



## Alpha 3.3 Series User Manual

Alpha 3.3  
Beat Alpha 3.3  
CM Alpha 3.3  
FREE Alpha 3.3

## Welcome

The Alpha is a synthesizer classic. It was first released in 2000 under the name GakStoar Alpha by Peter Linsener, who in 2001 founded LinPlug. While the first version was free, with version 2, in 2002 the Alpha became a commercial product. It was on sale for about 5 years when it was replaced in 2007 by the Alpha 3 which is one of LinPlug's best selling products of all time.

All these years the fundamental design principle of the Alpha was simplicity. While the main market for synthesizers gave birth to more and more complex instruments the Alpha was kept as simple as possible. Only few features were added over time, like the Chorus in Version 2 or the Ring-Modulator and Noise Oscillator in Version 3.



Since Version 2 Branislav Pakić is designing the user interface of Alpha (and many other LinPlug instruments), giving it its distinctive and almost minimalistic look. While the user interface is often used as an eye catcher to generate sales, our priority has always been useability. It should assist you, and not distract you. For many more years.

Now the Alpha is 14 years old and still going strong, a big thank you is in order to all the people who bought a license and thus allowed us to continue this synthesizer. Our last update did not get the number 4, because it's still the same strong audio engine, but the user interface and the handling has made another step forward.

This manual describes all aspects of the Alpha Synthesizer and is designed so that your use of this software is as efficient and enjoyable as possible. At LinPlug we're very proud of the Alpha Synthesizer; it's the result of many years of research and synthesizer programming experience. We hope you get a lot of pleasure using the Alpha Synthesizer and that it becomes an integral part of your music-making. Thank you.

Your LinPlug Team,

Berlin, Germany, January 2007 and reworked January 2014

## Credits

Instrument by Peter Linsener and Pavol Markovič

Manual by Peter Linsener

Alpha Sounds by (alphabetic order)

Dubhad – dh sounds  
Frank Von Haeven  
Frank Neumann – Xnx sounds  
Guilherme Kalfelz aka WilliamK  
Ken Fennell  
Klaus-Dieter Pollak aka Summa – sum sounds  
Laurent Gaudin - ksn-sounds  
Nico Herz - BT sounds  
Peter Schelfhout - PS-sounds  
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## Installation

### Installation on Mac

After downloading the installer for Mac you will find a file named "Alpha Installer nnn.dmg" (or "Alpha Demo Installer nnn.dmg" if you are trying the demo) located in your download folder. "nnn" is the actual version number, for example "Alpha Installer 330" is version 3.3.0.

Most Mac will mount the disk image automatically. If not, please double-click this file to open the image.

Then double-click the installer program which will guide you through the installation process. The plugin file for Alpha and the factory sounds (presets) will be placed in the right directories for AU and VST plugins on your Mac.

The next time you start your DAW software Alpha will be listed in the AU and VST plugin list.

The installation is always only for the current user; if you want to use the plugin with different users on your computer, each user must run their own installation.

### Installation on PC

After downloading the installer for PC you will find a file named "Alpha Installer nnn.exe" (or "Alpha Demo Installer nnn.exe" if you are trying the demo) located in your download folder. "nnn" is the actual version number, for example "Alpha Installer 330" is version 3.3.0.

Start this program and the installer will guide you through the installation process. Make sure you choose the correct VST plug-in directory so that your DAW software can find Alpha. Please check your DAW software's manual if you are unsure about where its VST 2 plug-in directory is located.

The plugin file Alpha.dll (if you choose the 32 bit version) or Alpha 64.dll (for the 64 bit version) and the factory sounds (presets) will be placed in the chosen directory. The next time you start your DAW software Alpha will be listed in the VST plugin list.

The installation is always only for the current user; if you want to use the plugin with different users on your computer, each user must run their own installation (however, you can choose the same installation location).

<continued on next page>

## Common to Mac and PC

The following does **not** apply to FreeAlpha, AlphaBeat and AlphaCM! The first time Alpha is started you should go to the Options page (the button to open the Options is in the lower right of the User Interface).



Copy your personal Serial number from the mail you got after your purchase and paste it into the field labeled "Serial Number" which reads "Enter Serial here". Some DAW may not allow copy/paste operations to a plugin, then you have to enter it manually. In case you enter it manually, make sure to not introduce any typos, capital letters needs to be entered as capital and small letters small.



If the serial number has not been entered or it has been entered incorrectly, Alpha will remain in demo mode and emit a noise sound about once a minute.

**Please keep your serial in a safe place as it's your primary proof of ownership and is required for support, updates and of course to reinstall Alpha.**

If you have any questions regarding the installation please contact us at <http://www.linplug.com/support/support.htm>.

## Overview

There are multiple versions of the Alpha synthesizer available: the full Alpha, a couple of custom Alpha synthesizers with a somewhat reduced set of features and the Free-Alpha which is publicly available but has the least features.

See the Appendix F on page 32 for the differences between the various versions of Alpha. This manual describes the Alpha so some parts may not be applicable to the custom versions of Alpha and the Free-Alpha.

The Alpha synthesizer is a 32 note-polyphonic AU and VST synthesizer based upon a classic subtractive design. It includes three oscillator modules, a filter module, two envelopes and three LFO modules, a chorus effect and an easy-to-use yet powerful Modulation Matrix.

The structure of the main edit page can be divided into several sections: Oscillators, Filter, Amplifier, Modulation Matrix, LFOs, Glide, Chorus and Master.

Audio signals are generated by the oscillators when the Alpha receives a Note-On from your DAW software. The Alpha contains three oscillators from which one generates noise only. The other two oscillators create a fixed mix of two waveforms each. Pitch, amplitude and symmetry can be modulated using the Modulation Matrix (Noise can be only modulated in amplitude). Moreover the second Oscillator has a Ring Modulation unit integrated.

The output of all three oscillators is mixed and routed to the Filter section. The filter modifies the harmonic spectrum of the mix and has its own dedicated envelope for controlling the filter cutoff parameter. The Filter also can be cross-modulated from one of the three Oscillators (Filter FM).

The output of the Filter section is then sent to Amplifier section, where it is shaped in volume and finally sent to the Chorus section. The Chorus feeds the Alpha output which sends the audio to your DAW's channel mixer. In your DAW you can set the pan position of the Alpha's output and apply additional effects.

At various points throughout the signal path you can modulate the signal using either envelopes, LFOs, MIDI controllers or other sources. The Alpha contains 2 independent envelopes and three LFOs that can be routed to any available modulation destination.

## User Interface / Controls

On Mac the interface supports both traditional and Retina displays automatically. You have the possibility to change the User Interface size with the menu on the very top left. Available resolutions are M=100%, L=150% on Mac and PC and additionally XL=200% on PC. The selected size is stored permanently and used whenever you open up a new instance of Alpha.

The controls (dials) on the user interface are operated in a linear manner, so move your mouse up/down to modify a control.

Holding down the ALT key while clicking on a control changes the selected control's value a minimum step upwards (when clicking in the upper half of the control) or a minimum step downwards (when clicking in the lower half of the control).

Holding down the CTRL key (Windows) or COMMAND key (Mac) while clicking on a control sets the control to its default value (e.g. for Volume controls it sets the control's value to 0 dB).

Holding down the SHIFT key while changing a control's value enables finer control values to be set.

Double clicking one on a Modulation Matrix Depth parameter sets the value to 0.00. Double clicking in the same parameter a second time resets it to its previous value. This is handy for quickly disabling (and re-enabling) one or multiple modulations.

All Controls can be controlled using external MIDI messages. To do this you need to use the ECS (MIDI-Learn) which is described in detail later in this manual.



## Oscillators

The Alpha's two main oscillators are located on the left of the front panel. Each of the Alpha's two oscillator modules outputs a waveform that is a combination of two basic waveforms. So the amount of possible waveforms it almost endless.



Each waveform shape is set by clicking on the waveform name in the Alpha's OSC 1 or OSC 2 sections. This opens a menu that allows you to select from the available waveforms. See Appendix B for a complete list of all waveforms.

The oscillators pitch is set to the right of the waveform within the range of -2 to +2 octaves by dragging the value up or down in one octave steps.

The “F” switch on the top allows you to switch between free-run (when switched on) and re-triggered start of the oscillator (when switched off). The main difference is that when free-run is Off, the Oscillator always starts at the beginning of the waveform cycle, and when its On the Oscillator continuously runs and when the note starts it can be at any point in its waveform cycle.

Below these controls is the “A wave B” balance dial. This dial allows you to gradually adjust the resulting waveform from the two basic waveforms. When the dial is at its leftmost setting, the waveform consists purely of waveform A. The more you move the dial clockwise, the more the waveform A is faded out and waveform B is faded in. When it is moved all the way to the right the waveform consists purely of waveform B. To try this set wave A to Sine and wave B to Sawtooth. As you slowly rotate the dial from one end of its range to the other you can hear the sound change from sine to sawtooth.

The detune dial in OSC 1 section determines the amount of detuning between the oscillator modules Osc 1 and Osc 2. A small amount of detuning can be used to create the swirling sound typical of many analog synth patches.

The “ringmod” parameter in Osc 2 section allows a gradually adjustment of how the output of oscillator 2 is made from oscillator 2 and the integrated Ring-Modulator (see the glossary on page 25 for an explanation of AM and Ring-Modulation).

Turned fully left the output of oscillator 2 is purely oscillator 2. Turned fully right the output of

oscillator 2 is fully the ring modulation result of oscillator 1 and oscillator 2. In middle position the output is mix of oscillator 2 and the ring-modulation result of oscillator 1 and oscillator 2 (this is what you find in some synthesizers as AM = Amplitude Modulation).

## Mixer

The upper dial in the Mix section is used to set the relative output volume of each oscillator module. When the dial is set to 1 (i.e. it is turned all the way counter-clockwise) the output signal consists entirely of oscillator 1. When the dial is set to 2 (i.e. it is turned all the way clockwise) the output signal consists entirely of oscillator 2. The mid position provides an equal mixture of oscillator 1 and oscillator 2.

The lower dial is the “noise” parameter that allows to mix gradually some noise to the two oscillators mix. Turned fully left no noise is added, turned fully right the oscillators are suppressed and only noise is audible. The generated noise is white noise, pink noise or brown noise and can be set by selecting one of the three color dots below the dial.

## Filter

The Alpha's filter module is located right of the oscillators. This also reflect the signal flow, which is from the Oscillators into the Filter.



The “drive” (or saturation) dial on the left can be used to overdrive the Alpha's filter. This can be used to produce a distorted sound which depends on the amount of drive applied and on the basic waveform. It also depends on the Level of the signal that comes from the oscillators: so if you for example amplitude modulate an Oscillator with an LFO via the Matrix, this will not only change the amplitude but also the distortion (if applied).

Below the drive there is the filter “fm” dial and filter fm source selector. With the dial the amount of filter cutoff modulation is adjusted from no modulation when turned fully left to maximum modulation. As the modulation source is in audible range we have a true Cutoff Frequency Modulation (Cutoff FM).

With the fm source selector the source of the modulation is selected, either Oscillator 1 , Oscillator 2 or Noise.

Note: Cutoff FM does **not** depend on the setting in the Mix section of the oscillators.

Moreover, if oscillator 2 is selected then its purely oscillator 2 and does not depend on the ring modulation setting in oscillator 2 section (the Ring Modulator is not available as source for filter FM).

The “cutoff” dial is used to set the operation frequency (in Hz) at which the filter begins to take effect. In the case of the Low Pass Filter (Settings 12, 24 and 24+), higher settings produce brighter sounds while lower settings result in darker sounds. In the case of the High Pass filter, higher settings produce thinner sounds while lower settings result in fuller sounds. The Band Pass filter combines a Low Pass and a High Pass filter. In this case, the Cutoff dial sets the midpoint of the filter’s pass band.

The “reso” (Resonance) dial is used to set the amount of emphasis around the cutoff frequency. Higher settings create a more pronounced peak in the signal while lower settings produce a flatter response.

Below cutoff and resonance is the filter type selector. It allows selection of 5 types of filter. These are:

- 12 Low Pass 12 dB/Oct.
- 24 Low Pass 24 dB/Oct.
- 24+ Low Pass 24 dB/Oct., alternative design
- HP High Pass 12 dB/Oct.
- BP Band Pass 12 dB/Oct.

The envelope parameters follow below the type selector, these are the classic attack, decay, sustain and release as well as an additional “fade” parameter. Note that the envelope has only an effect if the filter envelope depth (described below) is not 0. The following description of the envelope parameter assumes that the depth is positive.

The “att” (Attack Time) dial is used to set the time it takes for the cutoff envelope to reach the full envelope depth. For example, if the Attack is set to 0% (i.e. the dial is turned completely counter-clockwise), the filters cutoff will move from cutoff parameters value to maximum envelope depth immediately. If the Attack dial is turned completely clockwise the filters cutoff will move from cutoff parameters value to maximum envelope depth in 10 seconds.

The “dec” (Decay Time) dial determines the time that the filter cutoff envelope takes to move from the maximum envelope depth to the Sustain value. For example, if the Decay is set to 0% (i.e. the dial is turned completely counter-clockwise), the filters cutoff will move from the Attack peak value to the Sustain value immediately. If the Decay dial is turned completely clockwise the same change in cutoff frequency will take 10 seconds.

The “sust” (Sustain) dial determines the cutoff frequency after the initial Attack/Decay phase while a note is being held.

The “rel” (Release Time) dial is used to set the time that the cutoff frequency takes to move from current Sustain/Fade value to silence after the note is released. If the Release dial is set to 0% (i.e. the dial is turned completely counter-clockwise), the filters cutoff will move from the current level to the cutoff parameters value immediately. If the Release dial is set to

100% (i.e. the dial is turned completely clockwise) it takes full 10 seconds.

The “fade” (Fade Time) dial is used to set the rate at which the filters cutoff frequency moves from the Sustain value to either the cutoff parameters value (when the dial is turned counter-clockwise) or maximum envelope depth (when the dial is turned clockwise). A middle Fade setting (dial is set to 12 o'clock position) means that the filters cutoff frequency remains at the Sustain level until the note is released.

The “depth” dial is used to set the degree to which the filter's envelope effects the signal. Setting “depth” to -100% (i.e. turning the dial completely counter-clockwise) means that the envelope has full negative effect on the filter. Adjusting the dial to the middle position means that it has no effect on the filter. Setting “depth” to 100% (i.e. turning the dial completely clockwise) means that the filter is modulated by the envelope's full range. The negative envelope depth can be used for opening the Filter when a note is released (which would be impossible with a non-inverted ADSR envelope).

### **Filter Keyboard Tracking**

Alpha has no dedicated control in the filter section to adjust filter keyboard tracking. You can however, achieve this using the Matrix:

Note 0.4000 Filter Cutoff

gives a pretty exact tracking, so that the filter opens just like the notes go up. For example from two notes played one octave apart the filter on the higher note will be one octave more open (Cutoff is twice the frequency of the lower note).

The tracking is pretty good for the usually keyboard range of 4 or 5 octaves, and while it actually covers 10 octaves, its up to 4 semitones inaccurate at the outmost ends of the spectrum.

On some old hardware you also found half-tracking which would translate in a matrix depth of 0.2 while no tracking obviously requires no entry in the matrix. Moreover negative tracking is possible this way as well, just use a negative modulation depth here (the Filter than closes more and more the higher the played notes are).

## Amplifier

The amplifier sections follows right of the filter section. Again this reflects the signal flow, after the filter the amplitude of the sound is shaped.

It features a “vol” volume control to adjust the overall output volume. The “vel” velocity control allows adjustment of the influence that the notes velocity has on volume. When turned fully to the left the Velocity has no influence on the volume while turned fully right means that the volume is completely controlled by velocity.



The “att” (Attack Time) dial is used to set the time it takes for the amplitude envelope to reach the full envelope depth. For example, if the Attack is set to 0% (i.e. the dial is turned completely counter-clockwise), the signal's amplitude will move from zero to full volume immediately. If the Attack dial is turned completely clockwise the signal's amplitude will move from zero to full volume in the maximum time of 10 seconds.

The “dec” (Decay Time) dial setting determines the time that the amplitude envelope takes to move from the Attack peak level to the Sustain level. For example, if the Decay is set to 0% (i.e. the dial is turned completely counter-clockwise), the signal's amplitude will move from the Attack peak level to the Sustain level immediately. If the Decay dial is is turned completely clockwise, the signal's amplitude will move from the Attack peak level to the Sustain level in 10 seconds.

The “sust” (Sustain Level) dial setting determines the amplitude level after the initial Attack/Decay phase while a note is being held.

The “rel” (Release Time) dial is used to set the time that the amplitude envelope takes to move from current Sustain/Fade level to silence after the note is released. If the Release is set to 0% (i.e. the dial is turned completely counter-clockwise), the signal's amplitude will move to zero immediately. If the Release dial is is turned completely clockwise, the signal's amplitude will move to zero in 10 seconds.

The “fade” (Fade Time) dial is used to set the rate at which the signal amplitude moves from the Sustain level to either silence (when the dial is turned counter-clockwise) or full output (when the dial is turned clockwise). A middle setting (dial is set to 12 o'clock position) means

that the signal amplitude remains at the Sustain level until the note is released. The more the dial is moved from the center position the faster the level changes.

The “spread” dial allows you to play several voices simultaneously, which is sometimes called Unison. If the Spread is set to 0% (i.e. the dial is turned completely counter-clockwise), Spread is disabled. If the Spread is set to anything other than 0%, the Alpha plays 8 oscillators simultaneously, 5 times Osc 1 and 3 times Osc 2, all detuned with one another. Higher settings create more detuning, so producing a fatter, thicker sound. A small indicator light shows when Spread is enabled so you not overlook a very low setting. Opposed to classic Unison the Alpha remains fully polyphonic when using Spread.

## Modulation Matrix

The Alpha's Modulation Matrix allows you to create 11 user-defined modulation routings. See Appendix D for a listing of all modulation sources and destinations.



Modulation sources are shown in a column on the left of the display, while the destinations are shown on the right. The modulation amount is displayed in the middle. To change a routing click on the source or destination that you want to change. A menu will appear which lets you select the new source or destination. To remove a modulation source or destination select the " - - " entry in the menu.

To change the modulation depth click on the amount display and move the mouse (while keeping the mouse button pressed) upwards or downwards (increasing or decreasing the value) until the desired amount has been reached. A negative modulation depth inverts the shape of the modulation source.

A double click on the modulation depth sets it to 0, another double click will revert it to its original value, so you can quickly disable certain modulations to check their effect.

The modulation of pitch has a special display for modulation depth. In example a modulation depth of "2:40" means that the pitch is modulated to a depth of 2 semitones and 40 cents.



## LFO

An LFO (Low Frequency Oscillator) is an oscillator that generates a low frequency signal that can be used to modulate certain parameters of the Alpha synthesizer. The Alpha contains three LFOs on the right of the front panel.



First, the LFO can be set to either “M” Mono or “P” Poly mode with the switch below the LFO number. In Mono mode, one LFO is shared by all voices, it is constantly running and not re-triggered. In Poly Mode, each voice has its own LFO which is triggered when the note starts.

The waveform parameter is used to select one of the LFO waveforms. To select a waveform, click in the shown shape. It will go through the six LFO waveforms with each click. The available LFO waveforms are: Sine, Triangle, Sawtooth, Square, Noise and SamHo (Sample and Hold).

The “att” (Attack Time) dial is used to set the time it takes for the LFO to reach the full modulation depth. This can be used to slowly increase the amount of modulation applied. If the Attack is set to 0% (i.e. the dial is turned completely counter-clockwise), modulation commences immediately. If the Attack is turned completely clockwise it takes 10 seconds for the modulation to reach its full intensity.

The “freq” (Frequency) dial is used to set the the LFO's frequency. If the Frequency dial is turned completely counter-clockwise, the LFO oscillates at 0.01 Hz (that is 1 complete cycle every 100 seconds). If the Frequency dial is turned completely clockwise, the LFO oscillates at 32 Hz (32 cycles per second).

The frequency is shown in a display above the dial, which, when clicked, turns into a sync popup menu. This menu allows you to set the LFO so that it is "synced" to the tempo of the current song (see Appendix C for the range of possible sync settings) For example, a setting of 1/4 will make the LFO cycle last exactly one quarter note.

Choosing “Off” from the menu causes the LFO to be not synchronized to the song tempo, instead the LFO speed can be adjusted with the “freq” dial below the display.

## Glide

"Glide" continuously changes the pitch from one note to the next, connecting the notes and letting you smoothly "glide" from one to the other.



The **Glide Mode button** has 4 values: Off, On, Held and Bend. The On and Off functions turn "Glide" on and off. When "Glide" is on, the Time/Rate dial is used to set the time it takes for the pitch of one note to reach that of a following note. The "Held" setting works as follows. If notes overlap then Glide is applied, however if they don't overlap, then the notes are played without Glide. This makes it possible to apply Glide only to selected notes. "Bend" allows you to apply a predetermined pitch bend at the start of each note. When using "Bend", the actual bend range is set using the **Bend control** within the range of -48 to +48 semitones.

The **Time/Rate dial** is used to adjust the time it takes for the glide to happen. Turned fully counter clockwise the glide is so fast that its hardly noticeable. If the dial is turned fully clockwise it takes very long for the notes to glide from the previous notes pitch to their own pitch.

The **Time/Rate button** has two settings: Time and Rate. These settings determine the manner in which the pitch of one note moves to that of the a following note. When set to "Time", it takes a constant amount of time to move from one note to the next, no matter what notes follow each other. In this case it will take the same amount of time to reach the destination pitch regardless of whether the preceding note was a semitone away or an octave away. When set to "Rate", the pitch of one note moves to that of a following note at a constant rate. This means that the amount of time it takes to move from one note to the next depends upon how far apart the pitches of the two notes are. The further apart the notes, the longer it will take for the pitch of the first note to reach that of the following note.  
Note: The Time/Rate switch has no effect in "Bend" mode.



## Chorus

The Chorus effect can be used to "thicken" a single sound creating the impression that it contains multiple voices. The Chorus works by mixing delayed signals with the original signal.



The "wet" dial allows you to set the balance between the processed (wet) signal and the original unprocessed (dry) signal. A small indicator light shows when Chorus is enabled.

The "time" dial is used for setting the chorus' delay time. Longer times produce a "chorusing" effect while shorter times create a rather "flanging" effect.

The "rate" dial sets the rate at which the signal is modulated.

To set the chorus' sound we suggest you start with all controls at their default position (COMMAND-Click on Mac or CTRL-Click on PC to set them to default) and successively change them until you find a sound that you like.

## Master

The Master section at the bottom of the main edit page contains the Sound Display and next to it are some more functions and options described in this chapter.



The Sound Display consists of two areas. The left one shows the current Bank (Category) and the right one shows the current Sound. Whenever a Sound is loaded the whole display is updated with the name of the loaded Sound and the name of the Bank it comes from.

In this example, the sound is named “the fall” from the “Pads 4” bank. By default the Sound Browser points to the factory sounds installed with the Alpha. The location of the library can be adjusted on the options page (adjusting the Startup Sound).

Clicking on either the bank or sound name opens the Bank Browser or Sound Browser, which look like this:



Its title either reads Select Bank or Select Sound. If you are in Bank Browser, you can

- choose a bank from the list, Alpha will automatically advance to the sound browser
- click the cancel button on the lower right to close without selecting a bank
- click the bank name from the lower center to close without selecting a bank
- click the sound name from the lower center to not change the bank but advance to the sound browser

If you are in Sound Browser, you can

- click on a sound name, Alpha will load the sound but keep the browser open
- double-click on a sound, Alpha will load the sound and close the browser
- click the cancel button on the lower right to close and return to the sound you had before opening the browser window
- click the ok button on the lower right to close and keep the currently loaded sound
- click the bank name from the lower center to open the Bank Browser
- click the sound name from the lower center to close and keep the currently loaded sound

While this might seem complex at first glance, when you get used to it its a pretty natural and consistent way of navigating through the sound selection process once one get used to it

### **About Sounds and Banks on your HD or SSD**

The Alpha stores all its Sounds (Presets) on your computers drive. On PC thats by default next to the plugin (in the VST plugins folder you chose during installation), on Mac thats in the /Library/ApplicationSupport/LinPlug folder.

In the “Alpha 3 Banks” folder there are all sub-folders with all the Banks which ship with the Alpha. In each of these subfolders are all Sounds (Presets) of this Bank. These files do have the extension “.fxp”.

So the whole factory library is organized in Categories and can be easily extended by you. Whenever you are going to save a sound you are directed to the “My Sounds” folder. This is because most people do only few sounds themselves and so they have them all in one place.

You can however create many more banks by creating more folders in the Alpha 3 Banks folder right from your operating system (Explorer in Windows and Finder on Mac). Alpha can display up to 85 banks (and 85 presets in each bank).

Also, when you buy or download free sounds you may want to place them in the “Alpha 3 Banks” folder so that they are new subfolders in there.

The currently loaded Sound can be changed in four different ways.

- You can select the desired sound directly from the right display, which turns into a full screen file menu when you click on it (as described above).
- You can switch to the previous or next sound in the bank by clicking the < (Previous) or > (Next) arrow buttons located left and right of the sound name.
- You can also change a sound by sending a program change command from your DAW to Alpha (using the built in sound browser of the DAW, if available)
- Sending a MIDI program change from your MIDI controller (if permitted by the DAW)

The Bank can be changed by clicking the bank name and choosing the desired bank from the appearing Bank Browser or by sending a MIDI Bank Select message (if permitted by the DAW).

Note that when you load a sound the currently present sound will vanish. So if you want to keep changes in a sound, make sure you save it before loading another one. The Save button allows you to save the current settings as a new Sound.

## **ECS / MIDI Learn**

The ECS (Easy Controller Setup) section makes it simple to control the Alpha 3 from an external MIDI controller. The only precondition is, that your DAW allows the Alpha to receive MIDI from your controller. While some DAW don't do this by default, most will and if Alpha can receive MIDI Control-Change Messages you can use ECS easily.

All you have to do is switch on the ECS (a menu pops up when you click at the ECS letters, choose Learn), select a Alpha 3 parameter with the mouse and then send some MIDI Control Change Messages from your controller to the Alpha. That's all there is to it! From now on you can change this parameter with that controller.

In addition to this, more than one controller can be defined to change a particular parameter. In fact, you can define up to 128 parameter-controller-combinations. This does not depend on the type of controller you have nor the particular MIDI Control Change messages it sends.

Don't forget to switch off the ECS (pick Off from the menu) after you have finished using it (the ECS button remains red during ECS activity as a reminder)!

ECS settings can be saved and restored using the "Load" and "Save" functions from the menu. In addition, a single controller assignment can be cleared using the "Clear" menu entry. All you have to do is select Clear and select a controls that you wish to be cleared (de-assigned from MIDI CC's). Don't forget to switch off the ECS control after you have finished clearing assignments!

The "Clear All" function clears all controller assignments at once and the Rest.Fact. Function means Restore Factory, thus restoring the factory settings as described in Appendix G on page 33.

## **Sound Generation**

The Gen (Sound generation) button is used to generate random patch settings. To generate a new patch click on the button and some or all of the current parameter settings will be changed to new settings. The Gen range is set on the Alpha's options panel described in the next chapter.

## **Undo / Redo**

The next two buttons allow to undo (and redo) parameter changes. Sometimes when editing a sound you may accidentally change a parameter or not be happy with the sound you created. If you only changed a few parameters it might be easiest to undo these recent changes. You can do so by clicking the Undo button once or multiple times.

In case you went back to far, having clicked Undo once too often you can simply redo (that is kind of undo the undo) the last change. Note that you can redo as many times as you did undo before.

Note: complex operations like loading a new sound cannot be undone.

## Options Page

The options page of the Alpha is accessed by clicking on the Options switch at the bottom right of the Alpha's front panel.



On the left of the Options page you find the information about which version of Alpha you are running as well as the Serial Number input field. Once the full version of the instrument is unlocked the serial number edit box is no longer editable but shows "Valid" as in the example below.

**Please keep your serial in a safe place as it's your primary proof of ownership and is required for support, updates and of course to reinstall Alpha.**



The Options are divided into three kinds:

- Per Preset options: setting which are stored within each sound. Per presets options control how Alpha works when loading this particular sound. They are too saved with your song or project in your DAW.

- **Per Instance options:** These are valid for each instance of Alpha, that is for every instrument you open, and only for this one. They are not affected when you load another sound. They are used to make one particular instance of Alpha work in a desired way, of course this is saved with your song or project in your DAW.

- **Global options:** Global options are valid for all Alpha Instruments in every song/project, they store automatically and work in each and every instance of Alpha that will be opened on your computer. So neither loading of a sound or a song/project will affect them.

## Per Preset Options

The **Voices** setting allows to limit the polyphony of Alpha. When set to “mono” Alpha works like a monophonic synthesizer, that is, each note terminates the one being played before and only one note sounds at a time.

The **Precision** control is used to set the accuracy of the Alpha's signal generation. When the Precision control is set to less than 100% small inaccuracies are introduced at various point in the Alpha's signal chain. This is useful if you're trying to replicate the warmth of an old analog synthesizer. Precision can be set in a range from 90% to 100%.

The **Bend Up** Range control is used to set the Alpha's response to pitch bend messages. Bend Range can be set from 0 to 24 semitones. A setting of 0 means that the Pitch Wheel won't affect the sounds pitch.

The **Bend Down** Range control is used to set the Alpha's response to pitch bend messages. Bend Down Range can be set from 0 to 24 semitones.

**Alpha 2 Osc** is a switch that allows the Alpha to work with an older version of the oscillators which had a bug: the waveform was not exactly symmetric, this introduced some harmonics which are characteristic for these sounds. This switch is to maintain compatibility with older sounds. The effect of this switch however much depends on the other oscillator settings and can even be inaudible. Generally when you are after a broader, fatter sound, switch it On and keep it off for a clear, clean sound.

## Per Instance Option

**Microtonal Scale:** Here you can load microtonal scalings in TUN-format. See Appendix B on page 30 for a complete description of how to use TUN files in Alpha.

**Master Tune** is used to set the overall tuning of Alpha if no micro-tuning file is loaded . Tuning can be set from 415.3 Hz to 466.2 Hz.



## Global Options

The **Gen Range** control is used to set the range for random parameter generation. It has a range of 0% to 100%. Using lower values will effect the patch less than using higher values. Not all parameters are changed with each generated change. With lower settings (e.g. 5%) less parameters are affected. Normally, a setting in the range of 2 to 10% will produce the most interesting and useable results.

Next are two options which allow you too choose a different Controller to function as **Modulation Wheel** or **Aftertouch**. This is useful when you not play with a keyboard or when your keyboard has not Aftertouch (also called Pressure) function but you have for example a Breath Controller connected; in this case just choose “Breath Controller” for the Aftertouch function and you can play Alpha using the Breath Controller to control all sounds which are programmed using Aftertouch.

Next is the **Startup Preset** and **Library Location** setting. When you install Alpha the Library (thats the sounds) is placed in a default location. On PC that is in the same folder where you installed Alpha, on Mac it is “/Library/Application Support/LinPlug”. However. If you for some reason want to place the library in a different location you can do so and point Alpha to the library manually by choosing a sound from the new location:

Click on the Set button next to the Startup preset name and navigate to a sound that Alpha should load upon startup. This not only defines your initial patch that is loaded when you open a new instance of Alpha but also defines at which location Alpha should look for the rest of the factory library. The path to the factory library is shown below the startup preset and you will see how it adjusts to the startup preset when you change it.

Finally there are a couple of switches:

The **Allow MIDI Program Change** option can be used to suppress Alpha to respond to program change messages received as MIDI data from your DAW software. The default setting is On.

The **Allow Host Program Change** is similar, but allows or suppresses the DAW software to change a program (sound) in Alpha. The default is again On.

The **Allow MIDI Volume Change** option can be used to suppress a response to volume change messages (CC 7) received as MIDI data from your DAW software. The default setting is On as its used by a couple of DAW's.

**To switch back to the main edit page click the Ok button in the bottom right.**



## Glossary

**AM:** AM or "Amplitude Modulation" is a process where the amplitude of one generator (the carrier) is controlled by another (the modulator). When the frequency of the modulator is periodic and below the audio range (less than 20 Hz) it is called "tremolo". When the modulation frequency is within the audio range, Ring Modulation is produced. See also Ring Modulation below.

**Amplifier:** A signal processing device that changes the amplitude, and hence the volume, of a signal.

**DAW:** Digital Audio Workstation, that's the Term for (in example) a Computer running some software to produce music or process audio in a wider sense. Usually its used for just the software itself, so Cubase, Logic and other software is considered a DAW.

**Envelope:** A time-varying signal used to control the development of another signal after it has been triggered. Envelopes are most often used for controlling a signal's amplitude. The shape of the envelope is determined by the number of control parameters. Often four parameters are available: Attack Time, Decay Time, Sustain Level and Release Time.

**Filter:** A signal processing device that suppresses or "filters" out specific parts of a signal's frequency spectrum. Numerous types of filter are used in audio synthesis. These include Low Pass, High Pass, Band Pass and Notch.

**LFO:** An LFO or "Low Frequency Oscillator" is a periodic signal source (usually below audio frequency range) used to modulate another signal parameter. An LFO can be used for a variety of effects including vibrato (by modulating the pitch) and tremolo (by modulating the volume).

**Modulation Matrix:** A signal "junction" where a source signal can be patched so that it controls a destination signal. The Alpha's Modulation Matrix is used for tasks such as modulating an oscillator's amplitude by an LFO.

**Oscillator:** A signal source that generates a periodic waveform at a given frequency.

**Ring Modulation:** The process of combining two audio signals by multiplication. Ring Modulation produces sidebands but suppresses both the carrier and modulating frequencies. Though Ring Modulation is only a special form of AM, in practice "AM" is considered Ring Modulation plus the carrier (one of the oscillators, in Alpha its the oscillator 2) and "Ring Modulation" is considered pure Ring Modulation with no carrier.

## Appendix A: MIDI Implementation Chart

Product: LinPlug Alpha Synthesizer Version 3.x Date: 15.September 2006

Function	Transmitted	Recognized	Remarks
Basic Channel			
Default	no	no	
Changed	no	no	
Mode			
Default	no	<b>Omni</b>	
Changed	no	no	
Note Number	no	<b>yes</b>	
True Voice	no	no	
Velocity			
Note On	no	<b>yes</b>	
Note Off	no	no	
Aftertouch			
Poly (Key)	no	<b>yes</b>	
Mono (Channel)	no	<b>yes</b>	
Pitch Bend	no	<b>yes</b>	
Control Change	no	<b>yes</b>	
Program Change	no	<b>yes</b>	
System Exclusive	no	no	
System Common			
Song Position	no	no	
Song Select	no	no	
Tune Request	no	no	
System Realtime			
Clock	no	no	
Commands	no	no	
Aux Messages			
Local On/Off	no	no	
All Notes Off	no	<b>yes</b>	
Active Sensing	no	no	
System Reset	no	<b>yes</b>	

## Appendix B: Oscillator Waveform Types and Ranges

Waveform Types:

Sine, Triangle, Sawtooth, Square1, Square2, Square3, Organ1, Organ2, Organ3, Spectra1, Spectra2, Spectra3, Spectra4, RichSaw1, RichSaw2, RichSaw3, RichSaw4, SawSpec1, SawSpec2, VintSaw1, VintSaw2, VintSaw3, SawBass1, SawBass2, SawBass3, SawBass4, SawBass5, SawBass6, SawBass7, SawBass8

Waveform Ranges:

-2, -1, 0, +1, +2 Octaves

## Appendix C: LFO Sync Settings

Off, 16/1\*, 16/1, 16/1T, 8/1\*, 8/1, 8/1T, 4/1\*, 4/1, 4/1T, 2/1\*, 2/1, 2/1T, 1/1\*, 1/1, 1/1T, 1/2\*, 1/2, 1/2T, 1/4\*, 1/4, 1/4T, 1/8\*, 1/8, 1/8T, 1/16\*, 1/16, 1/16T, 1/32\*, 1/32, 1/32T, 1/64, 5/16, 7/16, 9/16, 5/8, 11/16, 13/16, 7/8, 15/16.

Note: "T" stands for Triplet and "\*" stands for a dotted note. In the case of a dotted note, the note duration is equal to 1.5 times its original un-dotted value.



For a translation of these Note values into 1/16<sup>th</sup> and back see this overview table:

SYNC	1/16 <sup>th</sup>	SYNC	1/16 <sup>th</sup>	1/16 <sup>th</sup>	SYNC	1/16 <sup>th</sup>	SYNC
'4/1	64	1/8*	3	0.25	'1/64	8	'1/2
4/1T	42 2/3	'1/8	2	'1/3	'1/32T	9	'9/16
2/1*	48	1/8T	1 1/3	'0.5	'1/32	10	'5/8
'2/1	32	1/16*	1.5	'2/3	1/16T	11	'11/16
2/1T	21 1/3	'1/16	1	0.75	1/32*	12	1/2*
1/1 *	24	1/16T	'2/3	1	'1/16	13	'13/16
'1/1	16	1/32*	0.75	1 1/3	1/8T	14	'7/8
1/1T	10 2/3	'1/32	0.5	1.5	1/16*	15	'15/16
1/2*	12	1/32T	'1/3	2	'1/8	16	'1/1
'1/2	8	'1/64	0.25	3	1/8*	24	1/1*
1/2T	5 1/3			4	'1/4	32	'2/1
1/4*	6	'5/8	10	5	'5/16	48	2/1*
'1/4	4	'7/8	14	6	1/4*	64	'4/1
1/4T	2 2/3			7	'7/16		

Note: Some Values are directly written in 16<sup>th</sup> like for example the 11/16<sup>th</sup>

## Appendix D: Modulation Sources and Destinations

### Modulation Sources

Note played Log	The note being played with exponential response. The modulation value follows the frequency of the played note (bipolar).	
Note played Lin	The note being played with a linear response. The modulation value follows the note number (bipolar).	
Velocity	The MIDI note-on velocity information. The harder the key is hit, the higher the modulation value (unipolar).	
Aftertouch	MIDI pressure / aftertouch information. Alpha responds to both polyphone and monophone aftertouch (unipolar)	
Pitch Wheel	The value of the pitch-Wheel is taken as modulation source, maybe it makes sense to reduce the Pitch Wheel range to 0 when using the Pitch Wheel as modulation source. The Pitch Wheel is bipolar	
Modulation Wheel	The MIDI modulation wheel (MIDI CC 1) (unipolar )	
Amp Envelope	The envelope of the Main Amplitude. This envelope controls the overall Volume, however if it makes sense within the sound to do so, the envelope can be used as a modulation source (unipolar).	
Filter Envelope	The envelope of the Filters Cutoff. This envelope controls the Filter cutoff frequency, however if it makes sense within the sound to do so, the envelope can be used as a modulation source (unipolar).	
LFO 1 to 3	LFO 1 to 3 (bipolar).	
Random	There are also two random sources: Random Unipolar and Random Bipolar. This random value stays constant during the note being played but changes upon each trigger of a new note. So it's different for each note. To have a modulation source which is random and changes while the notes are played, use a LFO with the Noise waveform.	
Alternate	The Alternate modulation source changes between values of 1 and -1 for each note. (bipolar).	
Constant	A constant value of 1, sometimes useful to apply static modulation, e.g. of the oscillator symmetry. (unipolar)	

## Modulation Destinations

Osc x Amplitude	<p>The amplitude of Oscillator 1 to 2, used in example for tremolo or for adjusting the balance of the oscillators in the sound.</p> <p>N.B. In order to create the classic tremolo effect it is better to use Main Amplitude as the modulation destination as this is applied to the whole voice (all Oscillators).</p>
Osc x Pitch	<p>The pitch of the respective Oscillator, used in example for vibrato or to control detuning of the oscillators.</p> <p>N.B. In order to create the classic vibrato effect it is better to use Main Pitch as the modulation destination as this is applied to the whole voice (all Oscillators).</p>
Osc x Symmetry	<p>Symmetry of the respective Oscillator waveform, used to thicken a sound or make it swirl (in the old days called PWM). The effect depends on intensity and modulation speed, typically used with an LFO as modulator.</p>
Osc 2 Ringmod	<p>Controls the Ring-Modulator of Osc 2, thus allows to dynamically change the mix of oscillator 2 and the ring modulators output.</p>
Noise Amplitude	<p>The amplitude of noise oscillator, mainly used for dynamically adjusting the balance of the noise oscillators in the sound.</p>
Filter Drive	<p>Changes the pre-Filter Drive (Filter Saturation)</p>
Filter Cutoff	<p>Cutoff frequency of the Filter</p>
Filter Cutoff FM	<p>The intensity of the FM modulation of the filters cutoff frequency.</p>
Filter Resonance	<p>Resonance of Filter 1 or 2, a rather subtle effect, sometimes used with an LFO or for Keyscaling (Note Lin or Note Exp source) to adjust Resonance over the key range.</p>
Main Amplitude	<p>Overall amplitude of all Oscillators. Often used for tremolo or with Key Lin to adjust the volume over the keyboard.</p>
Main Pitch	<p>Overall pitch of all oscillators. Often used for vibrato.</p>
Main Pan	<p>Overall pan position of the whole voice. Use that with “Alternate” as a modulator for a sound that changes sides with each new note.</p>

Matrix Depth x Intensity of the first eight entries (line 1 to 8) in the Modulation Matrix, often used with the ModWheel as source to control a specific modulation parameter.  
For Vibrato one would have for example line 1 of the Matrix read

LFO	0:00	Main Pitch
-----	------	------------

and another Line would be

Modulation Wheel	2:00	Matrix Depth 1
------------------	------	----------------

for a 2 semitones Mod-Wheel controlled Vibrato.

LFO Speed x Speed of LFO 1 to 3, this allows tempo changes of the LFO to be programmed. This can be used for example with “Key Lin” to make the LFO run faster with higher notes.

## Appendix E: Using TUN Files

By Jacky Ligon

### About Microtuning

Microtuning makes it possible for musicians and composers to change the underlying pitch scales of musical instruments, whereby they may explore and compose with many different types of ethnic, historical and contemporary alternative intonation systems. Microtuning musical instruments enables musicians to use unique sounding scales which may have pitches lying between the notes of our familiar Western 12 tone equal tempered scale.

These alternative intonation systems and methods of microtuning musical instruments to pitches found in the cracks of the Western 12 Tone Equal Temperament, are one of the things that gives the music of such places as Bali, India, Africa, Thailand, Turkey and the Middle East a special sonic quality, but is also something that is of immeasurable value to contemporary acoustic and electronic musicians and composers, who may require a more broad palette of pitches for their music.

The quest for creating beautiful and musically useful intonation systems has been a perpetual process of discovery and debate amongst musical theorists, mathematicians and musicians going back to early history. Quite often the reasons for microtuning musical instruments may involve improving the consonant intervals of a tuning-system for more euphonious sounding harmonies, as well as offering a wider variety of choices for melody. Microtuning an instrument can sometimes mean there may be less, or more, than 12 tones in an octave, or even that the octave itself may be stretched or compressed. Microtuning is a vast musical frontier, rich with historical lore, music and an infinity of exciting musical possibilities for the adventurous sonic explorer.

## Creating TUN Microtuning Files with Scala

Scala is a freeware musical software utility developed by Manuel Op de Coul in the Netherlands, which can be used for the creation and analysis of historical, ethnic and contemporary microtunings. A powerful capability of Scala is that it enables the user to create the proprietary tuning data required for microtuning a wide range of hardware and software synthesizers and samplers.

Scala may also be used to create the TUN format microtuning files needed to explore microtunings with this software instrument. The Scala home page is at: <http://www.huygens-fokker.org/scala/>

## Specifying the Reference Frequency of a Microtuning

One of the powerful capabilities of the TUN format and Scala is the ability to specify the frequency and MIDI-note number for the pitch base of a microtuning. This becomes a very important consideration when one is using a number of different synthesizers and wishes to keep them in all tune with a given base frequency.

It is a common requirement for musicians and composers to be able to specify concert pitches such as A440 Hz (MIDI-Note 69) or C261.6256 Hz (MIDI-Note 60) as a reference pitch for a microtuning, however, the flexibility of the TUN format enables one to specify this frequency arbitrarily, so that any base frequency may be assigned to any MIDI-note number. In Scala this important parameter is called the Reference Frequency.

Being able to specify a particular MIDI-note number on the MIDI controller and its associated Reference Frequency, provides a way to map a microtuning to a common base pitch, making it easier to navigate the instrument when the intonation system may have more or less than 12 tones per octave, or where one may need to map the notes of a microtuning to fall on certain physical keys.

## Important Note

When musicians use TUN microtuning files with this software instrument, the above mentioned mapping properties will override the Master Tune setting, which is set to a default of 440 Hz (found on the Setup page). Normally when one is using the default 12 Tone Equal Temperament tuning, the Master Tune setting can be used to set pitch offsets around the standard concert pitch of A440 Hz, however, when one has specified another pitch base for a microtuning when creating TUN files in Scala, these settings will determine the actual Reference Pitch for the intonation system being used.

## Appendix F: Alpha Versions

	ALPHA 3	Alpha Beat	Alpha CM	ALPHA FREE
Polyphony	32 voices	16 voices		
Oscillators	2 Osc + Noise Ring Modulator Freerun switchable	2 Osc	2 Osc Ring Modulator	2 Osc
Filter	LP12, LP24, LP24+, BP12, HP12 with Filter FM		LP12, LP24, LP24+, BP12, HP12	
Filter Saturation	Yes, can be modulated	Yes		
Envelopes	ADSFR (Attack, Decay, Sustain, Fade, Release) for Amplitude and Cutoff (Cutoff Envelope may be inverted)			
LFO	3 LFO's 6 Waveforms Syncable to Tempo Mono/Poly Mode (Freerun)	1 LFO, 6 Waveforms Syncable to Tempo		
Portamento/Glide	Mono & polyphonic Normal, Held and Auto-Bend-Mode adjustable Time or Rate	Mono & polyphonic Normal, Held and Auto-Bend-Mode adjustable Time		
Sound import of	Alpha 2, Free-Alpha, Alpha-Beat, Alpha-CM, Alpha 3	Alpha-Beat, Free-Alpha	Alpha-CM, Free-Alpha	Free-Alpha
Precision	90 (analog warm) to 100% (digital cold)	No		
Master Tune	Yes, 415.3 to 466.2 Hz	No		
Modulation Matrix	11 slots 14 sources 26 destinations	11 slots 9 sources 20 destinations	11 slots 9 sources 20 destinations	11 slots 9 sources 19 destinations
Pitch Bend Range	Adjustable up/down, each 1 to 24 semitones, saved per sound	Fixed 2 semitones		
Patch Creator	Adjustable 1% to 100% modification	No		
Variable UI Size	M (100%), L (150%) and XL (200%)	Single Size M (100%)		
Microtuning	TUN File support	No		
Factory Sounds	About 850	About 64		



## Appendix G: Predefined ECS assignments

The following MIDI-CC-parameter-assignments are automatically set up on start-up of Alpha 3. Of course, the assignments can be replaced with your own preferences using the ECS function.

CC 12 Main Level	CC 84 Glide Time
CC 14 Spread	CC 85 Glide Mode
CC 15 Exactness	CC 86 Glide Bend
CC 20 Osc 1 Waveform 1	CC 88 Amplitude Envelope Attack
CC 21 Osc 1 Range 1	CC 89 Amplitude Envelope Decay
CC 22 Osc 1 Waveform 2	CC 90 Amplitude Envelope Sustain
CC 23 Osc 1 Range 2	CC 102 Amplitude Envelope Fade
CC 24 Osc 1 Waveform Mix	CC 103 Amplitude Envelope Release
CC 26 Osc 2 Waveform 1	CC 104 Amplitude Velocity Response
CC 27 Osc 2 Range 1	CC 105 LFO 1 Attack
CC 28 Osc 2 Waveform 2	CC 106 LFO 1 Waveform
CC 29 Osc 2 Range 2	CC 107 LFO 1 Frequency
CC 30 Osc 2 Waveform Mix	CC 108 LFO 1 Sync
CC 31 Filter FM Source	CC 110 LFO 2 Attack
CC 69 Osc Detune	CC 111 LFO 2 Waveform
CC 70 Osc Ring-Modulation	CC 112 LFO 2 Frequency
CC 71 Osc Balance	CC 113 LFO 2 Sync
CC 72 Noise Level	CC 115 LFO 3 Waveform
CC 73 Filter FM Depth	CC 116 LFO 3 Sync
CC 74 Filter Envelope Depth	CC 117 Chorus Wet
CC 75 Filter Envelope Attack	CC 118 Chorus Time
CC 76 Filter Envelope Decay	CC 119 Chorus Rate
CC 77 Filter Envelope Sustain	CC 25 Matrix Modulation Amount 1
CC 78 Filter Envelope Fade	CC 87 Matrix Modulation Amount 2
CC 79 Filter Envelope Release	CC 109 Matrix Modulation Amount 3
CC 80 Filter Cutoff Frequency	NRPN 1024/1025 sound previous/next
CC 81 Filter Resonance	NRPN 1026/1027 Bank previous/next
CC 82 Filter Drive	
CC 83 Filter Type	