ANALOG WAVEGUIDE

Manual of operation

Compatible with Firmware version 1.2.3
Hardware Revision 3.2 (production)

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The Center for Haptic Audio Interaction Research

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Figure 1.1: Front view of the Analog Waveguide controls.

The Analog Waveguide is a resonator, the fundamental component of all physical modeling techniques. It transforms an external, unpitched excitation audio signal, such as a click, a burst of noise, or any sound captured by a contact microphone, into a rich, pitched sound. The Analog Waveguide can imitate e.g., a drum skin, a string, a hand pan, a woodblock, or a sheet of metal that vibrates in response to an external signal.

This Eurorack module is arguably the first analog resonator capable of producing rich inharmonic resonances with dense spectra. It effortlessly allows hundreds of partials to resonate in harmony with a continuous audio input signal. It is conceived to explore analog physical modeling techniques and adds this extra bit of analog warmth, tiny fluctuations and lush sounding saturation of true analog delays, VCAs and filters.

1.1 WHAT IS A WAVEGUIDE?

In physical modeling sound synthesis, a waveguide is a structure that simulates how waves propagate through a physical medium—typically air columns, strings, or acoustic tubes. A waveguide models the motion of waves by using:

- Delay lines → to simulate wave travel time
- Filters → to simulate frequency-dependent losses, dispersion, and reflections
- Scattering junctions → to simulate branching or impedance mismatches (e.g., tone holes, string bridges)

In essence, a waveguide is the equivalent of a vibrating physical object, broken down into simple propagation and reflection components. Waveguides can model different acoustic instruments very well, e.g.:

- **Strings instruments:** A string is modeled as two delay lines carrying waves in opposite directions; reflections occur at the bridge and nut.
- **Wind instruments:** The bore of a flute or clarinet is a tube-like waveguide; scattering models tone holes, the reed, mouthpiece geometry, etc.

- **Percussion:** Waveguides can approximate bars, plates, membranes (sometimes via networks of 1D waveguides).

1.2 HOW DOES THE ANALOG WAVEGUIDE WORK?

The module has two input and two output channels. They can be understood as as a stereo pair of signals: left and right, but they can also be mapped to x- and y-axes in a two-dimensional Cartesian space.

The left channels is oscillating on the x-axis, while the right channel is oscillating on the y-axis.

In recording and broadcasting this view is known as stereo phase analyzer or stereo phase scope.

If the input is a monophonic signal (X and Y are identical), the image on the X/Y view is a signal oscillating in a diagonal line from the top right quadrant (+x, +y, I) to the bottom left quadrant (−y, −x, III). If the left channel is the shifted 180° in phase to the right channel, the signal will oscillate from the top left quadrant (−x, +y, II) to the bottom right quadrant (+x, −y, IV)[1].

$$\begin{array}{c|c} II & I \\ \hline III & IV \end{array} \tag{1}$$

Equation 1.1: The four quadrants of a Cartesian plane.

The Analog Waveguide has the Y channel prepatched to the X channel internally,[2] so if you have nothing plugged into the right input, the left input is used on both channels. If you intentionally want the right input to be mute just put a dangling cable into the right input or turn the gain all the way down.

1 Mixed to mono this signal would be silence; very inconvenient for listeners on the kitchen radio. Hence the importance of the stereo phase analyzer in broadcasting.

2 A connection “behind the front panel” is also called normalled connection, we use this term in the further document.

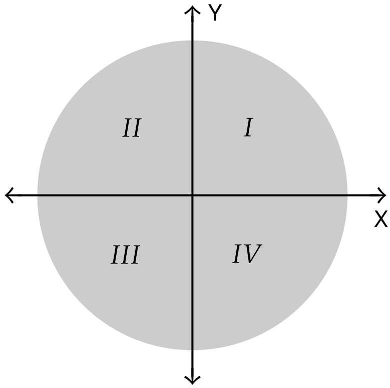


Figure 1.2: Analog Waveguide Display with quadrants and axes.

Now the rotation is simple if you understood everything so far. Every time we feed the delay output back to the input for the resonating effect, we can rotate this point on the two dimensional space, representing the stereo signal, around the origin point of the coordinate system (0,0) by a given angle. Since the tail of the resonance gets rotated every time in feeds back into the delay line, the rotation also represents the speed of the resonance changing its timbre in the tail. This velocity is the highest at 90° and 270° while at 180° is a flip-flopping reflection.

Mathematically, and also electrically, the rotation is a multiplication, subtraction, and addition with sine and cosine of the angle. If θ[3] is the angle of rotation x, y the old position and x', y' the new position in the Cartesian space:

$$\begin{aligned} x' &= x \times cos\theta - y \times sin\theta \\ y' &= x \times sin\theta + y \times cos\theta \end{aligned} \tag{2}$$

The output of the module is the left and right (or X and Y) component of the signal. In other words: the two di-

[3] The Greek lower case letter Theta, in geometry a commonly used symbol for an angle.

mensional signal gets projected back on the two axis. It's basically a specific way of cross mixing the two channels.
This rotation mix has two effects:

1. The sum is always power preserving. This is an essential property for everything you do in resonators, because then your resonating sound is not exploding or dying.
2. The spectral spread of the overtones gets bent in-harmonically. A single feedback delay (Karplus-Strong synthesis) always sounds like one dimensional resonators: strings) our Analog Waveguide can do string sounds (with the rotation angle of 0°, but additionally can sound like two dimensional resonators: gongs, plates, cymbals, membranes, drum-heads, hang drums, etc.

The paper we published is providing an explanation why this in-harmonic distortion happens: Wegener and Neupert 2024

1.3 TECHNICAL SPECIFICATION

1.3.1 *Dimensions*

Dimension	**Size**
Width	24 HP
Depth in rack (under panel)	34 mm
PCB height in rack	112 mm (some racks may be smaller!)

1.3.2 *Current draw per Eurorack rail*

10-Pin Eurorack power IDC connector, cable is supplied.

Rail	Current
+12 V	138 mA
-12 V	110 mA
+5 V	Not Connected

1.3.3 *Input/Output specification*

Port	Range
1 V/Oct	–1.5 V to 5.5 V
FM	–8 V to 8 V
Lowpass	–8 V to 8 V
Feedback	–8 V to 8 V
Rotation	–8 V to 8 V
Audio In	–8 V to 8 V
Audio Out	5 V to 5 V (Wet); –10 V to 10 V (Dry)
USB-C	For Firmware update. DFU Mode Device ID 0483:df11

1.4 SIGNAL FLOW DIAGRAM

The Analog Waveguide is designed to be flexible and represent multiple physical or pseudo-physical acoustic models. It consists of several blocks that can be partially reconfigured to realize different physical models or acoustic functions.

Fig. 1.3 shows the simplified form of the signal flow block diagram with the base functionality (CV sources, normal connections and knobs omitted). It features:

- Two delay lines → to simulate wave travel time,

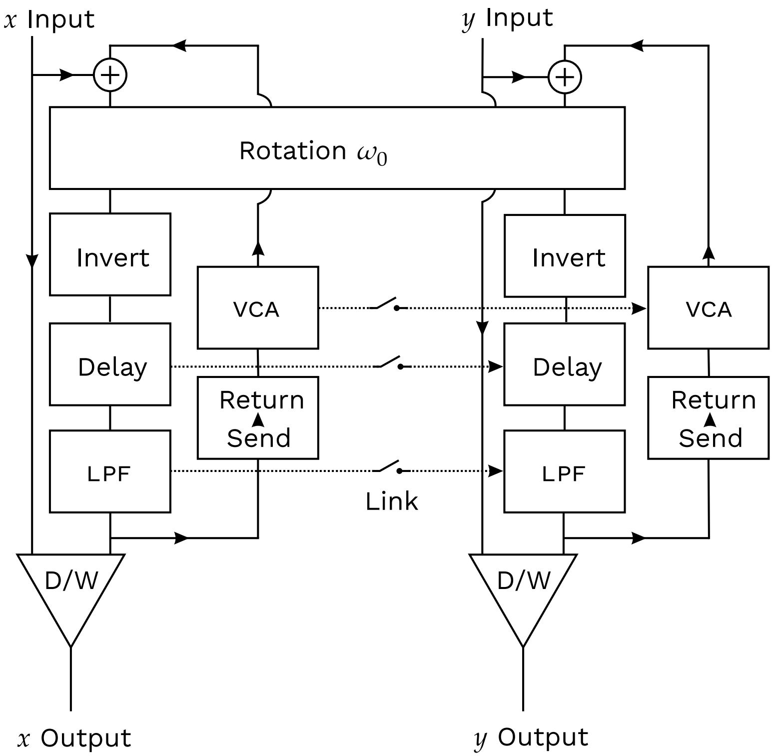


Figure 1.3: Simplified signal flow block diagram.

- Two lowpass filters → to simulate frequency-dependent losses,
- A rotation scattering junction → to simulate branching and reflections of the acoustic waves,
- Two Invert switches to invert the individual delay outputs,
- Several link switches to couple the CV control of the Y channel to the X channel, and
- Two Dry/Wet output mixers.

With these building blocks we can realize a series of physical models, that we describe in the patch examples (Section 4).

Note that the Analog Waveguide can be also understood in the context of reverberator design, this is where its concept originated. See Gerzon 1971 and Gerzon 1972 for the original design. It can be mathematically best described as a Unitary Feedback Delay Network. A general description of such Networks can be found in Stautner and Puckette 1982. An implementation in Pure Data can be found in Puckette 2011.

2 INDIVIDUAL FUNCTIONS

The following sections are realized in pairs for the X- and Y channel.

2.1 PITCH SECTION

Figure 2.1: Pitch section.

1V/OCT CV INPUT: Input for pitch information coming from a keyboard or sequencer. This affects the delay time of the BBD delay chip. The higher the pitch, the shorter the delay time. The 1V/Oct CV Input of channel Y is normalled to the 1V/Oct CV Input of channel X. The input range is −1.5 V to +5.5 V. The pitch tracks perfectly over this seven octaves range. The sample rate is 1.4 kHz.

PITCH KNOB: Adjusts the pitch of the channel. The pitch ranges from C0 to C4 when nothing (or 0 V) is plugged into the 1V/Oct input. Note that the 1V/Oct input adds an offset to the pitch selected with this knob. Most pitch sources are positive only, that means you would want to turn the pitch knob counter clock wise, when a 1V/Oct source is plugged in, to let the 0V CV signal play the lowest note. If nothing is connected into the 1V/Oct CV Input of channel Y and the pitch link switch is turned off, its pitch knob controls the offset to the pitch of the X channel.

FM CV INPUT: Frequency modulation input to modulate the delay frequency with an LFO or another modulation source. The FM CV Input of the Y channel is normalled to the FM CV Input of the X channel. The input range is -8 V to 8 V, the sample rate is 1.4 kHz.

FM CV ATTENUVERTER KNOB: Adjusts the depth of the frequency modulation. The zero point is in the middle with a center detent. Turning the knob left inverts the modulation.

FM CURVE SWITCH: This switch selects exponential (left position) vs linear (right position) frequency modulation of the delay frequency. For slow vibrato effects choose exponential modulation, for fast audio rate timbre changing FM effects choose linear frequency modulation, as it preserves the perceived pitch at high rates.

2.2 LOWPASS SECTION

LOWPASS KNOB: Controls the 6 dB/octave lowpass filter to simulate frequency dependent damping. The frequency range is $\sim$50 Hz to 20 kHz.

LOWPASS CV INPUT: CV input to control the cutoff frequency of the lowpass filter. This input is normalled to the

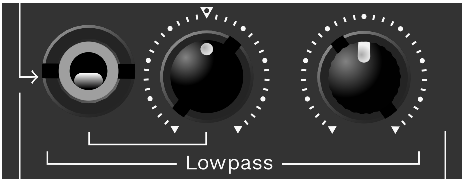


Figure 2.2: Lowpass section.

1V/Oct pitch input to allow key tracking of the filter. The input range is $\pm 5\,\mathrm{V}$.

LOWPASS CV ATTENUVERTER KNOB: Attenuverter to attenuate or invert the CV signal. When no input signal is connected to the Lowpass CV input, this knob controls the amount of key tracking. Turn the knob clock wise for positive keytracking and counter clock wise for negative key tracking.

2.3 FEEDBACK SECTION

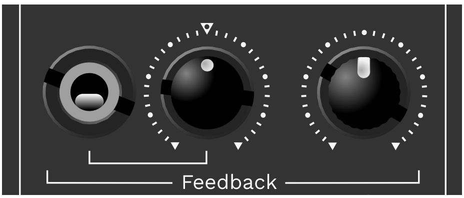


Figure 2.3: Feedback section.

FEEDBACK KNOB: Controls the VCA to simulate damping. Turn counter clock wise for short notes (fast decay) and clock wise for long sustained notes or self oscillating drones.

FEEDBACK CV INPUT: CV input to control the VCA. The input range is $\pm\ 5\,\mathrm{V}$.

FEEDBACK CV ATTENUVERTER KNOB: Attenuverter to attenuate or invert the CV signal.

2.4 INPUT AND OUTPUT SECTION

Each channel, X and Y has its own in- and output section, see Figure 2.4.

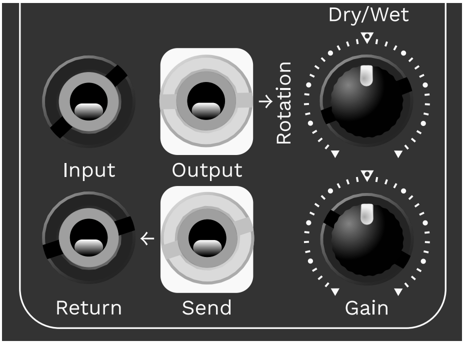


Figure 2.4: Input and output section.

INPUT JACK: This is the audio input to excite the analog waveguide. Each channel channel has its own input, so they can be independently excited, e.g. for duophonic play. The input range is $\pm 12\,\mathrm{V}$. Note that the Y input is normalled to the X input. That means the signal patched into X will automatically show up on the Y input. You can break this connection by inserting a cable into the Y input.

OUTPUT JACK: Each channel has an independent output that can be patched as stereo signals. The output range is $\pm 5\,\mathrm{V}$. Note that the audio output of channel X is normalled to the rotation CV input.

SEND/RETURN JACKS: The Send/Return jacks allow to patch in a other modules, such as filters, phasers, flangers, or another waveguide, into the feedback loop of the delay lines. The Send output is normalled to the Return input. Once a signal is inserted into the Return jack, the normal connection is broken. This allows you to use the Send output as a copy of the audio output of the channel and e.g. use it as an FM modulation source.

DRY/WET KNOB: This knob mixes the audio input with the audio output of the analog waveguide. The middle position is 50 % in / 50 % out at the center detent.

GAIN KNOB: This amplifies or attenuates the audio input. The gain spans from 0 to 10 ×.

2.5 ROTATION SECTION

ROTATION KNOB: Controls the angle of rotation between the channels. This is an endless potentiometer,[1] so the angles of 0 degrees and 360 degrees fall together to the same setting of the knob. See Figure 2.5.

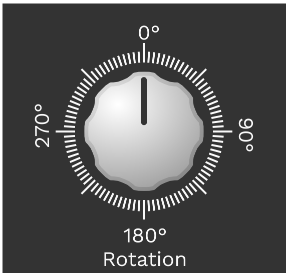


Figure 2.5: Rotation knob.

ROTATION CV INPUT: CV input signal for the rotation angle. This input is normalled to the X channel's audio output for non-linear FM like sounds. The input range is ±8 V. The sampling frequency of this input is 48 kHz.

1 Endless potentiometers are somewhat exotic devices but are more robust and have a better longlevity then digital encoders.

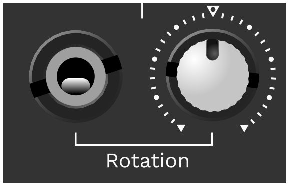


Figure 2.6: Rotation CV input and attenuverter.

ROTATION CV ATTENUVERTER KNOB: Attenuverter to attenuate or invert the CV signal. Turning this knob (See Figure 2.6) when no input is patched, will control the depth of feedback between the audio output of the X channel and the CV input for rotation angle control. This will introduce non-linear (loudness-dependent) behavior to the sound. Louder sounds will be heavier modulated and will therefore have brighter spectra. This is great for metallic drum sounds.

INVERT SWITCHES: Each channel has an invert switch to invert the signal after it is rotated. This is akin to a phase inversion of 180 degrees or a multiplication by -1. Inverting both channels and rotating by 180 degrees has the same effect. So using both at the same time will effectively cancel the rotation or inversion. Inverting one of the two channels however will transform the rotation into a mirroring operation with a mirror axis at half of the set rotation angle.

2.6 LINK SWITCHES

Each link switch has an on and an off position marked with I and O. When the Link is active, the Y controls are overwritten by the X controls. See Figure 2.7.

LOWPASS LINK SWITCH: This switch controls whether or not the lowpass cutoff frequency of the Y channel follows the X channel. Flipping the switch to the right enables the link, flipping it left disables it. With enabled link the lowpass section of the Y channel is disabled.

FEEDBACK LINK SWITCH: This switch controls whether or not the amount of feedback of the Y channel follows the X channel. Flipping the switch to the right enables the link, flipping it left disables it. If you want use both channels independently and have different note lengths, then disable the link. With enabled link the feedback section of the Y channel is disabled.

PITCH LINK SWITCH: This switch controls whether or not the pitch of the Y channel follows the X channel. Flipping the switch to the right enables the link, flipping it left disables it. With link enabled the Y channel can still be independently frequency modulated, but the pitch knob and 1 V/Oct input of the Y channel are disabled.

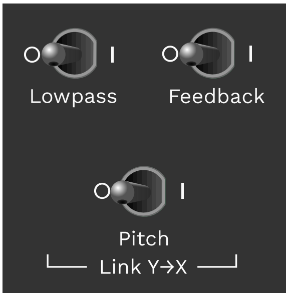


Figure 2.7: Link switches.

3 DISPLAY AND LEVEL METERS

3.1 DISPLAY

The Display draws the modules output on xy coordinates, where the x-axis represents the amplitude of the signal in the X channel and the y-axis represents the amplitude of the signal in the Y channel. This signal representation is also known as a XY mode in oscilloscopes and the observed curves are called Lissajous patterns.

A basic understanding of Lissajous patterns can help to understand how the parameters of the module get reflected in the display.

STRAIGHT LINE: If the display shows a straight line with a 45 degrees angle, that means both signals are exactly the same amplitude, frequency and phase. If the amplitude of the X channel is greater then the Y channel, the line moves towards the x-axis and vice-versa. So having a signal only in the X channel will result in a horizontal line whereas having a signal in the Y channel results in a vertical line on the display.

CIRCLES OR ELLIPSES: If the display shows a circle or a 45 degree tilted ellipse, that means both signals are exactly the same frequency and amplitude, but at different phases. If the phase shift is 0 degrees we have a straight line at 45 degrees again, that turns into a 45 degrees tilted ellipse when the phase shift is between 0 and 90 degrees. At 90 degrees we see a perfect circle on the screen. A phase difference between 90 and 180 degrees will yield an ellipse tilted at 135 degrees that will reach another perfect circle at a phase difference of 270 degrees.

SQUARES OR RECTANGLES:

LISSAJOUS PATTERNS: When the frequencies of the delay channels differ by integer multiples we get so called Lissajous patterns. The number of lobes on each axis tells us about the ratio of frequencies in X and Y. E.g. if we have three lobes against Y and one lobe against X, the frequency ratio in the channels is 3:1. When the difference in frequencies is a non-integer multiple, the pattern will appear to rotate three-dimensional.

Note that Lissajous patterns mostly appear in harmonic settings of the Analog Waveguide. This is the case at 0 or 180 degrees of rotation.

AXIS-SYMMETRIC PATTERNS: When one of the two delay channels is inverted, we get axis symmetric patterns around a symmetry axis at half the angle of the rotation knob setting.

ROTATION-SYMMETRIC PATTERNS: With none or both of the channels inverted and the frequencies of both channels linked, we get point- (or rotation) symmetric images around the coordinate origin. The complexity of the pattern increases with the rotation angle until it reaches 90 degrees and decreases until 180 degrees, where it increases again.

3.2 CLIPPING INDICATORS

Above the round display are two level meters acting as clipping indicators. Distortion can occur, when the indicators light up red, levels are close to clipping when lighting up yellow, levels are good when green and fade to black when no signal is present. The left clipping detector shows the peak levels of channel X, the right clipping indicator shows the peak levels of channel Y. Use the gain knobs to reduce the input gain or reduce the amount of feedback to avoid clipping if you want to achieve a sound without distortion.

4 PATCH EXAMPLES

Although the Analog Waveguide is not build to faithfully recreate the sound of acoustical instruments, it can create sounds with acoustic behavior, because it is made up of building blocks inspired by physical acoustics. Personally we don't believe that exactly reproducing the sounds of acoustic instruments is necessarily the main goal of sound designers or electronic musicians. We still came up with patches that mimic acoustic instruments. We think this is a great way to memorize settings and what they do to the sound, this is, because it is easier to relate to the sound of a guitar, than it is to a 'exponentially decaying, frequency-dampened harmonic series'. And last but not least: it's fun!

4.1 RECOMMENDATIONS FOR EXCITATION

Finding a good excitation signal for your patch is absolutely vital to good sound design! We have experimented a bit what modules and characteristics make up good exciters. We have found that a two-stage approach is favorable.

The first stage is about creating the initial energy for the note to play, this should be a short needle impulse of up to 50 ms in length. The peak voltage of this impulse dictates the loudness of the note. A needle impulse or a short noise burst contains energy in all frequencies, just like white noise. The impulse just needs to rise fast enough to develop a good bright sounding click to be usable.

The second stage is about the timbre of the note to be struck. We found that bandpass and lowpass filters work well for this purpose. The bandpass will only activate the partials of the resonator that it 'lines up in frequency' with. That means, if we use a bandpass filter to filter out everything except the frequencies between 400 and 600 Hz, then in the Analog Waveguide, only the partials between 400 and 600 Hz will resonate, even if the fundamental is 100 Hz.

For the first “needle impulse” stage you can use:

- Piezo microphones (connected through a preamp to the Analog Waveguide). Attach them to your desk, or a piece of wood and strike it with drum sticks or brushes to capture structure borne sounds and impact noises.
- ADSRs or other short exponentially decaying envelopes with very fast attack times. Beware that not all envelopes have fast enough attacks!
- Drum triggers (short gates with up to 50 ms length).
- Sample and hold circuits.
- Square wave or saw tooth LFOs, if you want to have repeated note patterns anyway.

For the timbre coloration stage, we found the following, non exhaustive, list of filters very suitable. We recommend using dual/stereo filters to color two needle impulses independently and send them to the X and Y channels of the Analog Waveguide.

- Weston Precision Audio: *SF1 Dual / Stereo Filter*
- Vermona: *TwinVCFilter*
- Cwejman *DFA-2, MMF-2*
- Bastl Instruments: *Ikarie*
- Rob Hordijk/Klanbauköln: *Twin Peak Resonator*
- Epoch Modular: *Twinpeak* (Discontinued)

Other approaches for excitation can be:

- Saw tooth wave forms for voice like formant filter sounds.

- Filtered and modulated noise
- Drum Sounds
- Slow sine and triangle LFOs, when the Waveguide is close to self-resonance

4.2 DUOPHONIC STRING & DRUM

You can play the two delay lines as two independent Karplus-Strong physical models of strings and drums. To be able to control every parameter of the two delays independently make sure to

- Switch off all link switches
- Set the rotation angle to zero degrees
- Plug two different 1V/Oct CV sources in 1V/Oct X and 1V/Oct Y
- Plug two different exciters into Input X and Input Y and adjust input levels

Two play two string models set the invert switches to off, to play a drum like sound set the invert switch to on. Play with the lowpass cutoff to control the frequency dependent damping. Use the lowpass attenuverter knobs to control the pitch tracking of the lowpass filter cutoff with the 1V/Oct signals. Make sure that nothing is plugged into the rotation CV input, then turn the rotation CV attenuverter knob to add some non-linearity to the sound. The strings can sound as If struck with a hammer, when this knob is turned up wide enough.

4.3 STRING WITH PLUCK POSITION

To model the pluck position of the string, we send our excitation signal through a comb filter effect before passing it

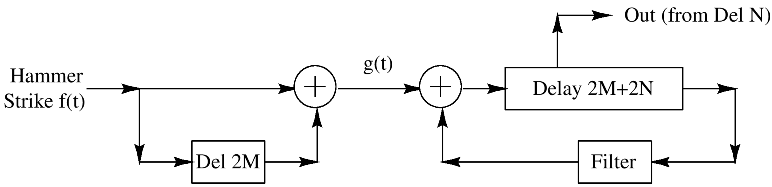


Figure 4.1: Diagram for pluck position. Source: Julius O. Smith https://ccrma.stanford.edu/~jos/pasp/Equivalent_Forms.html

Figure 4.2: Folded string with pluck position modeling.

into the delay line. For this patch we will use both delays in a single voice.

In Fig. 4.2,[1] delayline X constitutes the comb filter, delayline Y models the string resonance. We need to patch the output of delayline X into delayline Y. It is important that the two delaylines don't mix (Rotation = 0°) and their feedback gains and filters are not linked (switches are in 0 position). Their pitch however should be synchronized to

[1] The illustration of this patch was made with a dummy Analog Waveguide module for VCV Rack without any DSP in it, there is no functional Analog Waveguide module for VCV Rack as of today.

implement a tracking comb filter that adapts to the pitch of the string.

Use a modwheel or another modulation source to modulate the frequency of the comb filter (FM input, exponential curve setting). This parameter resembles the pluck position on the string. For every delay time, that is a multiple of the period of the string length, the fundamental will be excited strongly. All other delay times will excite different partial overtones.

The comb filters feedback (delayline X) deepens the effect of partials emerging from the pluck position. It should not resonate longer than the actual string in delayline Y, though.

It helps to use a little bit of damping, we can use the low pass filter in delayline Y to do that. Also, you will get a more convincing string sound when using a high pass filter on the click (or noise burst) signal before it enters the comb filter. For the high pass you will need an additional module that is not shown in fig. 4.2.

> *Tip*
>
> To reliably unmix the delay lines, try to send the click only to the left delay line while listening to the right delay line and move the rotation knob slightly around the zero point. Find the setting where you hear the least of the input signal. Since the Y channel input is normalled from the X input, turn down the Y gain completely or plug a dangling patch cable into the Y input to break the normalled connection.

4.4 MIRROR HARMONICS

4.5 GONG

4.6 SINGING SAW

4.7 HAND PAN

4.8 FRAME DRUM

4.9 WOOD ORGAN

4.10 VINTAGE REVERB

4.11 UNFOLDED STRING MODEL

Figure 4.3: Unfolded string with pluck position modeling.

Here the X channel can be regarded to carry the right going wave and the Y channel the left going wave on a string. Both of them combined form the waveguide string model. In this case our modified rotation scattering junction behaves like an inverting reflection at the nut and the bridge of a string instrument. See Figure 4.3 for an illustration how this would be patched. We can make a model of a string that represents physical reality a bit closer by using two delay lines instead of just one.

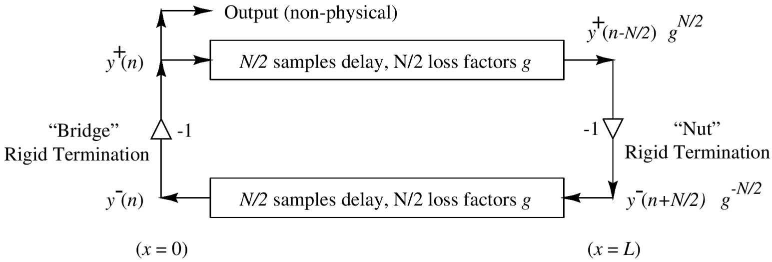


Figure 4.4: Diagram of 'unfolded' string. Source: Julius O. Smith https://ccrma.stanford.edu/~jos/pasp/Computational_Savings.html

When plucking a physical string on an instrument, there are two waves of vibration running through the string in opposite directions, emanating from the pluck position. We can call them the left and right going waves. They reflect at the bridge and the nut and continue to travel in opposite directions, crossing each other once every period without interference until the sound dies out. We can model this process by utilizing one delay per wave direction. E.g., in fig. 4.4 the upper delay line carries the right going wave, the lower delay carries the left going wave. When changing directions, the waves reflect with an inversion, hence the multiplication by −1 at each end of the string.

Hence, if we want to model this with the Analog Waveguide, we need to feed both delay lines into each other with a sign inversion before entering into each delay. So we need to make some consideration on how to configure it to give us two inversions and feed the delay lines into each other in a big loop.

The rotation scattering junction at the Analog Waveguide's core is governed by the following two equations. Each delay output x_d and y_d is mixed into every output x' and y' by a rotation operation.

$$
\begin{aligned}
x' &= x_d \cos\theta - y_d \sin\theta, \\
y' &= x_d \sin\theta + y_d \cos\theta.
\end{aligned} \tag{1}
$$

Additionally there are two invert switches to invert either the X or Y channel, or both. They are indicated by the $\pm$ sign:

$$
\begin{aligned}
x' &= \pm(x_d \cos\theta - y_d \sin\theta), \\
y' &= \pm(x_d \sin\theta + y_d \cos\theta).
\end{aligned} \tag{2}
$$

In a special parameterization, this model can behave just like the waveguide string model. To achieve this, we need to set the rotation angle $\theta = 90$, so $sin\theta = 1$ and $cos\theta = 0$. If additionally the y channel is inverted, we get the original governing equations for the waveguide string model.

$$
\begin{aligned}
x' &= -y_d \\
y' &= -x_d
\end{aligned} \tag{3}
$$

Here the X channel can be regarded to carry the right going wave and the Y channel the left going wave on a string. Both of them combined form the waveguide string model. In this case our modified rotation scattering junction behaves like an inverting reflection at the nut and the bridge of a string instrument.

In fig. 4.4 you can see the necessary parameter settings. Note that it is not necessary to patch the output of one delay into the other since it is already routed (mixed) internally via the rotation core. It is important that you mix the outputs of the two delay lines together in a mono signal, otherwise you will not hear the sound of a string, but instead the right and left going vibrations separately, which is physically impossible.

If you add a delay on the input of one of the two delay lines, you can again simulate a changing pick position as depicted in fig. 4.4. This is a directly analogous situation to the physical string - if you change the plucking position the left and right going waves will arrive at different times at the nut and bridge.

With this model you can also implement different reflection (impedance) behaviors on the bridge vs. the nut. You can access the bridge and nut via the send and return jacks on each delay line.

5

MENU SYSTEM

The menu allows you to adjust some parameters of the module that are not accessible from the front plate, show the firmware version, to initiate firmware updates or to re-calibrate the module.

Navigating the menu

- To enter the settings menu, press the factory settings button on the right side of the module, next to the USB-C port.
- To scroll the menu entries use the rotation control knob.
- To enter a submenu or edit a parameter push the factory settings button.
- To exit or go back to the previous menu, scroll to the "Go back" menu entry and push the button.

On entering the menu, you will be presented with the following options:

Main Menu
Settings
Show firmware version
Firmware update mode
Calibration
Exit Menu

To enter the setting menu, navigate to the “Settings” entry and press the factory settings button.

5.1 SETTINGS

Here you can set parameters related to the display and internal sound processing.

5.1.1 *Display Settings*

This menu allows you to set the timer for the screen saver and the brightness of the display.

DISPLAY BRIGHTNESS
: Adjusts the screen brightness in steps of 1%. In case you encounter noise issues related to the display, try setting this value to 100%. This will deactivate the display dimming that is based on high frequency switching of the back light.

SCREEN SAVER
: Sets the timer for the screen saver between 5s to 2h or “off”.

5.1.2 *Limiter Settings*

The analog waveguide features a digitally controlled analog limiter in the feedback path of each delay line. They operate with the same parameters. You can adjust them in the limiter settings menu.

THRESHOLD [DB]
: Adjusts the threshold in decibels above which limiting starts. Values above -3 dB allow for more distortion. A value of 0dB will deactivate the limiter. If you still encounter distortion try values below -3dB and longer release times.

SOFT KNEE [DB]

Allows for a smooth transition between limiting and distortion. For hard limiting choose 0dB, a positive value will enable smooth limiting, with a bit less pumping and some added distortion.

RELEASE [MS]

Adjust the release time in milliseconds. Larger values will limit more efficiently, while smaller values will result in less of a pumping effect.

RESET

Resets the limiter parameters to the default settings.

6 CALIBRATION PROCEDURES

Your module arrives fully calibrated and it is not foreseen that you would have to adjust the trimmers on the circuit boards or change the calibration values in the firmware. If, maybe with aging components or other circumstances it is necessary to re-calibrate the module, here are all the steps explained. A full calibration of the module requires BBD calibration and rotation & channel calibration.

For a totally uncalibrated module, it is recommended to follow all calibration routines in exactly the order they are presented here. For fine adjustments of an already calibrated module, single steps can be carried out in any order.

For best results, let the module heat up for approximately 15 minutes before starting calibration.

6.1 BBD CALIBRATION

For the following steps you will need an oscilloscope and a voltmeter. We are assuming a module with up to date firmware which is powered via the Eurorack power connector.

6.1.1 *Adjust Delay Input Offsets*

1. Connect the voltmeter/oscilloscope to testpoint marked **DlyIn** on the left side of the bottom circuit board.
2. Adjust trimmer labelled **Vbias X** until you read 6 V.
3. Repeat the same for the opposite side with **Vbias Y**.

6.1.2 *Adjust Clock Trimmer*

1. On the module turn the feedback pots to their minimum CCW. Rotation offset to center position, Rotation to 0°.
2. Probe the testpoint **DlyOut X** with the oscilloscope on the south end of the lower circuit board.
3. Set oscilloscope to AC-coupling, the vertical axis to a 100 mV/grid, horizontal axis to 20 µs.
4. Set the trigger on the oscilloscope to DC-coupling with a threshold of 100 mV.

→ You should see a signal with 250-700 mV peak-to-peak.

5. Adjust trimmer **ClkTrim X** until the waveform becomes evenly repetitive.
6. Repeat the same for the opposite side with **ClkTrim Y**.

6.2 ROTATION & CHANNEL CALIBRATION

For the following steps you will need a very small slotted screwdriver and a small Philips screwdriver.

Some steps are mutually dependent which means that it may be necessary to follow the calibration procedure more than one time to get optimal results.

6.2.1 *Trim CV feed-through*

Due to manufacturing tolerances of the analog multiplier chips on the module, it is necessary to trim voltage offsets at the rotation control voltage input, such that higher frequency modulation signals will not be heard at the output.

There are two trimmers on the top side of the upper PCB labeled **X OFST** and **Y OFST**. These adjust a millivolts' worth of offset to the inputs of the multipliers. You can adjust them with a small slotted screw driver (or a dedicated trimmer adjustment tool). See figure 6.1.

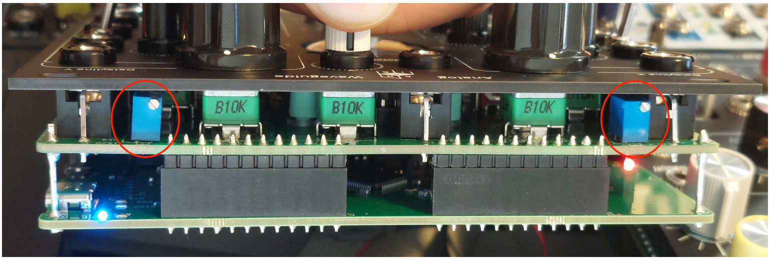


Figure 6.1: The two trimmers for the CV feed-through can be found on the top of the module.

To help with the procedure, there is a program in firmware to generate a sine wave modulating the rotation automatically. Follow these steps to minimize the CV feed-through:

Preparation

1. Turn the gain X and Y knobs on the front panel fully CCW to suppress any input signal
2. Turn the feedback X and Y knobs on the front panel fully CCW to suppress feedback.
3. Turn the lowpass knobs fully CW to open both filters.
4. Connect outputs X and Y to a speaker or headphone.

Calibration

1. Press the factory settings button.
2. With the rotation control knob, navigate to the

   ```
   Calibration → Trim CV feedthrough
   ```

   menu entry and press the factory settings button to enter.
3. You should here a sine wave with a frequency of 220 Hz.
4. Now turn the left trimmer until you reach the point where the sine tone is the quietest.
5. Now turn the right trimmer until you reach the point where the sine tone is the quietest.
6. Repeat steps 4. and 5. until the sine tone cannot be trimmed quieter in volume.
7. The resulting tone should be so quiet, that you can here some background noise with enough amplification.

6.2.2 *Trim XY decay boost*

To give the feedback knobs on the front panel a musical range, i.e. self resonance should start to occur at a position of around 3 o'clock on the feedback knobs, we need to adjust the one-turn trimmers on the bottom side of the upper PCB. The trimmers are labeled **Decay Boost X** and **Decay Boost Y**. Be careful when adjusting them, they are very touchy. See figure 6.2.

To help with the procedure, there is a program in firmware to measure the loudness of self resonance. Follow these steps to adjust the trimmers:

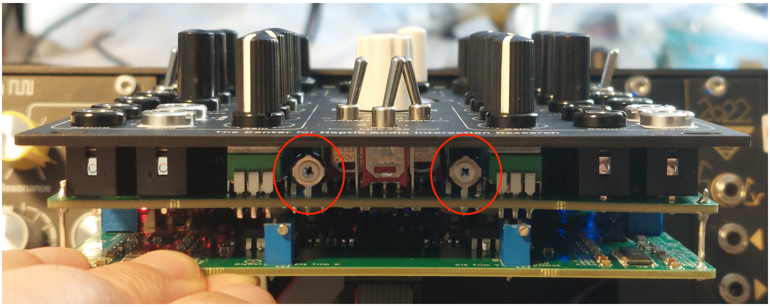


Figure 6.2: The two trimmers for the XY decay boost are the black and white trimmers with the cross slot through hole directly under the frontplate.

Preparation

1. Let the module heat up for approximately 15 minutes.
2. Turn down the volume of your speakers or shut them off totally.
3. Turn the Gain X and Y knobs on the front panel fully CCW to suppress any input signal
4. Turn the Feedback X and Y knobs on the front panel to a 3 o'clock position.
5. Turn the Lowpass knobs fully CW to open both filters.
6. Deactivate the Feedback Link switch on the front panel.

Calibration

1. Press the factory settings button.
2. With the rotation control knob, navigate to the

   ```
   Calibration → Trim XY decay boost
   ```

   menu entry and press the factory settings button to enter.
3. With a small Phillips screwdriver, turn the "Decay Boost X" trimmer CCW until you hear a feedback tone on the left speaker and you see the bar graph rms loudness meter showing a signal.
4. Carefully adjust the trimmer such that the top of the bar graph resides in the green back lit area.
5. The RMS signal power should be between −3dB and −9dB. The ideal target being at 6dB.
6. Press the factory settings button again and repeat the process for "Decay Boost Y" trimmer.
7. When the RMS value is good, press the factory settings button tu return to the menu.

6.2.3 *Trim XY channel balance*

To balance the X and Y channels in their overall volume it is necessary to adjust the **Y Gain** trimmer on the right side of the upper PCB. See Figure 6.3.

Follow these steps to adjust the trimmer:

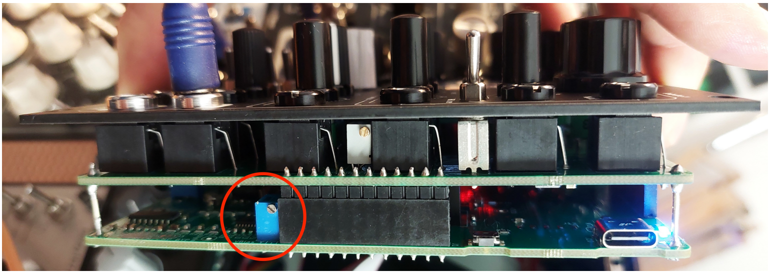


Figure 6.3: The **Y Gain** trimmer is the one *directly under the front-plate* on the right side of the module.

Preparation

1. Turn the gain X and Y knobs on the front panel fully CCW to suppress any input signal
2. Turn the feedback X and Y knobs on the front panel to around 3 o'clock.
3. Turn the lowpass knobs fully CW to open both filters.
4. Activate the feedback link switch on the front panel.

Calibration

1. Press the factory settings button.
2. With the rotation control knob, navigate to the

```
Calibration → Trim XY channel balance
```

menu entry and press the factory settings button to enter.

3. You will hear the playback of tones on the output of the module. The screen should show a red and a blue line representing the RMS volume of the tones played back. The blue line represents the X channel (left speaker) and the red line represents the Y channel (right speaker). At the bottom of a screen their average difference in signal power in decibels is shown. The goal is to reduce this average difference to less than 2dB.
4. Adjust the left Feedback knob in such a way that the blue line will be somewhat diagonal.
5. With a small slotted screwdriver, turn the **Y Gain** trimmer to adjust the decay of the red line. (See Figure 6.3) Turning CCW increases the decay, turning CW decreases the decay.
6. Adjust the trimmer until both graphs overlap as much as possible. You are controlling the shape of the red graph with the trimmer.
7. When the 2dB target is met, the routine finishes automatically.

TROUBLESHOOTING If you see strange dips in one or both RMS power measurement lines, this is due to large offset errors on the DAC outputs that are corrected in the auto-calibration step. Should you find it impossible to reach the target difference of 2dB here, move on to the auto-calibration explained in the next section by canceling the process with a press of the factory settings button.

6.2.4 *Auto-calibration*

The module implements a mathematical rotation in 2D space (made up by the signals in the analog delay lines) with the help of digitally controlled analog multipliers. To make sure this rotation is carried out with the highest possible precision the controller for the multipliers needs to be calibrated. Luckily this process can be carried out automatically by adjusting different parameters and simultaneously taking RMS measurements of the outputs.

If your module's rotation operation is not well calibrated then the decay of sounds will vary with the rotation angle. In this case a re-calibration of the controller is recommended.

Preparation

1. Turn the gain X and Y knobs on the front panel fully CCW to suppress any input signal
2. Turn the feedback X and Y knobs on the front panel to around 3 o'clock.
3. Turn the lowpass knobs fully CW to open both filters.

Calibration

1. Activate the feedback link switch on the front panel.
2. Press the factory settings button.
3. With the rotation control knob, navigate to the

   ```
   Calibration → Start auto-calibration
   ```

   menu entry and press the factory settings button to enter.
4. Let the automatic calibration procedure run. It will take approximately two minutes to finish.

The module should be fully calibrated now. To check if the module has healthy values, read the next section.

6.2.5 *Show calibration data*

This menu shows the calibration parameters for the digitally controlled rotation operation. It is here to help sanity check the calibrated values. References for sane values are shown in Figure 6.4.

Value	**min.**	**max.**
sine/cosine X/Y scale	$+0.4$	$+0.5$
sine/cosine X/Y offset	-50	$+50$

Figure 6.4: Sane ranges for automatically calibrated values.

TROUBLESHOOTING If the values of your module lie outside of these ranges, consider resetting the calibration values to the defaults as described in the next section and then running the auto-calibration routine again. If the problem persists, please post on our forum at discourse.chair.audio or contact us directly via at support@chair.audio.

6.2.6 *Edit calibration data*

This menu is a back up method for the calibration. It is not recommended to change the calibration values manually. If a previous auto-calibration failed, you can reset the calibration values to their defaults here. After resetting, you will need to run the auto-calibration again, otherwise the module will be uncalibrated and will not sound good!

To reset the calibration values to their defaults (=good initial guesses) proceed with the following steps:

1. Press the factory settings button.

2. With the rotation control knob, navigate to the “Edit calibration data” menu entry and press the factory settings button to enter.

3. Let the automatic calibration procedure run. It will take approximately two minutes to finish.

4. In this menu, navigate to the “Reset to defaults” entry.

5. To confirm, turn the rotation angle knob CW.

6. The scale values should now be reset to +0.45 and the offset values to 0.0.

You need to run the auto-calibration procedure again to re-calibrate the module.

To edit the parameters, navigate to them with the rotation angle knob, click the factory settings button, then turn the rotation angle knob CW for an increase of the parameter and CCW for a decrease.

7 FIRMWARE UPDATE

We will make firmware updates available occasionally. Firmware updates may improve performance and stability, add features or fix bugs. They may also introduce new bugs, but we still encourage you to stay up-to-date. However, since the signal path is analog, the power of the micro controller code over the sound has limitations. Curb your expectations in what a firmware update can do.

7.1 FIRMWARE VERSION

You can check which firmware you are using by pressing the factory settings button on the right side of the bottom board, next to the USB-C port. The menu entry "Show firmware version" shows the currently installed firmware version along with the build date. If your Firmware version is is smaller than v. 1.2.3, please update. See Chapter 9 for a changelog of the firmware versions.

7.2 FIRMWARE UPDATE

7.2.1 *Using STM32 Cube Programmer*

If you prefer an application with graphical user interface, you can use the STM Cube Programmer[1] to update the firmware. You will have to register for an account on STM to download this free software.

1. Press the factory settings button momentarily.

[1] https://www.st.com/en/development-tools/stm32cubeprog.html

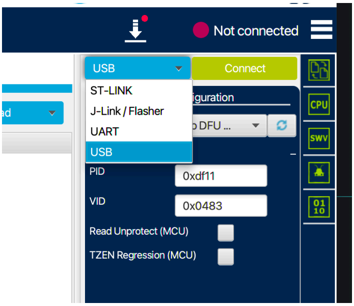
2. With the rotation control knob, navigate to the “Firmware update mode” menu entry and press the factory settings button to enter.

3. Press and hold the firmware update button for two seconds to put the module in firmware update mode.

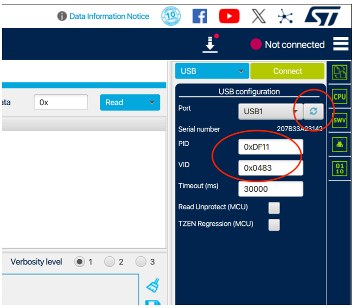
4. You are now in firmware update mode. The module will stay unresponsive until the update is complete.

5. Connect your computer with a USB-C data cable.

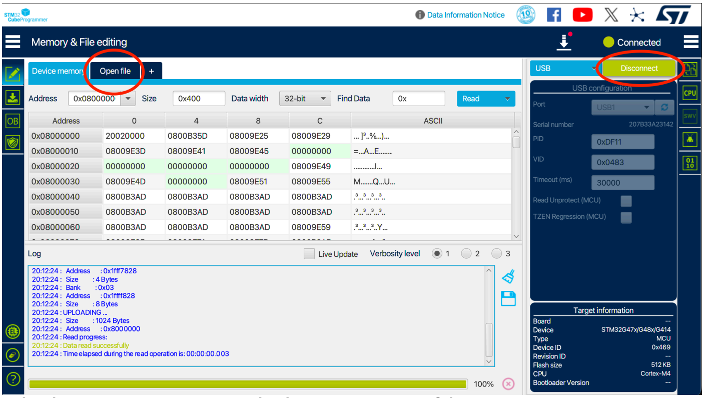
6. Open STM Cube Programmer

7. 

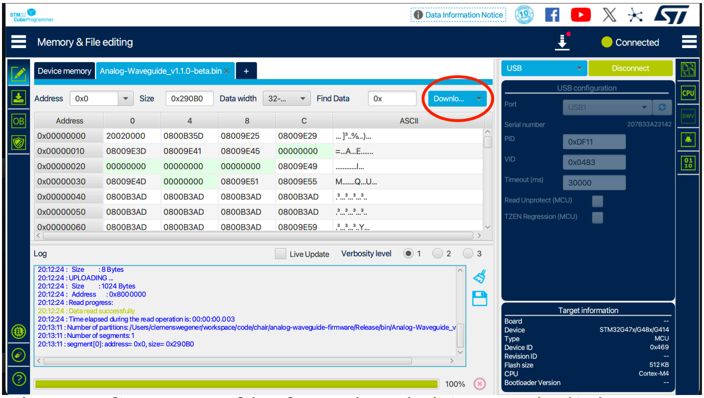
Select “USB”

8. 

Click “refresh”, choose device with PID: 0xDF11 VID: 0x0483

9. 

Click "Connect" and then "Open file"

10. 

Choose firmware file from hard drive and click "download" button

11. The download process should start, wait for it to finish
12. Unfortunately the device doesn't restart automatically and there is no option to do so from the programmer. So you have to power cycle the rack.

Congratulations, you have updated your firmware.

7.2.2 *Using command line tool DFU util*

You can use the command line tool dfu-util,[2] To update via command line follow these steps:

1. Press the factory settings button momentarily.
2. With the rotation control knob, navigate to the "Firmware update mode" menu entry and press the factory settings button to enter.
3. Press and hold the firmware update button for two seconds to put the module in firmware update mode.
4. You are now in firmware update mode. The module will stay unresponsive until the update is complete.
5. Connect your computer with a USB-C data cable.
6. Using dfu-util run:

   `sudo dfu-util -a 0 -i 0 -s 0x08000000:leave -D Analog-Waveguide-Firmware_v1.2.3.bin`.

   You may get this error: `dfu-util: Error during download get_status` which you may safely ignore.
7. When the download is finished the module will restart. You may disconnect the USB-C cable now.

Congratulations, you have updated your firmware.

7.3 TROUBLESHOOTING

7.3.1 *Calibration Data is lost*

In some cases it is possible that the calibration data stored in memory is overwritten. For example when the option "full erease" in the STM Cube Programmer is set. You need to re-calibrate. Follow the steps from subsection 6.2.4

[2] https://dfu-util.sourceforge.net

7.3.2 *Getting into Firmware Update Mode*

Power cycling the module in firmware update mode is not a problem as long as the firmware update process from a USB connection was not initiated. However, if something went wrong during the data transfer and the module does not behave as intended after the update, there is no risk of bricking the device. Simply re-initiate the update with the following steps:

1. Connect the Eurorack ribbon power cable, but don't supply power yet.
2. Connect the module via a USB-C data cable to your computer.
3. Press and hold the firmware update button.
4. Switch the power supply to the Eurorack ribbon cable on.
5. You are now in firmware update mode. You can release the firmware update button now.
6. Proceed as in step 6. in subsection 7.2 above.

If you still encounter problems please consult our forum discourse.chair.audio or contact us at support@chair.audio.

8

CLEANING AND MAINTAINANCE

Use a dust protection like a cloth or a cover to keep dust from your Eurorack system. If dust has accumulated anyway, use a manual hand air blower (the kind for photographic equipment) or alternatively a vacuum cleaner to remove the dust. Only if those contact-less methods do not suffice, use cleaning gel (usually sold for car interiors, a slimey/gooey material) to remove dust, or alternatively a soft brush and gently wipe. If you need to remove greasy stains, use a small amount of mild soap in lukewarm water on a soft fabric. Do not rub on the glass. Do not apply force. **Never use alcohol** (rubbing alcohol, Isopropanol, spiritus, etc.) to clean acrylic surfaces under tension like the watch-glass on the Analog Waveguide. If you need professional repair work, get in touch with us.

FIRMWARE VERSION HISTORY

1.2.3 Digitally controlled analog limiter with soft knee on internal feedback path minimizes input and feedback distortion. Limiter parameters settable in the menu. More flexible menu parameter editing and rendering. Use only rotation knob + push button in menu. MPU/RAM optimizations allowing for slightly bigger vertical waveform image size and higher rotation CV sample rate (81.92kHz). Higher waveform resolution thanks to higher internal sample rate. Fixes flickering peak indicators. Mute when entering menu. Screen saver with settings. Display brightness with settings. New higher order digital filters on digital interface elements resulting in lower noise on all knobs, most noticeable on rotation CV attenuverter. Lower noise and lower latency on pitch and FM CV inputs. Fixes zipper noise on rotation control knob.

1.0.5 Long press (to enter firmware update mode) is now a little bit less long (2 instead of 3 seconds).

1.0.4 Corrects endless potentiometer rotation direction and waveform display (y-axis was inverted on display, 360 degree potentiometer rotation direction was reversed, these two bugs almost canceled each other out and therefore remained undetected).

1.0.2 Accelerated display. Made button press more robust. Reordered menu items to reflect recommended order for calibration. Firmware update (bootloader) mode now accessible through menu.

1.0.0 Initial release.

MANUFACTURER

Made on planet Earth with components from all over the world. Printed circuit boards made in Europe, assembled in Germany. Conceived and designed in Weimar and Marseille.

9.1 ADDRESS

Neupert & Wegener GbR
Martersteigstr. 4
99423 Weimar
Germany

https://chair.audio

Gerzon, Michael (Dec. 1971). "Synthetic Stereo Reverberation Part One". In: *Studio Sound*, pp. 632–635. URL: https://www.worldradiohistory.com/Archive-All-Audio/Archive-Studio-Sound/70s/Studio-Sound-1971-12.pdf (cit. on p. 10).

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Stautner, John and Miller Puckette (1982). "Designing Multi-Channel Reverberators". en. In: *Computer Music Journal* 6.1, p. 52. ISSN: 01489267. DOI: 10.2307/3680358. URL: https://www.jstor.org/stable/3680358?origin=crossref (visited on 02/09/2026) (cit. on p. 10).

Wegener, Clemens and Max Neupert (July 2024). "The Aliased Complex Oscillator as a Paradigm for Analog Physical Modeling Sound Synthesis". en-US. In: *Proceedings of the 49th International Computer Music Conference 2024*. Ed. by Hee Seng Kye and Richard Dudas. Seoul: The International Computer Music Association, p. 4. ISBN: 978-0-9713192-4-0 (cit. on p. 7).