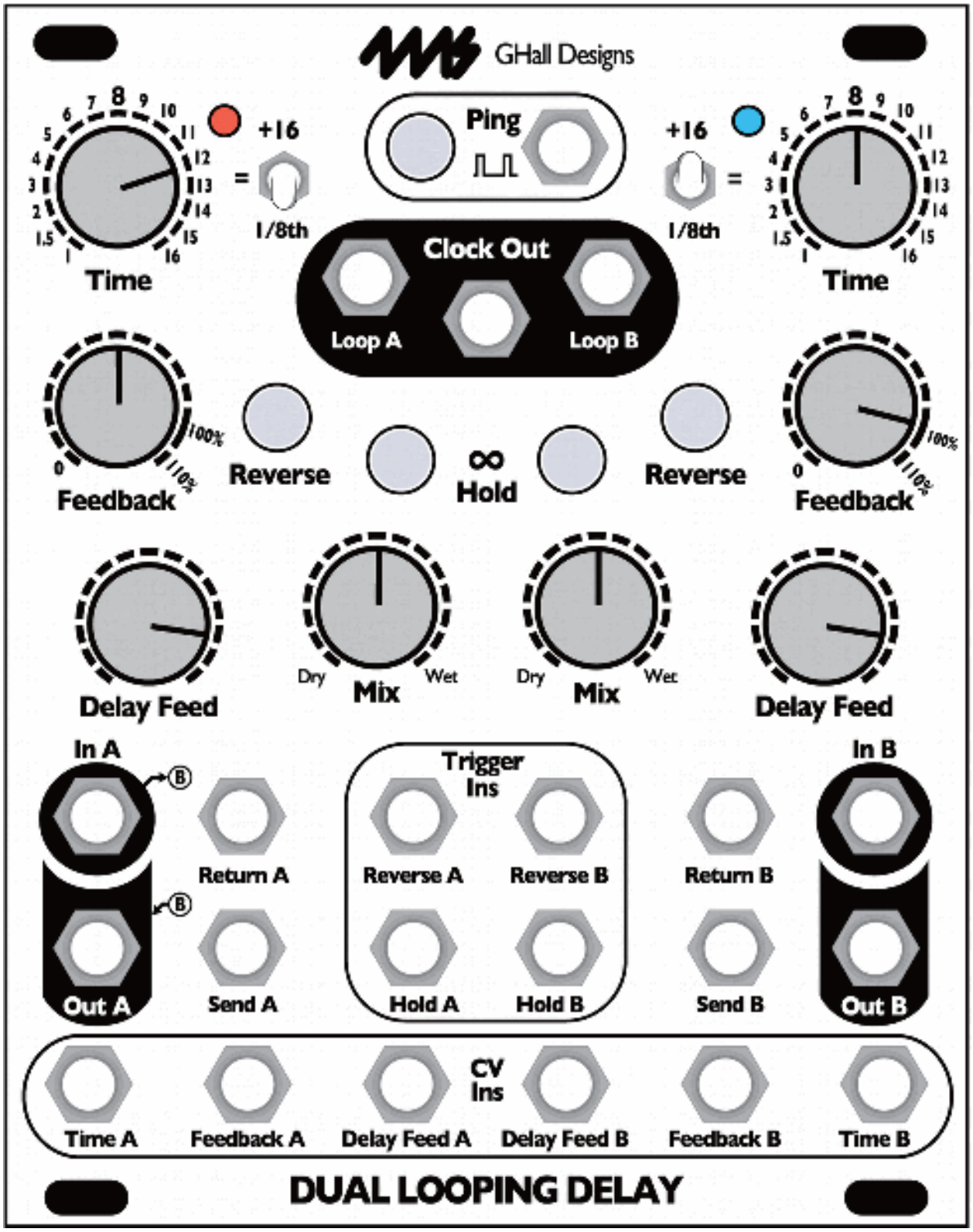
Dual Looping Delay
from 4ms Company

Eurorack Module User Manual 1.0b (2016-June-2)
For firmware version 4



The Dual Looping Delay (DLD), designed by 4ms Company and Gary Hall, is an advanced audio processor for creative synthesis. Not a tape or analog emulation, but a modern crystal-clear digital delay, the DLD combines features of delay, looping, and sample-tight synchronization for powerful and dynamic sound capture and modification. The DLD is designed to integrate seamlessly with modular timebase and sequencing devices such as the 4ms Quad Clock Distributor (QCD), etc.

- Two channels, each with nearly 3 minutes maximum delay/loop time at 48kHz/16-bit (option for 24-bit)
- Extremely quiet, low noise, low jitter design
- Stereo, cascading, or dual independent mono use
- Delay and loop time is sync-able to a clock, or a division or multiplication of a clock
- Infinite Hold mode for looping and windowing
- Reverse feature for toggling direction of playback
- Feedback ranges from 0% to 110%
- 1V/octave tracking for Resonant Delays (Karplus-Strong)
- Wet/Dry mix output, as well as dedicated Send/Return for feedback loop
- CV and trigger inputs for all major features

Basic features:

- Two independent delay/loop channels, synchronized to a common time base
- Almost 3 minutes of time per channel (2:54) in 16-bit mode, for a total of nearly 6 minutes
- 48kHz/16-bit sampling rate, option available for 24-bit sampling (results in 88 seconds delay/loop time per channel)
- Extremely quiet, low-noise and low-jitter design
- Normaled connections of input and output for flexible use in mono, stereo, cascading, or parallel mode
- Tap tempo button and clock Ping input set the timing for one "beat"
- Delay/loop time set as a number of musical beats (or fractions of beats) using the Time knob, switch, and CV jack
- Sample-accurate master clock output for perfect synchronization
- Loop clock outputs for each channel
- Time switches change range of Time knob from 1/8th notes up to 32 bars
- Digital feedback, up to 110%
- Delay Level control, independent of dry/wet signal mix
- Infinite Hold mode disables recording and plays a loop
  - Start and End points of loop can be modified in real-time for “scrubbing” or “windowing” effects
- Reverse mode plays memory contents backwards
- Resonant Delays (ala Karplus-Strong)
  - External clock input can track an external VCO up to 2kHz, and multiply it up to 16kHz
  - Time CV jack can respond at 1V/octave in *Unquantized Time Mode*
- Trigger inputs for toggling Infinite Hold and Reverse
- Send and Return on each channel for feedback with external modules
- CV jacks to control Time, Delay Feed (record level), and Feedback
- As new features are developed, firmware can be updated by playing an audio file into the DLD
  - Open-source design allows the DLD to be a versatile platform for custom firmware
- System Settings Mode for fine-tuning parameters such as 1V/oct tracking response, and gate/trigger clock output
- Buffer can be cleared with a button press
- 20HP Eurorack module

Controls and jacks:

Channels A and B are identical and separate, except for the common clock and audio normalizations.

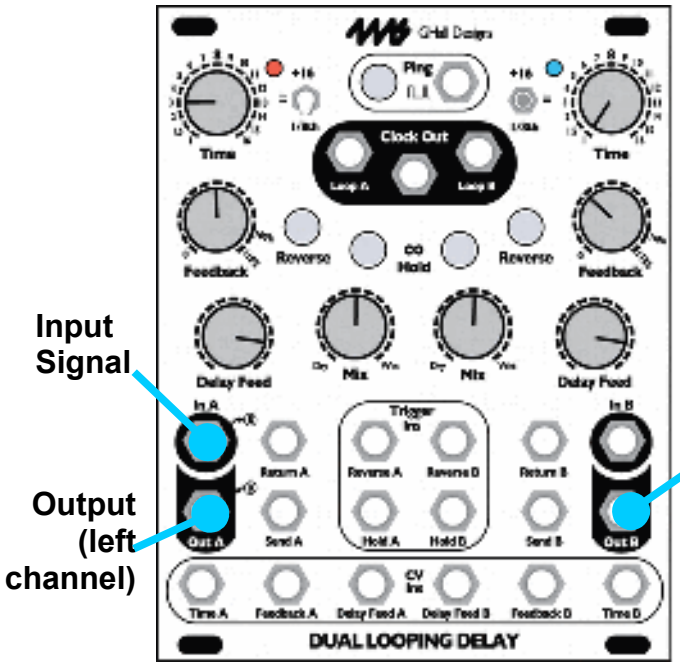
Controls

- Ping button: Tap a tempo to set the base clock = one *beat*
  - Base time can also be set by an external clock
- Time knob per channel: sets the loop time in number of *beats*. Ranges from 1 to 16
- Time switch per channel: modifies the Time knob setting
  - Down = 1/8 notes. Loop/delay time will be 1/8 to 16/8 = 2 *beat*
  - Center = normal. Loop/delay time will be 1 to 16 *beat*
  - Up = +16. Loop/delay time will be 17 to 32 *beat*
- Feedback knob per channel: sets amount of signal read from memory to be written ahead in the future
- Delay Feed knob per channel: sets amount of input signal from the In jack to be written to memory
- Mix (Dry/Wet) knob per channel: cross-fades between signal on In jack (dry) and the signal read from memory (wet)
- Reverse button per channel: reverses the direction of memory read/write
  - Reverse feature is toggled when the button is released (not when it is depressed)
- Infinite Hold button per channel: toggles between Delay and Loop mode
  - Delay mode works like a traditional echo or looping delay by recording and playing continuously
  - Loop mode disables writing, and plays a loop whose length is set by the Time parameter
  - Hold is toggled when the button is released

Jacks:

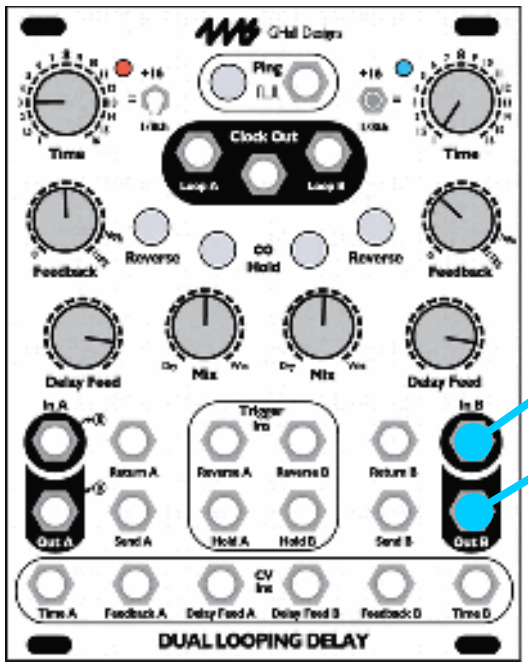
- Two audio inputs per channel
  - In (Main)
  - Return (from external feedback loop)
- Two audio outputs per channel (A/B)
  - Out (Dry/wet mix)
  - Send (Delay signal only)
- Ping input (clock sync)
- Main Clock output (stable master clock, quantized to sample rate)
- Loop clock output per channel (loop/delay time of each channel)
- Three CV input jacks per channel
  - Time (-5V to +5V)
  - Feedback (0V to +5V)
  - Delay Feed (0V to +5V) (can be re-assigned to control Mix)
  - CV jacks respond into the low audio range
- Two trigger inputs per channel (A rising edge of at least 1.6V on either jack toggles the parameter)
  - Reverse
  - Hold

Getting started: basic patches



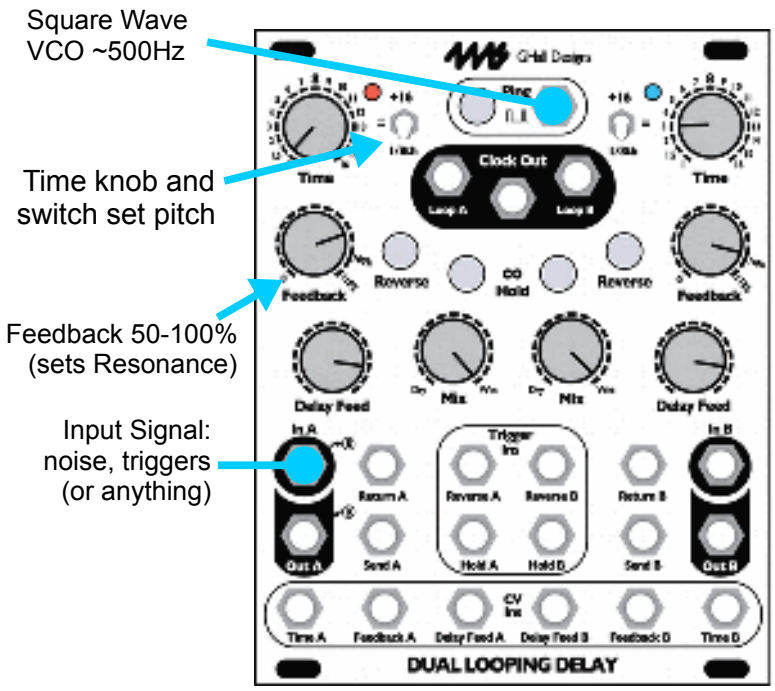
Two channel delay: Mono input, stereo output
In this patch, both delay channels will be used (Channel A and Channel B). First, patch a signal into the **In A** jack, and take the outputs from **Out A** and **Out B**. The signal you feed into **In A** will automatically be fed to **In B** (as long as you don't plug anything into the **In B** jack).

Set the knobs and switches as shown. Patch audio into the **In A** jack, perhaps a sequenced melody or a drum pattern. Monitor the output on **Out A** and **Out B** jacks. If possible, pan the two outputs in stereo. Or you can have both outputs mix together into the **Out A** jack by unplugging the **Out B** jack.



One channel delay: Mono input, mono output
You might think to run a mono signal, you should patch in and out of channel A. However, this will actually use Channel A and B in parallel (which is great fun, but can be confusing if you're expecting only one channel to be active). The reason is that Channel A's input and output normalize to Channel B, this is will actually.

To use just one channel, use Channel B as shown on the left.



Resonant Delay (ala Karplus-Strong effect)
Resonant Delays are delays with short delay time (in the audio range) and enough feedback to create a resonant sound somewhat like that of a plucked string.
The DLD supports Resonant Delays. You can tap a very fast tempo and flip the **Time switch** to 1/8, or (even better) feed a square wave VCO into the **Ping** jack. The VCO should be in the low or mid audio range, less than 2kHz (250-500Hz is ideal). Feed a signal into the **In** jack: noise works well, so do triggers (10V triggers will resonant more, 5V triggers are barely enough to trigger the resonance). Use the **Time switch** and knob to set the Pitch. **Feedback** sets the resonance (which determines the amplitude of the pitch), and **Delay Feed** effects the level and resonance as well. Make sure **Infinite Hold** is off.

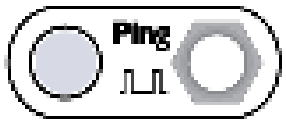
Playing Resonance with a keyboard or sequencer
Holding down **Infinite Hold** and turning the **Time** knob changes the DLD channel into *Unquantized Time Mode* (see section below for discussion). This feature is great in combination with Resonant Delays. First get a nice resonant delay sound happening. Then hold down the **Infinite Hold** button and turn the **Time** knob slowly. You should hear the pitch of the resonance change smoothly (not stepping through quantized pitches). Now plug a 1V/octave keyboard or sequencer output into the **Time CV** jack. Plug the velocity output (or an envelope output that's triggered by each step of the sequencer) into the **Feedback jack**. Turn down the **Feedback knob** so that the velocity/pressure or the envelope make the DLD create individual “notes”. You can also patch into Delay Feed to get a different effect. Since there are two DLD channels, you can use it as a duophonic melodic instrument!

Concepts

On the back page of this manual is a block diagram of the control and signal paths for one channel. The metaphor is that each DLD channel is a tape loop, with a separate read and write head, and the ability to move each heads to a new location. The tape is assumed to move at a constant speed (constant sample rate), but can be reversed in direction. The magic ability to change the length of the tape loop is also assumed in this metaphor (Infinite Hold mode).

Ping: Setting the beat

The DLD requires a base time, which is referred to as a "*beat*" in this manual. Each channel can have a different delay or loop time, but both channels' delay/loop time will be mathematically related to the length of one beat. If you are familiar with other 4ms modules, you may be familiar with the concept of "Ping". In the DLD, the *beat* is the Ping.



There are several ways to establish the beat:

- Tap the **Ping button** twice. One beat will equal the time between your taps.
- Patch an external clock into the **Ping jack**. One beat will equal the time between the last two pulses.
- Just turn on the DLD and do nothing more! The DLD automatically boots up at around 150BPM

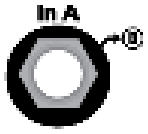
- *There is a fourth way of setting the timing, which is by enabling Clock Bus support on your DLD and using a Clock Bus master such as the QCD to send the clock over the power distribution board. See the* Clock Bus Jumper *section.*

The DLD's clock is extremely stable and jitter-free. It's highly recommended to use the internal DLD clock (tap clock) if possible, because it's quantized to the sample clock and has less jitter than most common commercially available clock sources. Using the DLD as a master clock will provide the tightest timing possible in a patch with the DLD. However, if it's not possible to use the DLD as a master clock, you can sync to an external clock by patching it into the Ping jack. The DLD will quantize the external clock to its own sample rate and output quantized clocks on the two Loop jacks. If there is any jitter or drift in the external clock, the DLD will track and follow these discontinuities, which can result in interesting (or noisy! artifacts).

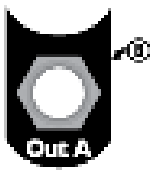
Channels

There are two channels: A and B. Each channel has a separate audio path, controls, and CV/trigger jacks. The channels are identical — except for the fact that the inputs and outputs automatically connect together if nothing is patched into Channel B (see below). Both channels share a common time base, or *beat*, which is set by the Ping button or jack. Each channel can multiply, divide, or add to the base time by using the Time knob, switch, and CV. Anytime the base time is changed (by tapping the Ping button, or changing the external clock), both channels will instantly change tempo while maintaining the same ratio to each other.

Channel normalizations



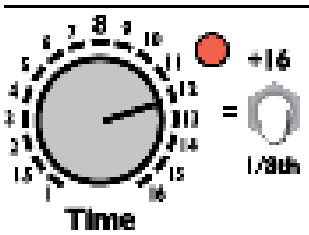
The main **In A** jack feeds into the switch tab of the main **In B** jack. Therefore, plugging into the **In A** jack will run the audio signal to Channel B as well. If you plug a cable into the **In B** jack, then the normalization will be disabled. *The "-->B" graphic on the panel near the In A jack reminds you of this normalization.*



Similarly, The switch tab of the **Out B** jack feeds into the signal tab of the **Out A** jack. Therefore, if nothing is plugged into the **Out B** jack, then the output from channel B will be mixed with the output of channel A and appear on the **Out A** jack. *The "<--B" graphic on the panel near the **Out A** jack reminds you of this normalization.*

To use the DLD as a two-channel delay in parallel, plug into the **In A** and **Out A** jacks. Both delay channels will operate in parallel, mixing together into the **Out A** jack.
To use the DLD as a one channel delay, plug into the **In B** and **Out B** jacks. Channel A will not be used. Alternatively, patch into **In A** and **Out A**, and patch a dummy cable into **Out B** to disable the normalization.
To use the DLD as a two channel series (cascading) delay, plug into the **In A** jack, and patch from **Out A** to **In B**, and then take the output from **Out B**. Alternatively you can patch from **Send A** to **In B** or **Return B**. See the *Using Send and Return* discussion in the *Applications* section below.

Time



The Time parameter sets the length of the loop, or amount of delay time. This period of time is relative to a *beat*, which is set by tapping the Ping button or patching an external clock into the Ping jack.
The Time knob, CV jack, and switch perform simple arithmetic on the beat time to produce a delay/loop time.

- **Time Switch** center (=): the **Time knob** directly sets the number of beats for the loop or delay time. *Example: if you turn Time to "5", the channel will make an echo (or loop) every 5 beats.*

- **Time Switch** up (+16): 16 is added to the whatever the Time knob shows.
  Example: if the Time knob is pointing to 5, the channel's delay or loop time will be 5 + 16 = 21 beats.

- **Time Switch** down (1/8): the **Time knob** is divided by 8, making the delay/loop time 8 times as fast (eighth notes).
  Example: Time knob is at 5, the channel will echo in 5/8th notes, or eight loops for every five beats.

Using CV will modify the **Time knob's** setting by multiplying or dividing the knob's value. Positive CV makes the delay time longer, negative CV makes it shorter.

In math:

Time switch up: `Delay time = ([Knob x CV] + 16) x beat time`
Time switch center: `Delay time =  (Knob x CV)  x beat time`
Time switch up: `Delay time = ([Knob x CV] / 8)  x beat time`

```
Knob ranges from 1 to 16
CV ranges from 1 to 16 if positive voltage is applied to the Time CV jack (0 to +5V)
CV ranges from 1 to 1/16 (0.0625) if negative voltage is applied to the Time CV jack (0 to -5V)
```
Note that with 0V on the CV jack (nothing plugged in), CV=1, so it has no effect on the knob's setting.

For example, tap a tempo of 0.5 second. This makes one *beat* be ½ second (120BPM).
Set the **Time knob** to 8, and the **Time switch** to center. If there is no CV plugged in, the delay time will be 8 beats, or 4 seconds.
Now flip the **Time switch** down to “1/8”, the delay time will be eight 1/8th notes, or ½ second.
Now turn the **Time knob** down to “2”, and the delay time will be two 1/8th notes or 0.125 seconds.
Now flip the **Time switch** up to “+16” and the delay time will be 18 beats, or 9 seconds.
Now apply some CV, the Time will get slower as you apply positive CV, and faster as you apply negative CV.

Setting two channels to different delay loop periods that are multiples of the same clock can result in very interesting phasing arrangements! =

Loop start and stop points
When **Infinite Hold** is on, changing **Time** will change the loop start point in order to make the loop the new length. If you hold down the **Reverse** button while changing Time, you will change the loop end point. You also can use this to change a loop that's, for example, 4 beats long into one that's 5 beats long, in two different ways (adding a beat to the end, or to the beginning). There are many creative possibilities to be explored with adding and subtracting from the beginning and/or end of a loop. One idea is to “inchworm” up and down memory. If you want to move more quickly than an “inchworm”, try windowing: see the section below on *Windowing.*

Delay Feed



Delay Feed is the amount of signal from the In jack that's recorded in memory. You can think of it as “Record Level” or “Input Level”, but there is one important detail to remember:
Delay Feed doesn't effect the *Dry* signal (which is what's present on the **Out** jack when the **Mix** knob is set to *Dry*). It just effects how much signal is laid down or recorded. This detail is intentional because when doing sound-on-sound techniques, you can play a sound continuously in the *Dry* channel while modulating **Delay Feed** to fade in portions of the signal onto the loop.
The knob and CV jack are added together and have a maximum combined value of 100%. Negative voltage on the CV jack is ignored. Note that in **Inf Hold** mode, **Delay Feed** has no effect since there is no recording.

Feedback



Also known as “regeneration”, “echo amount”, and other names, feedback is a common feature on delay effects. Feedback is the amount of signal read from memory that's written back to memory in a different location. This new location will be read again in the future, and written back again at yet another location in the future. Take a look at the block diagram on the back page to see how **Feedback** fits into the signal flow.
On the DLD, **Feedback** ranges from 0 to 110%.
With no feedback (knob at 0), and you will hear one echo. As you turn **Feedback** up, you will gradually hear more and more echoes (the amount of time before the echoes go to silence will increase).
With **Feedback** at 100%, the signal read from memory is written back to memory unaltered. Thus, the echoes will never fade out — an infinite loop. But if you input a signal on the **In** or **Return** jacks, they will add to the feedback signal, which can result in the sound gradually getting louder and louder as sound is added but never reduced. The knob's range has been modified to create a large area which is exactly 100%.

With **Feedback** at 110%, the signal read from memory is boosted before written. This makes the echoes louder and louder each time. For short delay times, this is a well-known “blooming echo” effect used in various styles of music including Dub Reggae.

Note that in **Infinite Hold** mode, **Feedback** has no effect. However, see the section below on “Windowing” for the special usage of the **Feedback** knob and jack while holding down the **Infinite Hold** button.

The knob and CV jack are added together, and are have a maximum combined value of 110%. Negative voltage on the CV jack is ignored. If the knob is set to 0%, then applying a voltage of 5V or greater will cause a **Feedback** amount of exactly 100%. This is to allow for easy CV control of accessing 100% **Feedback** (any voltage greater than 5V = 100%, if the knob is set to 0)

Mix



Mix controls the blend between the Dry signal and the Wet signal on the **Out** jack. It does not effect the **Send** jack. The Dry signal is taken directly from the **In** jack (and not the **Return** jack) before the **Delay Feed** parameter has any effect. *See the block diagram on the back page of the manual.*

Mix does not have a CV jack, but starting in firmware v4, there is an option to make the **Delay Feed** CV jack control **Mix** instead (see *System Settings* section).

Infinite Hold



The DLD can operate as a delay/echo or it can play a loop. While you can make the DLD play loops by setting **Feedback** near or at 100%, each channel has the ability to immediately jump in and out of a special mode called **Infinite Hold** mode by pressing a button or receiving a trigger.

Pressing the **Infinite Hold** button toggles the state of **Infinite Hold** when the button is *released*. The light on the button indicates whether the mode is on or off.

Normally (**Infinite Hold** off), the channel records audio and plays it back after the delay time has passed. Every sound the channel makes is continuously recorded into memory.

When **Infinite Hold** is on, the channel stops recording and only plays what's already recorded in memory, cycling through a loop. The size of the loop is set by the **Time** parameter. When you activate **Infinite Hold** mode, the DLD will start looping what you *just recorded.* The loop is defined by start and end points in memory. Wherever the DLD is reading from memory will become the start of the loop, and where it was writing to memory will become the end of the loop. You can get a sense of where the read and write positions are by listening to the Wet and Dry signals, respectively—the Wet signal is what's being read from memory, and the Dry signal is more or less what's being written to memory (especially if **Delay Feed** is up and **Feedback** is down). So

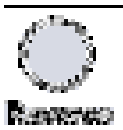
What's the difference between turning on Infinite Hold versus setting Feedback to 100% and Delay Feed to 0%?

Since setting **Feedback** to 100% (or close to 100%) and **Delay Feed** to 0% creates an infinite loop, the DLD acts similarly to how it does when **Infinite Hold** is on. However, there are a few important differences:

When **Infinite Hold** is on, you can adjust the Time parameter from one setting to another, and then back to original setting and it will sound exactly the same as it did originally. This is not the case with **Infinite Hold** off and **Feedback** at 100%: changing **Time** to a faster setting will actually record the new shorter echoes into memory. Then when you turn back to the original slower **Time** setting, the DLD will be reading the shorter echoes and will echo these echoes in a longer echo. Therefore, even though the Time setting is the same as it was originally, it will sound different! The reason is that with **Infinite Hold** on, the DLD does not record anything, but with **Infinite Hold** off and **Feedback** at 100%, it records all the echoes.

Another difference is that when **Infinite Hold** is on, the **Feedback** knob has a special function: windowing. Hold down **Infinite Hold** while turning **Feedback** to change the start and stop points of the loop (see *Windowing* section in this manual). There is no windowing with **Feedback** at 100% and **Delay Feed** at 0%, but there is the ability to fade in new sounds (layers) by fading up the **Delay Feed** knob, or by applying CV (perhaps an envelope?) to the **Delay Feed CV** jack. Or you can create blank spaces by turning **Feedback** down momentarily. The loop is more dynamic and mutable, versus when **Infinite Hold** is on, the loop is more static and immutable.

Reverse



Reverse reverses the direction that memory is read and written.

Pressing the **Reverse** button toggles **Reverse** on/off when the button is *released*. If you turn a knob while **Reverse** is held down, the **Reverse** action will be cancelled (this is because holding a button while turning a knob is reserved for special features such as Windowing)

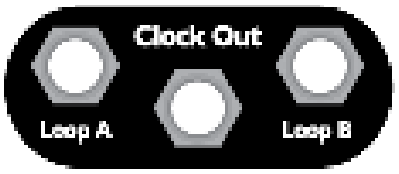
Applying a trigger on the **Reverse** jack will toggle **Reverse** on the rising edge of the trigger.

Reverse has slightly different effects depending on if you are in **Infinite Hold** mode or not:

In normal mode (**Infinite Hold** off), toggling **Reverse** causes playback and record to reverse direction. The read and write positions are also swapped. This means that any audio that's already been recorded will be played backwards, but any new audio will be played forwards (since it's recorded backwards and played backwards, it comes out sounding forwards). Note that when **Time** is very fast, **Reverse** will seem to have little effect because it only reverses what's already recorded (which has a short duration when Time is fast). *Note: In firmware version 4 and later, toggling Reverse crossfades between the old and new directions, but in earlier firmware versions it can cause an audible click. Upgrade if you have this issue.*

In **Infinite Hold** mode, toggling **Reverse** plays the loop backwards. The loop start point (as indicated by the Loop clock output jack) will become the moment that the loop is pressed. So by reversing forward and backwards you can shift the loop's phase relative to the master clock of the other channel's loop. This is especially useful if you are using the Loop Out jacks to trigger something else in your patch (such as an envelope and VCA).

Clock Outputs



There are three clock output jacks:

- **Loop A** sends a clock that's in sync with the delay/loop time of channel A
- **Loop B** sends a clock that's in sync with the delay/loop time of channel B
- **Clock Out** (middle jack) sends a clock that's in sync with the master Ping clock (whether you tapped in a tempo, or are using an external clock).

The red light near **Time A** shows the speed of the **Loop A** clock, and the blue light near **Time B** shows the speed of the **Loop B** clock. The jacks may output triggers, but the LEDs show equal on and off times, which is merely for aesthetic reasons.

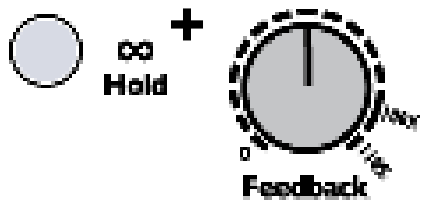
All the clock outputs are quantized to the sample-rate. This insures they are jitter-free (assuming your external clock has a low degree of jitter). The DLD clocks are some of the lowest jitter clocks available using Eurorack modules and should be used whenever possible as a master clock.

The high voltage of the clock outputs is 8V. The pulse width is 22ms when in Trigger mode (default), as long as the clock period allows for a 22ms pulse. At fast clock speeds (audio rate), the jacks output square waves. There is a System Setting mode that allows you to change the clock outputs to gates rather than triggers (see *System Settings Mode* section).

In **Infinite Hold** mode, the clock goes high when the loop starts. This can be used to trigger an envelope that VCA's the loop, for example.

Special Features

Windowing aka Scrolling aka Scrubbing



When **Infinite Hold** is on, the **Feedback** and **Delay Feed** parameters have no meaning since they are effectively at 100% and 0%. However, the **Feedback** knob and CV jack have been given a special purpose when **Infinite Hold** is on: Windowing, also known as scrolling or scrubbing.
To use this feature, hold down the **Infinite Hold** button while turning **Feedback.** This will cause the start and stop points of the loop to scroll forward or backwards, depending on which way you turn **Feedback**. The amount they scroll is determined by the amount you turn **Feedback**: one full turn of the knob equals one loop length. So if you have a 2 second loop, then turning **Feedback** from 0% to 100% will shift the loop forward by two seconds. Turning **Feedback** from 100% to 50% will scroll the loop backwards by one second.
*Note: Windowing only works when **Infinite Hold** mode is on.*

Windowing Tutorial
Let's try it. First get set up: Set **Time** so you have a loop length that's about a second or so. Keep **Feedback** low to keep things simple for now. Play some sounds for 10-20 seconds and then turn on **Infinite Hold** to lock them in.

Listen to your loop, pay attention to what sounds the loop starts with, and what sounds it plays right before it repeats. These are your loop start and stop points. Now, we're going to change those start and stop points by Windowing:

1. Turn **Feedback** knob all the way up (the sound will not change).
2. Hold down **Infinite Hold** while you turn **Feedback** to 0%.

You just shifted the loop backwards by one whole loop length. Let it play for a bit as you listen to the new start and stop points. Hear it? The loop is the same length (same timing/tempo), but now it will be playing the sounds you recorded one second earlier. Play with this some more: Press and hold **Infinite Hold** again and turn **Feedback** back half a turn. Hear how the loop now starts in the middle? Remember that turning **Feedback** has no effect unless you're holding down **Infinite Hold**. This is critical for the next tip:

Tip #1: If you want to scroll more than one loop length, do this maneuver:

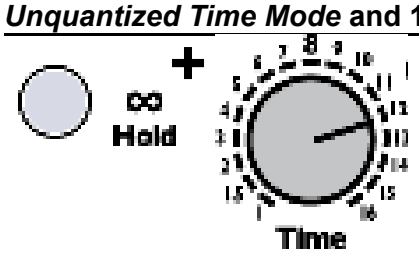
1. Turn **Feedback** to 100%.
2. Depress **Infinite Hold** and turn **Feedback** to 0%.
3. Release **Infinite Hold**.
4. Repeat as needed (turn Feedback to 100%, then press Inf and turn Feedback back to 0, release Inf...)

Tip #2: If you want to scroll back very far even more quickly than Tip #1, change the Time parameter to very long (perhaps flip the time switch up to +16). Since turning **Feedback** + **Infinite Hold** scrolls by *one loop size*, making the loop size enormous lets you scroll by enormous amounts with just one knob twist! You can scroll back a maximum of 2 minutes 54 seconds.

Tip #3: Set Time to a very short period and window around a loop with CV for a sort of granular effect.

Using CV with Windowing
The Feedback CV jack also allows you to window using external CV control. To enable the CV jack, you must first manually hold down **Infinite Hold** and turn **Feedback**, even just a small amount. The **Feedback** CV jack will now control the window. If you turn the **Feedback** knob at any time without holding down **Infinite Hold**, the **Feedback** CV jack will no longer control the window.

Unquantized Time Mode **and 1V/oct CV**



Both channels of the DLD are always linked to a common time base. Normally the **Time** knob and CV are quantized to integer amounts (1-16), and simple fractions (1 – 1/16). This is called *Quantized Time Mode*, and is the default mode. It's possible to change to *Unquantized Time Mode*, where the knob and CV provide continuous control of the Time parameter (not quantized to integer or simple fractional amounts)
To change modes is simple:

- *Unquantized Time Mode:* Turn **Time** knob *while* holding down the **Infinite Hold** button
- *Quantized Time Mode:* Turn **Time** knob *without* holding down the **Infinite Hold** button

In *Unquantized* mode, the **Time** knob behaves as usual, except it does not snap to the whole numbers between 1 and 16. So you can sweep a slowly changing tempo, or set an exact tempo in between two integer amounts. To adjust the **Time** knob in *Unquantized* mode, hold down the **Infinite Hold** button down while you turn the **Time** knob.

The **Time CV** jack behaves differently in *Unquantized* mode: It responds over a 1V/octave curve for positive CV (5 octave range). Applying up to +5V will multiply the **Time** knob's setting in an exponential curve relative to the voltage. That is, for every additional volt on the CV jack, the Time period will halve. This is the opposite response as *Quantized* mode, where additional voltage makes the period increase. The 1V/octave response in *Unquantized* mode is very useful for resonant delays.
Note that if the Time switch is up, the 1V/oct response will be altered by the addition of the extra 16 bars. For a true 1V/oct response, keep the Time switch centered or down. You also can adjust the exact 1V/oct tracking in System Settings.

Buffer Clear

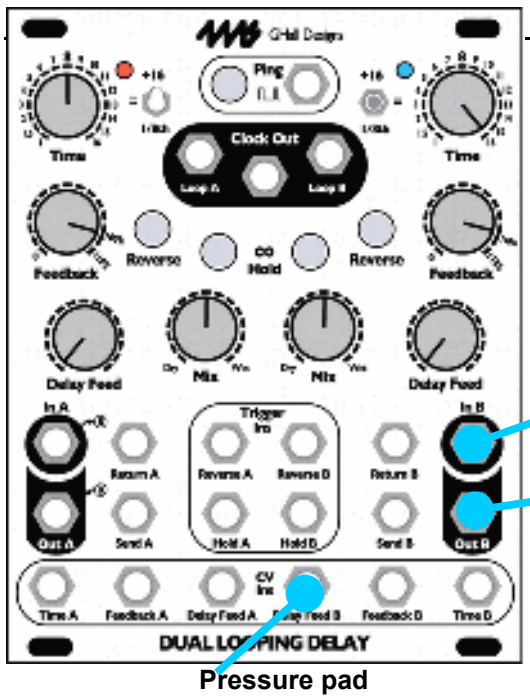


When using the DLD with Feedback up and long loop times, it can take a while for the sound to die out after the input signal is muted. This is, of course, a very nice and useful effect, but if you find yourself needing to clear the memory quickly you can do so by holding down **Reverse** and **Infinite Hold** for two seconds. The sound will cut out briefly while the memory is cleared and the lights will flash. Let go of the buttons and the buffer for that channel will be clear. Each channel has its own buffer, and can be cleared independently of the other.

Why do I hear something I played a long time ago? Notes on the nature of Looping Delays

When **Infinite Hold** is off, the Wet signal (i.e. whatever's being output from the **Send** jack) is continuously recorded into memory, going back for about three minutes (2:54 to be exact, which is the amount of each channel's memory). This means that the results of every knob you twist and every bit of CV you input is being recorded into memory*. Let's say you're playing a melody into the DLD with a nice rhythmic echo. If, for example, you change the **Time** parameter to make triplets for a moment, then switch back to eighth notes, then maybe bring **Feedback** up to make a “bloom”, then pull down **Delay Feed** to cut the audio out, etc... all of this will be recorded into memory. If you think in terms of the metaphor of a tape loop, then it's as if your whole session is being recorded onto the loop.
Normally you won't have to think about this: the DLD operates “as you think it should”, overwriting whatever it recorded 2:54 ago with what you're playing now. But in some special circumstances, you can access this old memory: Windowing large chunks in *Infinite Hold Mode* is the primary method whereby you might hear some sounds that were recorded onto the “tape loop” several minutes ago. This can be surprising! But it also can be very useful.

Applications



Pressure pad
(to fade a new sound into the loop)

Sound on Sound Looping #1

This is a great patch for long loops. Set a long gentle clock speed (at least 1 second between taps). The diagram on the left shows how to patch this for Channel B-- once you master it, try patching both channels A and B and using the Out A mix jack for dual loops!

Turn **Time B** knob to 16 (or something slow). Turn **Feedback B** to 100%. Turn **Delay Feed B** to 0%. Turn **Mix** to 50% (or to taste). Now run audio into **In B**. Try running a melody or a percussive sequence or an evolving drone, perhaps.

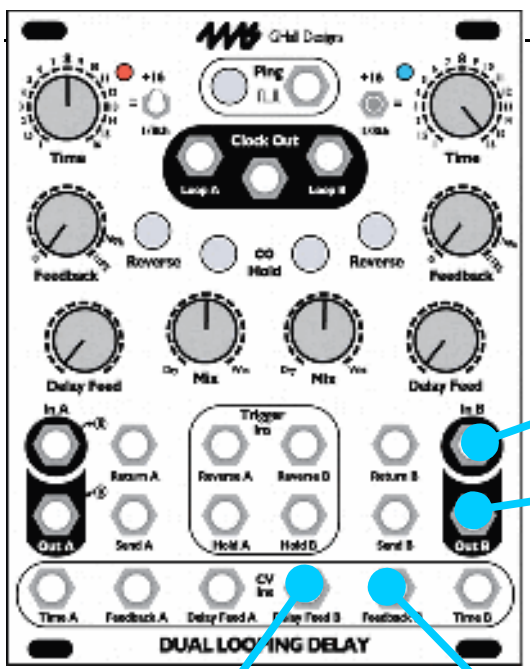
Here's the trick: Patch a manual CV source into the **Delay Feed B CV** jack. It can be a Pressure Points, or an FSR pressure output, or perhaps an envelope output that's triggered manually. When you press on the Pressure Points or FSR, you will bring up the **Delay Feed** parameter which causes audio to be recorded onto the loop.

Input Signal

Output

Since **Feedback** is at 100%, the loop is infinite and everything you lay down will remain (but take care to keep your levels not too hot or else you will eventually get clipping)

A variation is to use the **Send B** jack for the loop output. Then set the **Mix** knob to center and use the **Out B** jack to monitor the incoming audio



Pressure pad to record

Inverted and offset pressure pad to clear

Sound on Sound Looping #2

In this variation, you can use one pressure pad to bring in sound, and another to clear the loop, or fade it down. This requires a module that can invert and offset a CV signal, such as the 4ms SISM, the Makenoise MATHS, or any number of CV utility modules.
Patch the previous patch. Then turn **Feedback B** down to 0%. Patch a second manual CV source into the 4ms SISM (or other utility module). Turn the SISM's Scale knob all the way down to – (invert), and turn the SISM's Shift knob up to about 2:00. Patch the SISM channel's output to the DLD's **Feedback B CV jack.**
Now you can press on the “record” pad like in the previous patch to lay down new material. But now you can also press on the “clear” pad to fade out material from the loop. You can even press on both pads at the same time to replace loop material with new material.

Input Signal

Output

What's happening in this patch is that the SISM is turning the 0V to 5V (or 8V or whatever) signal from the pad module into a 5V to 0V signal. So, the SISM will output around 5V if you are not pressing on the manual pressure pad (no CV signal). This means the DLD's **Feedback** will be 100%. As you press on the pad and increase the voltage from the pad module, the SISM will decrease its output voltage, which decreases the **Feedback** parameter. When Feedback is low, material from the loop fades away.

Note that if your loop is not looping at 100%, then the SISM's Shift knob needs to be turned up a bit. This insures at least 5V is coming out of SISM when nothing is being input.
Hint: Turn **Infinite Hold** on for one channel, to create a “base layer”, and use the other channel as a Sound on Sound loop to make a dynamic “middle layer”. Use the Dry signal as a “top layer”.

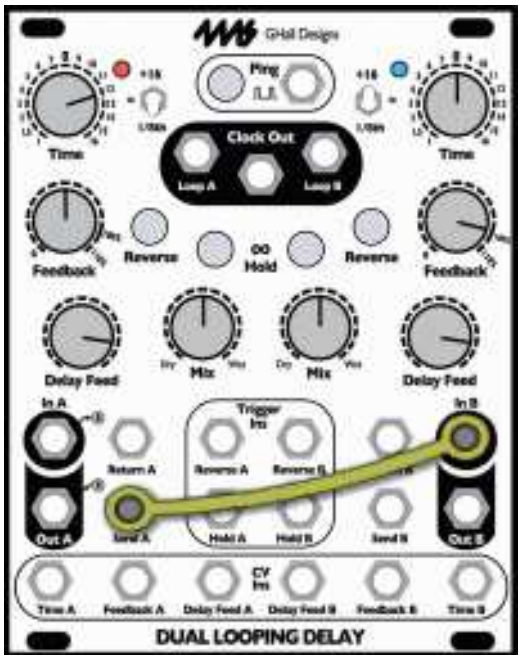
*except for the Wet/Dry mix as set by the Mix knob. Note that the CV itself is not being recorded, rather the audio that results from the CV.

Signal Routing: Using Send and Return

The internal signal routing of the DLD and its normal connections provide a good set of capabilities of many uses. Variations on the internal routing can be accomplished with simple patches at the front panel as well.
The **Send** and **Return** jacks of each of the two delay channels are simple in function, but offer a wealth of possibilities for creative patching.

- **Send A and Send B:** The two **Send** outputs simply carry the delayed signal, unmixed with the input. It's the same as the main **Out** jacks if the **Mix** control is set fully clockwise — except that it's always that way!

- **Return A and Return B:** The Return inputs are summed with the corresponding main **In** (A or B). However, they do not appear in the "Dry" signal and are not attenuated by the **Delay Feed** knob and CV. This provides a rudimentary input mix capability and a flexible signal routing for both external patching and variations on the unit's basic signal paths.

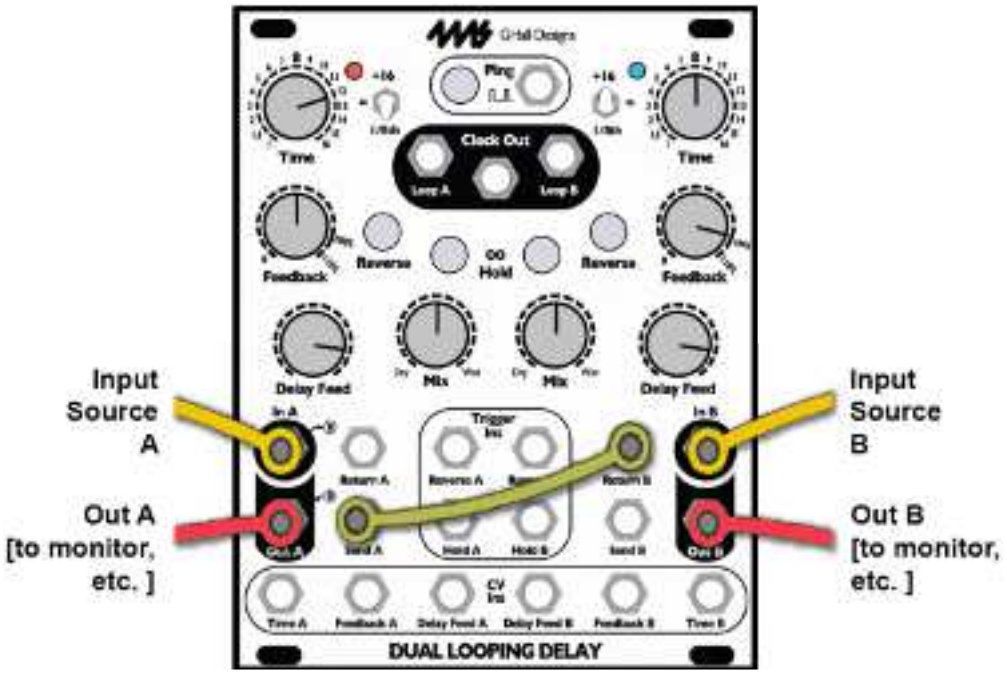


Cascading Delay

If you want to route the delays in series (cascade) with the output of Channel A feeding the input of delay B, simply connect a single patch cord between **Send A** and **In B** as shown on the left. Controls of Channels A and B can be set independently for a variety of effects. You can also monitor the **Out A** jack and use the **Mix A** knob to blend between the dry signal and Channel A's delay.

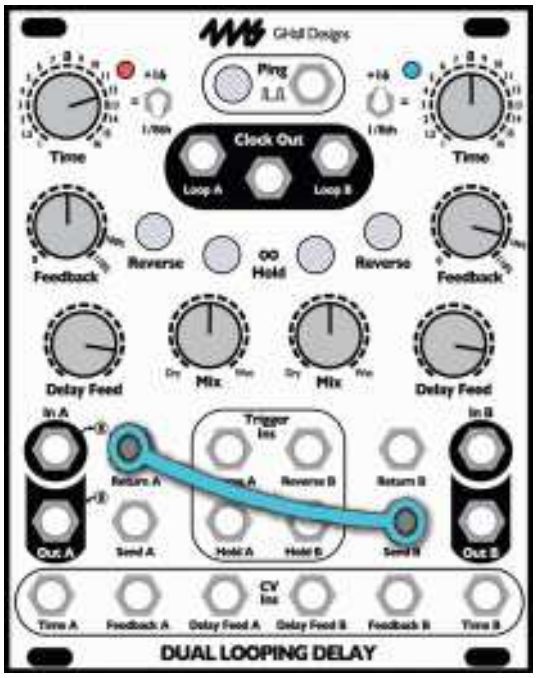
Reverb-y echo room

Patch as shown on the left. Try setting one channel's **Time** to a fast delay such as "1" with the **Time switch** flipped to 1/8. Set the other **Time** to a slow delay such as "8". Set both channel's feedback amounts to around 75%-90%. Take the output from **Out B**. Tap a very fast tempo (as fast as your finger will go!)
While this is not a true reverb effect, the combination of two delay speeds plus high feedback results in a similar effect. Try modulating the **Time A** or **B** with CV, and try pushing **Feedback** right to the edge!



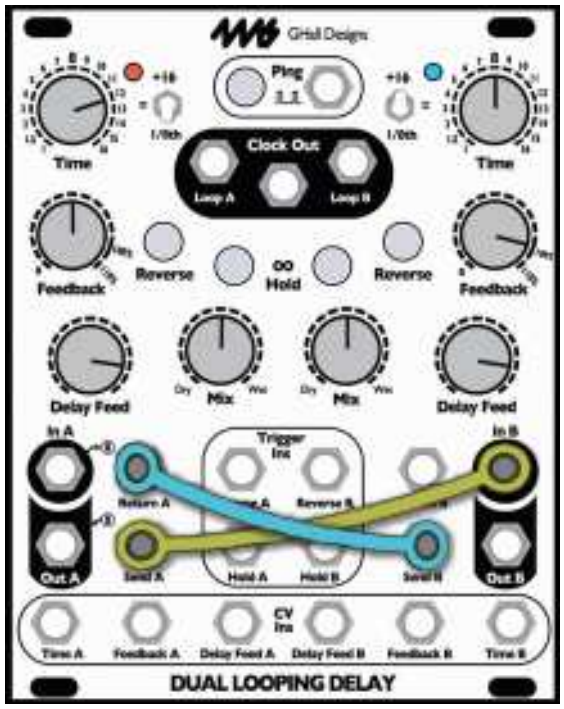
Dual-input Cascading Delay

A variant on the previous patch is to connect **Send A** into **Return B**. The principal advantage of this is that you can input another signal into the main **In B** while also taking feed from a separate source in **In A**. In the case that no plug is inserted in **In B**, all you will be doing is adding an additional portion of Channel A out in the input mix of Channel B. See the patch above (the input and output signals are made explicit). In this configuration, both channels will process **In A,** while **In B** is processed only by Channel B.

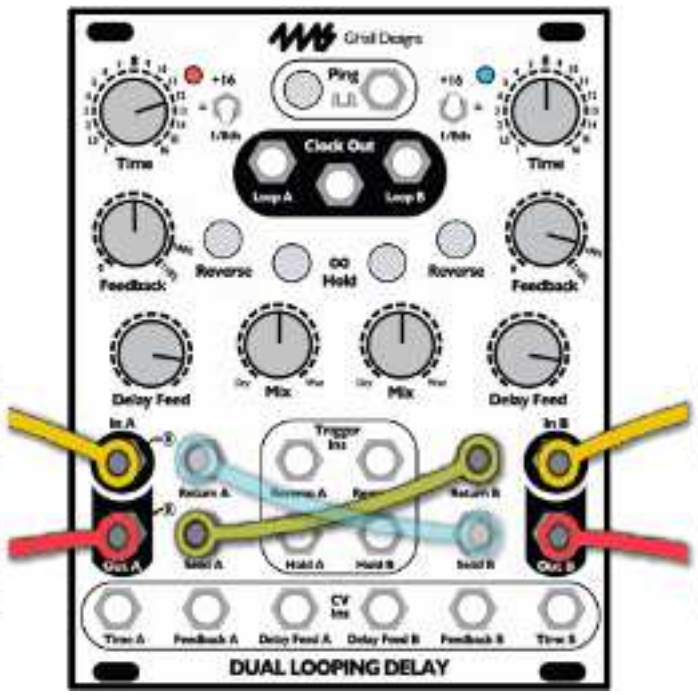


Feedback from Channel B to A

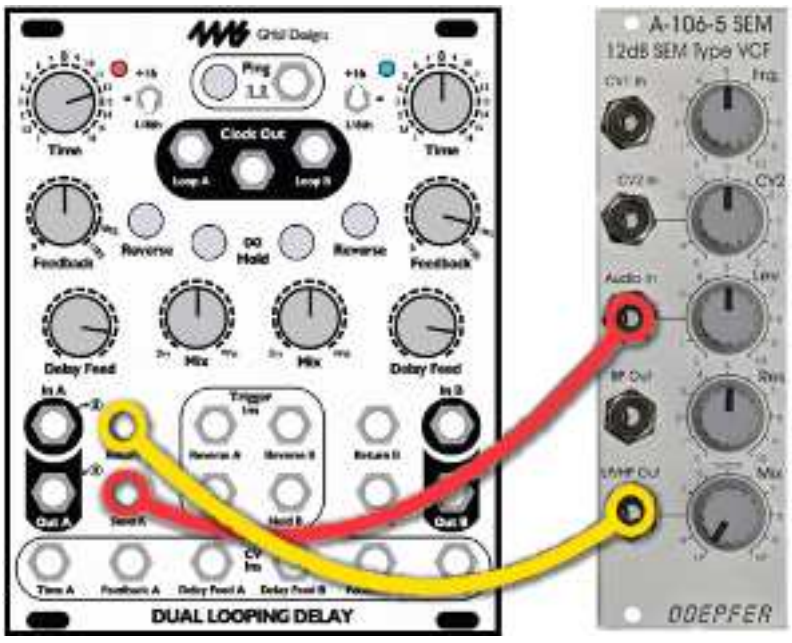
You can incorporate feedback from Channel B back to A and thus through the whole chain by putting a patch cord between **Send B** and **Return A**, as shown on the right. This configuration is generally stable, but high levels of feedback on both delays at the same time could run out of control.



Cascading Delay + Cross Feedback
This combination gets rather interesting. The settings of **Delay Level** and **Feedback** of Channel A and Channel B will interact strongly, and it is all too easy to encounter runaway feedback. For easy control from the front panel, adjust **Delay Feed** of Channel B for the desired total feedback. Alternatively, you can patch an external attenuator between **Send B** and **Return A**.



You also can combine the previous patch's “cross-feedback” with separate audio sources. In this figure, the final patch of the cross-feed is ghosted to show that it may be a path of runaway feedback if you’re not careful. But with appropriate settings of the **Delay Feed** and **Feedback** controls, this patch provides a very nice range of rhythmic bouncing and spatial effects.



Feedback with External Processing
The classic use of Send and Return ports in a time delay processor is to provide a “side-chain” for modifying the delayed signal before it is returned to the input mix.

Probably the most classic application of this kind of side-chain processing is a simple low-pass filter in the feedback path. This causes succeeding repeats to soften, with less high-frequency content on each pass. This is similar to the behavior of echoes in an acoustic environment, and for this reason a low-pass filter is included in the internal feedback path of many conventional delay processors.

In patches involving externally processed feedback, the gain of the processing chain is always a concern. It may be too little, or it may be too much! In the case of filters in particular, the gain may be suitable overall, but not at the frequencies you want to recirculate. If the filter is resonant, runaway feedback may occur even when the overall gain is low.

Audio Bootloader

The DLD contains a bootloader that is used to update the firmware by playing an audio file the **In B** jack on the right side of the module. Although using the **In A** jack may work some times, it's recommended to use the **In B** jack for more reliable operation.
Firmware audio files can be downloaded at http://4mscompany.com/dld.php

1. To enter bootloader mode, power off the DLD and connect a computer or smart phone audio output to the **In B** jack. Either a stereo or mono cable is fine. Connect the **Out B** jack to an amp/speakers so you can listen.
   Remove your phone case, it may be preventing the cable from fully plugging in.

2. **Set the computer/phone's volume to 100%** and the audio player software to 100% volume. Turn off all audio and vibrate notifications (use Airplane mode). Close any applications that make notification sounds such as Facebook.

3. Depress both Reverse buttons (A and B) and the Ping button while powering on the DLD. When you see the Channel A Hold button blink, the DLD is ready to receive firmware. Release the buttons.

4. Begin playing the file. Immediately you should see red Channel A Loop LED blink. The blue light will flash from time to time as well. Do not interrupt the process! You can monitor the audio by listening to the **Out B** jack.

5. If the monitored audio stops before the end of the file and/or the lights stop blinking, an error has occurred and you should try again. Verify the cable is not loose, all sounds/vibrate/notifications are off, and that you have downloaded the audio file completely (avoid streaming or playing from the browser). Check the volume is at 100%. Remove the protection case from your smart phone. Stop the audio file, reset it back to the start, and tap Reverse A button to reset. The Hold A button should light up. Play the file from the beginning again.

6. If the file loads successfully, the DLD will immediately start running. In some cases, you may need to do a Factory Reset (see Factory Reset section).

**Estimated late June 2016, the open-source licensed source files (in C, for compiling with gcc-arm) can be found at https://www.github.com/4ms/DLD
The audio files are not at this location, they are available from http://4mscompany.com/dld.php

Clock Bus Jumper

Clock bus is a 1:1 clock that runs along the Gate pin of the Doepfer-specified power system. Modules such as the 4ms QCD, RCD, SCM, and PEG are compatible with the clock bus system. The DLD fully buffers the bus clock and has diode protection and a pull-down resistor. To use the DLD on a clock bus system, install the jumper in one of two positions:



Bus Clock Send:
Connect the jumper on the upper position (“BUS CLK SEND”) to send the master clock from the DLD to the clock bus. The clock will be identical (but separately buffered) as the signal on the Clock Out jack.

It is not recommended to have more than one device sending (master) on the same clock bus system. The DLD will not be damaged, but results will be unpredictable.



Receive Clock Bus:
Connect the jumper on the lower position (“BUS CLK RECV”) to receive a ping clock from the clock bus. The signal on the bus clock will be automatically patched to the Ping jack. By plugging a cable into the Ping jack, the bus clock will be disconnected from the DLD.
Note: When receiving a bus clock, you must stop the external bus clock or patch a dummy cable into the Ping jack in order to use the Ping tap button.
This is the factory default for some DLD serial numbers less than 500.

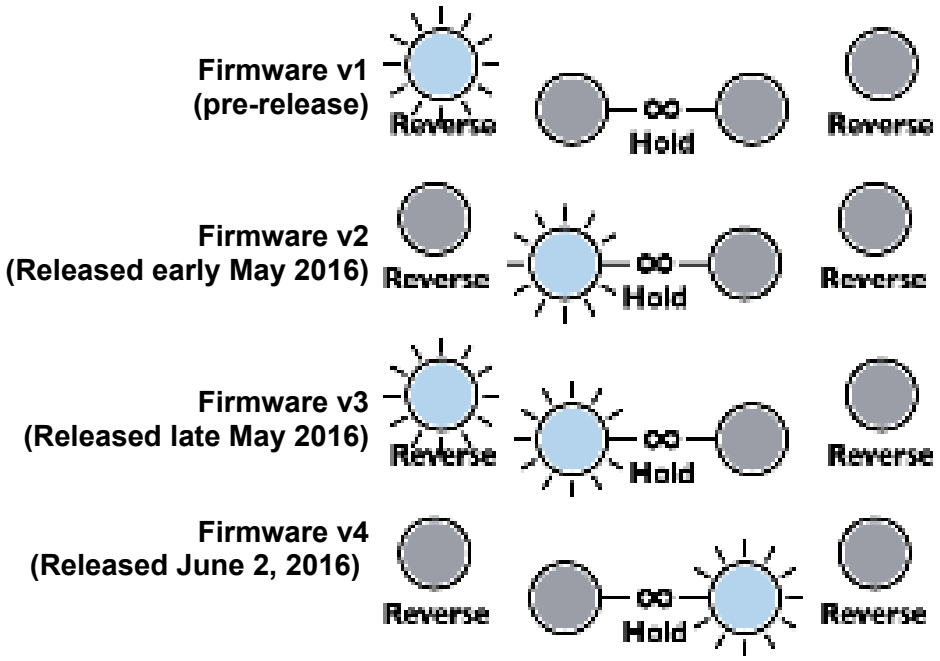


Clock Bus Disabled:
Remove the jumper completely to disable clock bus support. The jumper may be parked on one pin.

Special Modes

Firmware Version

To view the firmware version, enter *System Settings Mode* by flipping both **Time** switches up (to +16) and hold down all five buttons until all the buttons and both Loop lights turn on brightly (about 8 seconds). Then, release the five buttons and look at which buttons are lit up. Briefly press and release all five buttons again to exit System Mode



System Settings Mode

System Settings Mode allows you to change the way some features of the DLD behave. For most users, these are not necessary to modify, but advanced users may wish to explore. *It's recommended that new users get familiar with the DLD operation using default system settings before making changes!*

To enter System Settings Mode, flip both **Time** switches up and hold down all five buttons for about 8 seconds until both Loop lights and all five buttons turn on brightly. Release the buttons. You are now in System Settings Mode. **Proceed carefully and take the time to understand what you are doing before pressing any buttons, turning any knobs, or flipping any switches**. Changing a System Setting without realizing what you changed can cause confusion. You can always do a Factory Reset if you want to revert to safe settings (see Factory Reset section below). You also can exit System Settings mode without changing anything by briefly tapping all five buttons when the Time switches are both up.

The **Time** switches are used to determine what parameters you are editing. The buttons and knobs are used to edit these parameters

• Time A up + Time B down: Clock Output Trigger/Gate selection and LED brightness
The clock output jacks can output gates or triggers (22ms pulse width). In this mode, the loop lights and Ping light will each blink once in sequence. A short flicker means the associated jack is in trigger mode, and a longer blink means it's in gate mode. The associated button will also be lit to indicate gate mode, or unlit to indicate trigger mode.

- Gate/Trigger output for **Loop A** jack: Tap **Reverse A** to toggle mode
- Gate/Trigger output for **Loop B** jack: Tap **Reverse B** to toggle mode
- Gate/Trigger output for main **Clock Out** jack: Tap **Infinite Hold A** to toggle mode

The red and blue Loop LEDs can be dimmed:

- LED brightness (red and blue only): Hold **Infinite Hold B** while turning the **Feedback A** knob (In firmware v3 and earlier, it's not necessary to press **Infinite Hold B**)

• Time A up + Time B center: Auto Mute, Soft Clipping, and Mix CV
Auto Mute is a noise gate that silences the input when a signal less than 5mVpp is detected on any input. This prevents runaway feedback if **Feedback** is turned up and the DLD is allowed to run for a long time with no input signal. In most cases, it is not necessary to turn off the Auto-mute, since a 5mVpp signal is extremely quiet.
Soft Clipping enables limiting (compression) when the output signal exceeds 75% of the clipping point. Below this point, the signal is unaffected. This saturation distortion sounds is often more pleasing to hear versus harsh clipping. In most cases, it is not necessary to disable soft clipping.
Mix CV, when enabled, re-assigns the **Delay Feed CV** jack to control the **Mix** (*Dry/Wet)* amount

- *Auto Mute*: Tap **Reverse A** to toggle Auto Mute on and off. When the light is on, Auto Mute is enabled.
- *Soft Clipping*: Tap **Reverse B** to toggle Soft Clipping on and off. When the light is on, Soft Clipping is enabled.
- *Mix A CV*: Tap **Infinite Hold A** to toggle the Delay Feed A CV jack between controlling Mix and Delay Feed. When the light is on, the jack controls Mix.
- *Mix B CV*: Tap **Infinite Hold B** to toggle the Delay Feed B CV jack between controlling Mix and Delay Feed. When the light is on, the jack controls Mix.

• Time A center + Time B center: 1V/octave tracking compensation
The tracking of the **Time CV** jack when the channel is in Unquantized Time Mode can be adjusted to compensate for keyboards or sequencers that are not accurate, or to provide alternate tunings. See the earlier section for a discussion on Unquantized Time Mode and Resonant Delay patch ideas. When you flip both the switches to center, the DLD will automatically set the Delay Feed parameter to 100%, so that the Delay Feed knobs are free to adjust tracking. Also, the DLD will automatically enter Unquantized time mode. To calibrate, you should input a clock or VCO of about 50Hz - 100Hz into the Ping jack (the Ping and loop lights will not flash, that's OK!), turn the Time and adjust Feedback until you hear a resonant sound. Input a 1V/octave CV signal into the Time CV jack (i.e. from a sequencer or a keyboard).

- Adjust tracking for **Time A CV** jack: Press and hold **Infinite Hold A** while turning the **Delay Feed A** knob
- Adjust tracking for **Time B CV** jack: Press and hold **Infinite Hold B** while turning the **Delay Feed B** knob

Releasing the button will temporarily lock in the value of the knob (this prevents accidentally bumping the knob while testing the setting). With the **Delay Feed** knob at center, the tracking is 1.00V/octave, at minimum it's 0.75V/octave, and at maximum it's 1.25V/octave.

• Time A up + Time B up: Exiting and saving changes (and firmware version)
With both switches up, the firmware version is displayed on the button lights. See above section.

- To exit *System Settings Mode* without making changes, flip both **Time** switches up and briefly tap all five buttons
- To save your changes and exit *System Settings Mode*, flip both **Time** switches up hold down all five buttons for 8 seconds.

Calibration Mode

Your DLD has been calibrated at the factory and should not need re-calibration. Hardware hackers should be aware that the calibration data is stored in FLASH memory at `0x08004000` to `0x08007FFF`. If this sector is erased, the DLD will automatically re-calibrate itself on boot. Make sure nothing is plugged into any CV jacks the first time you boot after erasing this sector. The zero-point of the CV jacks, as well as the DC offset of the audio output jacks can be adjusted. Contact 4ms@4mscompany.com if you believe your DLD needs to be re-calibrated and we will step you through the process.

RAM Test Mode

The DLD is static-sensitive and care should be taken to not shock the unit with static electricity.
If you believe the DLD has been damaged, the RAM chip can be tested.
Unplug all cables. Power the DLD off. Flip **Time A** up, and **Time B** down. Hold down **Reverse A**, **Hold A**, and **Hold B**. Power the DLD on, and release the buttons when **Reverse A** turns on.
Watch the lights. After a moment, the **Reverse B** button will turn on. After another moment, the lights will display if there are errors. If there are no errors, all seven lights will turn on (the red light, the blue light, and the five buttons).
If your DLD has some lights off, or if the lights flash, contact 4ms@4mscompany.com

Factory Reset

A factory reset is not necessary unless you have changed System Settings.
To do a factory reset:
Unplug all cables. Power off the DLD. Flip **Time A** down, **Time B** up. Hold down **Infinite A**, **Infinite B**, and **Reverse B.** Power the DLD on. Release the buttons, the four Reverse/Infinite buttons will flash.
Now, flip **Time A** up, **Time B** down. Hold down all five buttons (**Rev A, Inf A, Rev B, Inf B, Ping**) for 10 seconds. Release the buttons when you see the lights make a chase sequence. Power off, and power back on. The DLD has now been reset to factory settings.

Technical Specifications

- 20 HP Eurorack format module
- 0.98” (25mm) maximum depth with power cable

- Power consumption:
  - +12V rail: 188mA max (jumper set to 5V)
  - -12V rail: 48mA max
  - +5V rail: not used
  - Connect red stripe of power cable towards the bottom of the module, which is marked with a white stripe and the words “-12V” and “POWER”

- Audio Inputs
  - 0Hz (DC) to 24kHz
  - 16.8V peak-to-peak maximum before clipping (when AC coupled)
  - Optional jumper to choose DC or AC coupling on input

- Audio Outputs
  - 0Hz (DC) to 24kHz with maximum -1.7dB difference between input and output (see technical specification section)
  - +10.5V to -10.5V maximum output
  - Soft limiting compression allows for saturation before clipping (can be disabled)

- Clock outputs (Main, Loop A, Loop B):
  - 0V to 8.2V
  - +/- 2.4µs maximum jitter (i.e. 0.001% at 120 BPM)
  - Rise or Fall time: <1µs
  - Pulse Width
    - Trigger mode: 22ms
    - Gate mode: 50% (square wave)
    - *At clock speeds faster than 22Hz, the clock output becomes 50% duty cycle*

- Sampling
  - 24-bit sample at 48kHz, 32-bit processing, 16-bit storage in RAM
  - 256Mbit nonvolatile SDRAM chip for a maximum of 2 minutes 54 seconds per channel (6m49.5s tota @48kHz/16bitl)
  - Future firmware will allow for 24-bit storage in RAM (thus reducing maximum delay time)

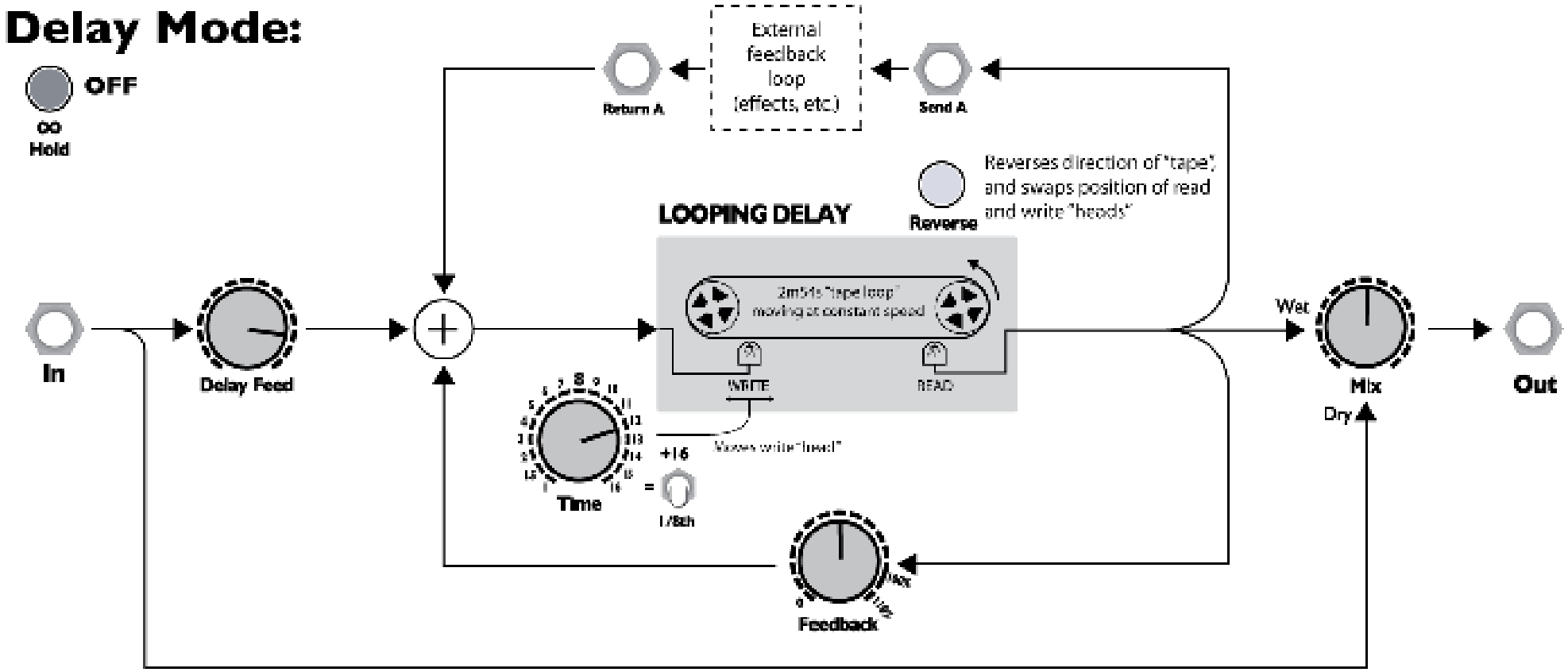
- Voltage ranges
  - The DLD is reverse-polarity protected. Plugging a Eurorack power cable in backwards will not damage the DLD. However, this could potentially damage your power supply or other modules because the +12V rail will be shorted to the +5V rail.
  - All jacks are tolerant of signals from -12V to +12V, but signals outside the active range are clipped to the active range

Change Log:

Version 4 (June 2, 2016)

- Added Memory buffer clear
- Feedback CV caps at 100% when knob is at 0
- Added more System Settings Mode features (and documented it)
  - Gate/Triggers for clock outputs
  - Dim the loop LEDs
  - Re-assign Delay Feed CV jacks to Mix
  - Enable/disable auto-mute and soft-clipping
  - Adjust 1V/oct tracking
- No clicking when doing a Reverse or exiting Infinite Hold mode
- Exiting Infinite mode quickly crossfades to play from the beginning of the loop
- Fixed the loop lights so they display better at high speeds

Delay Mode:



Inf. Hold/Loop Mode:

