your DAW and you like to have the real sound by Bus Group of the console, you can use the G.Bus preset as insert into DAW's submix bus group.
G.Bus has new concept in the Golden Edition+: instead to have more presets with different colors, there is only ones with the “Type” control which allows to choose different colors:

1 : Clean = Original Clean G.Bus, the pure sound from the console
2 : Drum = Original Clean G.Bus with 88RS channel eQ patched
3 : Percussions = Original Clean G.Bus with vintage Pultec patched
4 : Acoustic = Original Clean G.Bus with GML8200 patched
5 : Guitars = Original Clean G.Bus with vintage API 550A patched
6 : Synthpad = Original Clean G.Bus with vintage Moog PEQ patched
7 : BGVocals = Original Clean G.Bus with Neumann W492 EQ patched
8 : Ambient = Original Clean G.Bus with vintage Filtek PB1 parched

A16GE+ M.Bus
M.Bus has new concept in the Golden Edition+: instead to have more presets with different colors, there is only ones with the “Type” control which allows to choose different colors:

1 : Vintage = The sound of the vintage console Mixbus
2 : Clean = Original clean console Mixbus
3 : Modern = Original Clean Mixbus with modern API 5500 eQ patched

M.Bus has been sampled at full headroom so you have to follow the gain staging (pag.27). By using the G.Drive you get from clean to very hot/saturated signal out.

A16GE+ Panner
Console stereo panner and fader with -3dB Pan Law.

3.4 - Controls

The American 16 Console has only a few but intuitive and effective controls which are detailed below.

SOURCE **Source Control**
The “SOURCE” control selects the input source of the first stage of the console. The options are: DI, Line and Mic-Preamp.

TYPE **TypeControl**
The “Type” control selects the optional type of G.Bus or M.Bus.
See description.

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 aproach.
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.

DRIVE **Drive Control**
The “DRIVE” control affects the harmonic contents in an unnatural way, but suitable if you look for an effect.
The available range is ±12 dB.

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

NOTE2: *do not adjust the ATTCK and RELS controls, leave them at stock value (center 12 o'clock).*

END