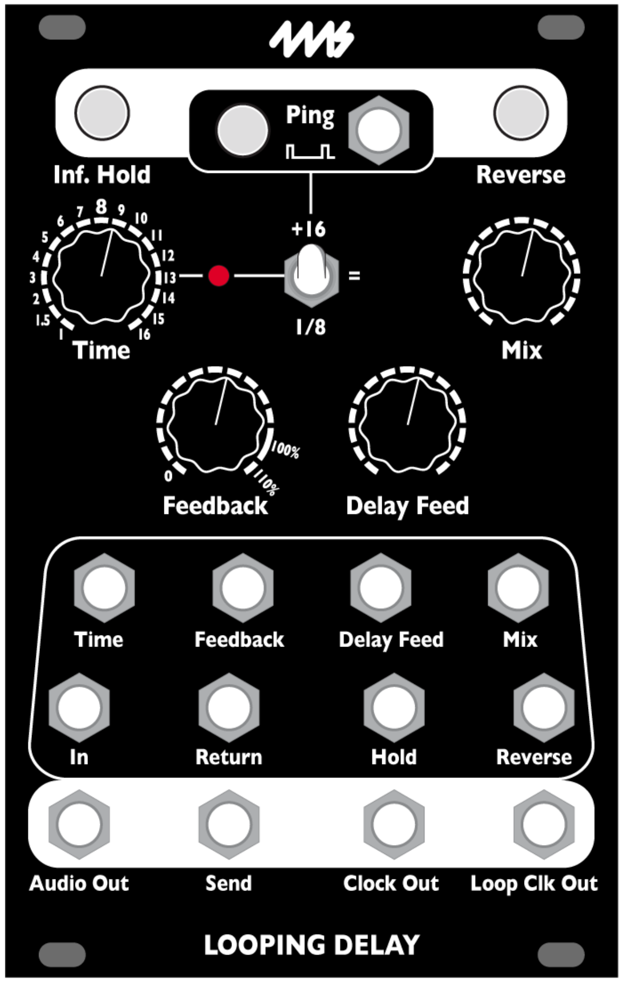
Looping Delay
4ms Company
User Manual 1.0 – June 11, 2023



The **Looping Delay** is a clock-based delay for creative synthesis. Not a tape or analog emulation, but a modern crystal-clear digital delay, the **Looping Delay** combines features of delay, looping, and sample-tight synchronization for powerful and dynamic sound capture and modification. Based on the **Dual Looping Delay**, the **Looping Delay** offers the same high-quality sound in a smaller, single-channel format.

- Stereo or mono mode (Return/Send becomes right channel In/Out in stereo mode)
- 87 seconds maximum delay/loop time in mono, or 43 seconds in stereo.
- 48kHz/24-bit sampling rate, loop recorded at 16-bit
- Extremely quiet, low noise, low jitter design
- 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 features

Table of Contents

Setting up your Looping Delay

1. Power off your Eurorack system.
2. On the back of the **Looping Delay** you will see a 16-pin header labeled “-12V”. The 16-pin header connects to a Eurorack power header using the included power cable. Connect one end of the power cable to a Eurorack power header on your power supply distribution board and the other end to the **Looping Delay** with the red stripe on the power cable oriented towards the bottom of the module.
3. Using the included screws, securely attach the **Looping Delay** to the rails of your case.
4. Power on your Eurorack system.

Note: The **Looping Delay** *is reverse-polarity protected, but incorrectly connecting any module in any system can damage other modules on the power bus.*

Basic features:

- 87 seconds of delay/loop time in mono mode, 43 seconds in stereo mode
- 48kHz/24-bit sampling rate, loop is stored at 16-bit
- Extremely quiet, low-noise and low-jitter design
- **Ping** button and external clock jack 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 output
- Time switches change range of **Time** knob from 1/8th notes up to 32 bars
- Digital feedback, up to 110%
- **Delay Feed** 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”
- **Reverse** mode plays memory contents backwards
- Resonant Delays (ala Karplus-Strong)
- Delay time can be as short as 650uS
- **Time** CV jack can respond at 1V/octave in *Unquantized Time Mode*
- Trigger inputs for toggling **Infinite Hold** and **Reverse**
- **Send** and **Return** for feedback with external modules
- **Send** and **Return** function as right channel In and Out in stereo mode
- CV jacks to control **Time**, **Feedback**, **Delay Feed** (record level), and **Mix**
- Various algorithms can be selected interfacing with jittery external clocks (*External Clock De-jitter*)
- *Quantized Change Mode* quantizes toggling of **Infinite Hold** and **Reverse**
- *16*HP Eurorack module

Controls and Jacks

Ping Button and Jack

The **Ping** button allows you to tap the tempo to set the base clock. One tap is equal to one beat. The base clock can also be set by an external clock by patching it into the **Ping** jack**.**

The **Looping Delay** requires a base time, which is referred to as a *“beat”* in this manual. The delay/loop time is 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 **Looping Delay**, the *beat* is the **Ping**.

There are several ways to establish a 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 received at the jack.
- Just turn on the **Looping Delay** and do nothing more! The **Looping Delay** automatically boots up at 240BPM.

Another way of setting the timing is by jumpering the Clock Bus header to RECV and using a Clock Bus master (such as the 4ms QCD or MiniPEG) to send the clock over the power distribution board. See the *Bus Clock Jumper* section on page 15.

The **Looping Delay**’s clock is extremely stable and jitter-free. It's highly recommended to use the internal **Looping Delay** clock (tap clock) if possible, because it's quantized to the sample clock and has less jitter than most commercially available clock sources. Using the **Looping Delay** as a master clock will provide the tightest timing possible. However, if it's not possible to use the

Looping Delay as a master clock, you can sync to an external clock by patching it into the **Ping** jack. The **Looping Delay** will quantize the external clock to its own sample rate and output quantized clocks on the **Loop Clk Out** jack. If there is any jitter or drift in the external clock, the **Looping Delay** will track and follow these discontinuities, which can result in interesting (or noisy!) artifacts. See section on *Using External Clocks* section on page 11 for more information.

Inf. Hold Button and Hold Jack

The **Looping Delay** can operate as a delay/echo, or it can play a loop. Pressing the **Inf. Hold** button toggles the state of Infinite Hold mode when the button is *released*. Similarly, sending a trigger into the **Hold** jack toggles the state on the trigger's rising edge. 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. This works like a traditional echo or looping delay by recording and playing continuously.

When **Inf. 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 **Looping Delay** will start looping what you *just recorded.* The loop is defined by start and end points in memory. Wherever the **Looping Delay** 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).

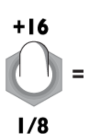
Reverse Button and Jack

Reverse reverses the direction that memory is read and written.

Pressing the **Reverse** Button toggles **Reverse** on/off when the button is *released*. When reversal is engaged, the **Reverse** Button will illuminate.

The **Reverse** CV jack toggles the reverse state on the rising edge of the trigger.

Reverse has slightly different effects depending on whether you are in Infinite Hold mode or not. In normal mode (**Inf. 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).



Time Knob, Switch and CV Jack

The **Time** knob, switch, and CV jack perform simple arithmetic on the base time to produce a delay/loop time. The red LED located between the knob and the switch blinks in time with the current delay/loop time.

The **Time** knob sets the loop time in number of beats, ranging from 1 to 16.

The **Time** switch modifies the **Time** knob setting as follows:

- Center (**=**): the **Time** knob directly sets the number of beats for the loop or delay time. For example, if the **Time** knob is pointing to 5, the channel will make an echo (or loop) every 5 beats. Loop/delay time will range from 1 to 16 beats.

- Up (**+16**): 16 beats are added to whatever the **Time** knob shows. So if the knob is pointing to 5, the delay or loop time will be 5+16 = 21 beats. Loop/ delay time will range from 17 to 32 beats.
- Down (**1/8**): The **Time** knob is divided by 8, making the delay/loop time 8 times as fast (eighth notes). For example, if the knob is pointing at 5, then the output will echo in 5/8th notes, or eight loops for every five beats. Loop / delay time will range from 1/8 to 16/8 (2 beats)

The **Time** CV jack accepts signals from -5V to +5V. 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.

Taken together, the **Time** knob, switch and CV jack set the length of the loop or amount of delay time. This period of time is relative to a *beat* (or base time), which is set by tapping the **Ping** button or patching an external clock into the **Ping** jack.

For example, tap a tempo of 0.5 second. This makes one *beat* be ½ second (120BPM).
Set the **Time** knob to 8, and center the **Time** switch. 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, 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.

Feedback Knob and CV Jack
The **Feedback** knob sets the amount of signal read from memory to be written ahead in the future.
On the **Looping Delay**, **Feedback** ranges from 0 to 110%. With no feedback (knob at 0), you will hear one echo. As you turn the **Feedback knob** 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.

The **Feedback** CV jack accepts signals from 0V to +5V. Negative voltage on the CV jack is ignored. The knob and CV jack are added together, and have a maximum combined value of 110%. However, if the knob is set to 0%, then applying a voltage of +5V or greater will cause a **Feedback** amount of exactly 100%. This allows you to easily access the 100% Feedback setting using CV.

Normally, **Feedback** has no effect in Infinite Hold mode. See the *Windowing* section on page 8 for special usage of the **Feedback** knob and jack while holding down the **Inf. Hold** button.


Delay Feed

Delay Feed Knob and CV Jack

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 **Audio Out** jack when the **Mix** knob is turned all the way down). 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.


Delay Feed

The **Delay Feed** CV jack accepts signals from 0V to +5V. Negative voltage on the CV jack is ignored.

The knob jack are added together and have a maximum combined value of 100%. Note that in **Infinite Hold** mode, **Delay Feed** has no effect since there is no recording.


Mix

Mix Knob and CV Jack

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 (not the **Return** jack) before the **Delay Feed** parameter has any effect.

The **Mix** knob crossfades between signals on the **In** jack (dry) and signals read from memory (wet). Turning the knob all the way to the left will yield a completely dry signal. Turning the knob all the way to the right will yield a completely wet signal.


Mix

The **Mix** CV jack accepts signals from 0V to +5V. Negative voltage is ignored. The knob and jack are added together to produce the amount of dry/wet signal.


In

Input Jacks: In and Return

The **In** jack is the main audio input.


Return

The **Return** jack has a different function in mono mode versus stereo mode (See *Stereo and Mono Modes*, page 10). In mono mode, it’s the audio input from the external feedback loop. Typically you will patch the **Send** jack (or a copy of the main audio input) into an effect, and then patch the output of the effect into **Return**. The signal on this jack will be mixed with the main audio input signal and the internal feedback signal (controlled by the **Feedback** knob and jack) and then recorded into memory. Note that the signal on Return jack will be written to memory at 100% volume, without any attenuation. Therefore it’s recommended to use an external attenuator to control the amount of external effect.

In stereo mode, the **In** jack is the left audio input, and the **Return** jack is the right audio input. There is no effects loop capability in stereo mode.

Both jacks accept signals from -10V to +10V and are AC-coupled.


Audio Out

Output Jacks: Audio Out and Send

The **Audio Out** jack outputs the Dry/Wet mix.

The **Send** jack has a different function in mono mode versus stereo mode (See *Stereo and Mono Modes*, page 10). In mono mode it outputs the delay signal only. This is typically used with the **Return** jack to create an external effects loop. Note that since the **Return** jack outputs the signal read from memory, the very first echo will not be effected by the external effects loop.


Send

In stereo mode, the **Audio Out** jack is the left audio input, and the Return jack is the right audio input.

Both jacks output signals from -10V to +10V and are DC-coupled.

Clock Out

Loop Clk Out

Clock Out and Loop Clk Out Jacks

The **Clock Out** jack outputs the stable master clock. It sends a clock that's in sync with the base clock ("beat" clock or Ping clock), whether you tapped in a tempo or are using an external clock.

The **Loop Clk Out** jack outputs a clock that's in sync with the the loop/delay time.

All the clock outputs are quantized to the sample-rate. This insures they are jitter-free (assuming your external clock has low jitter). The **Looping Delay** 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. At fast clock speeds (audio rate), the jacks output square waves.

In **Inf. 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.

Infinite Hold vs. 100% Feedback

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

When **Inf. Hold** is on, you can adjust the **Time** parameter from one setting to another, and then back to the original setting and it will sound exactly the same as it did the begin with. This is not the case with **Inf. 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 **Looping Delay** 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.

Another difference is that when **Inf. Hold** is on, the **Feedback** knob has a special function: windowing. Hold down **Inf. Hold** while turning **Feedback** to change the start and stop points of the loop (see *Windowing* section on page 8). There is no windowing with **Feedback** at 100% and **Delay Feed** at 0%.

The final difference is that with **Feedback** at 100% and **Delay Feed** at 0%, 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 **Inf. Hold** is on, the loop is static and immutable.

Why do I hear something I played a long time ago?

Notes on the nature of Looping Delays...

When **Inf. 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. 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 **Looping Delay** 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.

Normally you won't have to think about this: the **Looping Delay** 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. Another way often happens when **Infinite Hold** and **Reverse** are toggled many times while **Time** is being modulated. This can be surprising! But it also can be very useful.

Signal Routing: Using Send and Return

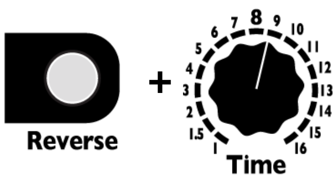
The **Send** and **Return** jacks are simple but offer a wealth of possibilities for creative patching. The **Send** jack simply carries the delayed signal, unmixed with the input. It's the same as the main **Audio Out** jack if the **Mix** control is set fully clockwise. The **Return** input is summed with the main **In**. However, it does not appear in the "dry" signal and is not attenuated by the **Delay Feed** knob and CV.

Feedback with External Processing
The classic use of **Send** and **Return** in a delay effect is to modify the delayed signal before it is returned to the mix. Probably the most popular 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. Note that the first echo will not be processed by the external effect, only the second and subsequent echoes will go through the Send/Return path.

In patches involving externally processed feedback, the gain of the processing chain is always a concern. 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.

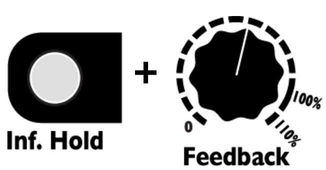
Advanced Usage

Loop start and stop points: Trimming the loop



When **Inf. 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. Therefore, you can change a loop that's 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 *Windowing* Section below.

Windowing (aka Scrolling or Scrubbing)



When **Inf. 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 a special purpose when **Inf. Hold** is on: Windowing, also known as scrolling or scrubbing.
To use this feature, first makes sure you're in **Inf. Hold** mode. Then, hold down the **Inf. 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 the **Feedback** knob. The amount of scroll is determined by how 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 **Inf. Hold** mode is on.

Windowing Tutorial
Let's try it. First get set up: Set **Time** so you have a loop length that's a few seconds long. Remember, the red loop light will flash once per loop. Set **Feedback** low (25 - 50% perhaps) to keep things simple for now. Play some sounds for 10-20 seconds and then turn on **Inf. 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 **Inf. 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 a few seconds earlier. Play with this some more: Press and hold **Inf. Hold** again and turn **Feedback** back half a turn. Hear how the loop now starts in the middle? Remember that turning **Feedback** while in **Inf. Hold** mode has no effect unless you're holding down **Inf. 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 the **Inf. Hold** button while you turn **Feedback** to 0%.
3. Release **Inf. Hold** button.
4. Repeat as needed (turn **Feedback** to 100%, then press button and turn **Feedback** back to 0, release button...)

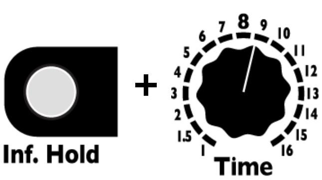
Tip #2: If you want to scroll back very far even more quickly than Tip #1, change the **Time** parameter to very long (perhaps flip the time switch up to +16). Since turning **Feedback** + **Inf. Hold** scrolls by *one loop size*, making the loop size enormous lets you scroll by enormous amounts with just one knob twist! You can scroll back a maximum of 87 seconds in mono mode, or 43 seconds in stereo mode.

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 **Inf. 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 **Inf. Hold**, the **Feedback** CV jack will no longer control the window.

Unquantized Time Mode and 1V/Oct CV

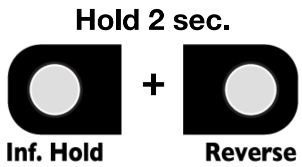


Normally the **Time** knob and CV are quantized to integer amounts (1-16), and simple fractions (1 – 1/16). This is called *Quantized Time Mode*, and is the default mode. It's possible to change to *Unquantized Time Mode*, where the knob and CV provide continuous control of the **Time** parameter (not quantized to integer or simple fractional amounts)
To change to *Unquantized Time Mode,* turn the **Time** knob while holding down the **Inf. Hold** button. To change back to *Unquantized Mode*, turn the **Time** knob without holding down the **Inf. 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 **Inf. Hold** button while turning 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 response is opposite to *Quantized* mode, where additional voltage makes the period <u>increase</u>. 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 beats. For a more accurate 1V/oct response, keep the **Time** switch centered or down.

Stereo and Mono Modes



Normally, the **Looping Delay** operates in *Mono Mode*, where a mono signal is input in the **In** jack, and the echoes, loops, and delayed signal is taken out of the **Audio Out** jack. In this mode, the **Send** and **Return** jacks function as additional inputs and outputs, but the delay signal path remains mono.
The **Looping Delay** can also function as a stereo delay by using the **Return** and **Send** jacks as right channel input and output, and the **In** and **Audio Out** jacks as left channel input and output. In *Stereo Mode*, there are two isolated signal paths (left and right channels), whose parameters are always the same. That is, the **Time**, **Feedback**, **Delay Feed**, and **Mix** amounts are always the same for both channels, being controlled by the knobs and CV jacks. In *Stereo Mode*, the maximum delay time is exactly half of the amount in *Mono Mode*.

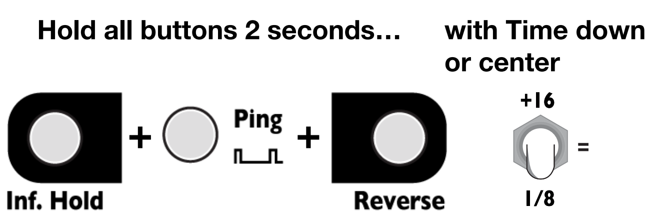
To change between Mono and *Stereo Modes*, hold down the **Reverse** and **Inf. Hold** buttons for about 2 seconds. The **Reverse** and **Inf. Hold** lights will flash to indicate which mode is now active: rapidly alternating means *Stereo Mode*, and more slowly flashing together means *Mono Mode*. Release the buttons when you see the lights flash. There may be a discontinuity, click, or some unwanted sounds when changing modes.

	Max Echo/ Loop Time	In jack	Audio Out jack	Return jack	Send jack
Mono Mode	87.4 seconds	Mono input	Mono output	FX Return	FX Send
Stereo Mode	43.7 seconds	Left input	Left output	Right input	Right output

Octave Up/Down When Changing Modes

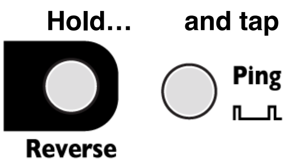
If you have any sound already recorded into memory (for example, if you're looping, or if you have **Feedback** set above zero and played something recently), the pitch of that sound will jump up or down by an octave. When changing to *Stereo Mode*, the pitch will jump up, and when changing to *Mono Mode* it will jump down. This can be used as a special effect, but if you don't want to hear this effect, clear the memory before changing modes (see next section).

Memory Clear



When using the **Looping Delay** 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 more quickly you can do so by making sure the **Time** switch is down or centered, and holding down all three buttons (**Reverse, Ping,** and **Inf. Hold)** for about three seconds. The lights will flash once and you may hear a beeping sound briefly while the memory is cleared. Let go of the buttons and the buffer will be cleared. Note that if the **Time** switch is flipped up, this button combination will enter *System Mode*. If you accidentally do this, press and release all three buttons quickly to return to Normal mode.

Quantized Change Mode



Normally **Reverse**, **Inf. Hold,** and **Time** work as you would expect: changes to the control take effect immediately, no matter when they occur. In the *Quantized Change Mode* (or *QCM*), any change to these parameters will be delayed until the next pulse of the **Ping** clock.
This means that if your press the **Reverse** and **Inf. Hold** buttons, turn the **Time** knob, flip the **Time** switch, or apply CV or gates to the **Time**, **Reverse** and **Hold** jacks, no change will be made until the next **Ping** clock. If the **Ping** clock changes (due to tapping the **Ping** button or a change in the tempo of an external clock patched into the **Ping** jack), then any queued changes to parameters will immediately take effect.

The factory default setting for *QCM* is off. To enter *QCM*, hold down the **Reverse** button and tap the **Ping** button once. The **Ping** and **Reverse** lights will flash three times evenly to show entry into *QCM*. Exit *QCM* with the same procedure, in which case the same lights will flash six times in a staggered pattern.

QCM is best explored with a rhythmic patch, sequencer, or drum unit locked with the **Looping Delay** clock. Asynchronous modulations and triggers can be used freely, since they will be forced to quantize with the **Ping** clock. In this sense, *QCM* might be thought of as a kind Sample and Hold that locks modulation sources with the fundamental time. Go ahead, experiment! It can be addicting.

Note that continually toggling **Reverse** in this mode can lead to quite a lot of “memory scraps” that come back at surprising times, especially when the **Reverse** switching continues after an input source is killed. This can be a real plus (as well as a real surprise) as these bits and pieces will remain locked to the clock. If it gets out of hand, you can use the *Memory Clear* function, or discontinue triggers to **Reverse** so that the “scraps” eventually clear themselves out.

Using External Clocks

Whenever possible, use the **Looping Delay** as a the clock source by patching out of the **Clock Out** jack to your other clock modules. However, sometimes you have to clock the **Looping Delay** with another source, in which case you would patch the external clock into the **Ping** jack. There are many issues with clock sync that arise from using different manufacturers’ gear together, and an investigation of every type of issue is beyond the scope of this section. However, here are some initial things to try if you are having difficulties syncing clocks:

- **Use the least jittery device as the clock source**. Try to use the **Looping Delay**, or some other high-end gear that guarantees a low-jitter clock. Analog devices generally drift over time. Digital devices can also have jitter due to rounding errors and processor lag or latency. If you're using something else as the clock source, try running it directly into the **Looping Delay** and then using the **Looping Delay’s Clock Out** jack as the clock source for other gear.

- **Try running the clock source into a clock divider** before into the **Looping Delay.** This will average out some jitter, and slow down how often the **Looping Delay’s** ping time is updated. You may wish to configure your clock source equipment to generate a faster clock (such as 24ppq or 48ppq) and then divide it down with a Rotating Clock Divider or some other clock divider module.

- **Try the Looping Delay’s five different *External Clock De-Jitter modes*** (see *System Settings Mode* section). One mode in particular may work best with your equipment. Analog clock sources tend to drift, so *Linear Average of 4* is best (or use a clock divider, see above). Digital clock sources with **Inf. Hold** on the **Looping Delay** might prefer a *Moving Average of 2*, or *Ignore 1%*. Digital clock sources without **Inf. Hold** on the **Looping Delay** might prefer *Ignore 1%* or *Ignore 0.2ms* if you are concerned with phasing between the layers.

Patch Ideas

Resonant Delay (Waveguide, Karplus-Strong)

Basic Resonant Delay

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 **Looping Delay** supports resonant delays. Simply tap a very fast tempo and flip the **Time** switch to **1/8.** Alternatively, feed a square wave LFO into the **Ping** jack. The LFO should be in the low or mid audio range, around 20-50Hz is the most stable, but up to 500Hz will work.
Next, feed a signal into the **In** jack: a noise burst or a short sample from the **STS** or **Sampler** (**Length** set near minimum) is an excellent sound source. You also can use a trigger (keep in mind the amplitude of the trigger will effect the sound).
Use the **Time** switch and knob to set the pitch. **Feedback** sets the resonance, and **Delay Feed** effects the level and resonance as well. Make sure **Inf. Hold** is off. The frequencies present in the input signal can ring loudly at particular settings, so be prepared for surprises!
If you consistently have unstable results when using an external clock into the **Ping** jack, try setting the *External Clock De-jitter* setting to a different value. Small amounts of jitter in the external clock can cause large changes in the sound.

Playing Resonance With a Keyboard or Sequencer

Holding down **Inf. Hold** and turning the **Time** knob changes the **Looping Delay** into *Unquantized Time Mode* (see section on page 9 for discussion). This feature is great in combination with Resonant Delays. First, get a nice resonant delay sound happening. Then hold down the **Inf. 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 **Looping Delay** create individual “notes”. You can also patch into **Delay Feed** to get a different effect.

Sound on Sound Looping #1: Creating and Removing Layers

With the **Looping Delay** you can build a loop by layering audio on top of itself. Since the **Looping Delay** is always recording when not in **Inf. Hold** mode, each time you add a layer, the audio will be recorded. You can then use the *Windowing* feature to scrub backwards in time, essentially peeling off newer layers, or scrub forwards to restore the layers.

Tap a tempo on the **Ping** jack of about one second. Turn the **Time** knob to 8, and flip the **Time** switch to = (center). This creates a loop of about eight seconds (or eight “bars", if you consider the **Ping** clock to be one bar). Make sure **Inf. Hold** is off.
Start with **Feedback** at 90-100%, **Delay Feed** at 100%, and **Mix** at center.

Play a sound into the **In** jack. It can be a short burst, a drone, random noises, anything. Since you’ll be layering more things on top, try to keep it under 8 seconds when you do this the first time. After you finish, press **Inf. Hold** to lock the loop in. If you want all your layers to be 8 bars each, then only turn **Inf. Hold** on or off when you see the red light go from off to on.
Tip: Enabling Quantized Change Mode will force enabling/disabling **Inf. Hold** to happen exactly on the **Ping** clock (which happens every second). You also could wait until the red light goes off, patch **Loop Clk Out** into the **Hold** jack, and then unpatch it after the red light goes on.

You should now be hearing the sound repeating every eight seconds.
Next, prepare to play a different sound. If this is your first time doing this patch, try to make this sound distinct from the first so you’ll recognize it easily. When you’re ready to play the second sound, turn off

Inf. Hold and play the sound. When you're done, turn **Inf. Hold** back on, either by just pressing the button or using any of techniques mentioned above.

Repeat this process to add more and more layers: turn **Inf. Hold** off, play the sound, turn **Inf. Hold** on.

You can control the level of each layer by adjusting the **Delay Feed** knob. Turning it down will make the new layer more quiet. If your sound is a constant tone or drone, consider turning Delay Feed to 0 before turning off Inf. Hold. Then fade Delay Feed. When you're done, turn Delay Feed back down to 0. This will make the new layer fade in and out.
You also can control the level of the previous layers by adjusting **Feedback**. Turning **Feedback** down will make older layers fade out each time they're played.

Peeling off Layers
After you've built up several layers, you can experiment with "peeling" them off. Make sure you are in **Inf. Hold** mode. Turn **Feedback** all the way up (110%). Press and hold **Inf. Hold** and turn **Feedback** down to 0. Release **Inf. Hold**. You should hear the last 8 seconds of audio removed from the loop. If you recorded exactly 8 seconds each time, then the last layer will be gone. What happens is that Inf. Hold + Feedback adjusts the loop start and end times. By turning Feedback counter-clockwise while holding Inf. Hold, an earlier portion of memory is played, before you recorded the latest layer.

To peel off another layer, turn **Feedback** back to 110%, then hold **Inf. Hold** while you turn it back to 0%. If you want to restore the layer, do the opposite motion: turn **Feedback** to 0%, hold **Inf. Hold** while you turn it to 110%.
You also can pull of parts of a loop by turning Feedback less than a full turn. For example, if you turn Feedback from fully up to center, only the previous four seconds will be removed.
Another technique is to turn the Time knob. The amount Inf. Hold + Feedback shifts the loop is relative to the Time parameter. If you turn the Time knob down to 1, then a full turn of Feedback will only shift the loop by 1 bar.

Sound on Sound Looping #2: External Control

This is a great patch for long loops. Set a long gentle clock speed. Turn the **Time** knob to 16 (or something slow). Turn **Feedback** to 100%, **Delay Feed B** to 0%, and turn **Mix** to 50% (or to taste). Now run audio into the **In** jack. Try running a melody, a percussive sequence or an evolving drone, perhaps.

Patch a manual CV source into the **Delay Feed** CV jack. It can be anything that generates CV when you activate it: a keyboard with velocity or pressure output, a manual CV knob (from the 4ms **SISM**), or perhaps an envelope output that's triggered manually (by quickly tapping the **Cycle** button twice on the 4ms **MiniPEG** or **EnvVCA**). When you apply CV, you will bring up the **Delay Feed** parameter which causes audio to be recorded onto the loop.

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** jack for the loop output. Then set the **Mix** knob to center and use the **Audio Out** jack to monitor the incoming audio.

Sound on Sound Looping #3: More External Control

In this variation, you can use one CV signal 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** 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 **Looping Delay**'s **Feedback** CV jack.

Now you can activate the “record” CV like in the previous patch to lay down new material, but you can also activate the “clear” CV to fade out material from the loop. You can even activate both pads at the same time to replace loop material with new material.

What's happening in this patch is that the **SISM** is turning the 0V to 5V (or 8V or whatever the maximum) 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 **Looping Delay’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.

Granular Scrubbing

A really neat effect can be obtained by changing the **Reverse** jack to respond to gates, and patching a PWM pulse wave into the Reverse jack. This allows you to scrub across “grains” in an audio loop at variable playback speed without changing the pitch.

First, enter *System Setting Mode* and set the **Reverse** jack to *Gate* mode (See *System Settings Mode*, page 15). Then patch audio into the **In** jack and take the output from the **Audio Out** jack. Tap a slow tempo, maybe one second, and set **Time** so the loop time is a couple seconds. Turn **Feedback** down and **Delay Feed** up. Play some audio, perhaps a drum loop, a vocal sample, or a melody line. Let the audio play through (make sure **Mix** is set at least 50/50), and then press **Inf. Hold** to lock in the loop. Listen to the loop play once or twice to get a feel for what it sounds like normally.

Now the trick! Patch a pulse wave with variable pulse width (sometimes called PWM) from your favorite LFO or clock module to the **Reverse** jack. The 4ms **QCD** with the **QCD Expander** works nicely by using the Gate PW knob to adjust pulse width. You also can use the **EOR** or **EOF** output from the any of the 4ms **EnvVCA** modules and use the ratio of Rise and Fall sliders to set the pulse width. The frequency should be between about 2Hz and about 20Hz.

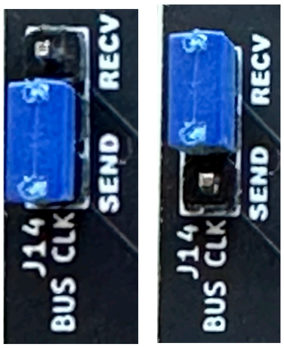
At first, set the pulse width to about 60-70% or so. You should see the **Reverse** light flicker, and the loop should immediately start playing slowly forward or backwards. Tap the **Reverse** button to make it play the other direction. Change the LFO’s pulse width to adjust the playback rate. As you approach 50%, the loop will slow down, until it hovers at 50% and then starts playing back in the opposite direction.

The frequency of the VCO changes the “grain” size. At 2Hz there is a noticeable “stutter”, and at 20Hz it sounds like a medium/low fidelity granular effect. If the VCO is too fast or if the pulse width is too extreme, the effect will be lost.

The reason this effect works is because it plays forward when the pulse is low and backwards when the pulse is high (or vice-versa if you toggled **Reverse** with the button). So, a pulse width of 50% will effectively hover on a single grain because it plays forward and backwards equally. But, a pulse width of 60% will play forward for 60% of the time and backwards 40% of the time, thus slowly moving forward at 20% speed. The makes the audio slow down to 1/5 speed without any pitch shifting.

Special Features

Bus Clock Jumper



Bus Clock is a 1:1 clock that runs along the Gate pin of the power system. Modules such as the 4ms QCD, RCD, SCM+, PEG, DLD, SWN, and MiniPEG are compatible with the clock bus system. To use the **Looping Delay** on a bus clock system, install the jumper in one of two positions:

Bus Clock Send:
Connect the jumper on the lower position (“SEND”) to send the master clock from the **Looping Delay** 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 on the same clock bus system.

Bus Clock Receive:
Connect the jumper on the upper position (“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 **Looping Delay**.
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.*

Bus Clock Disabled:
Remove the jumper completely to disable clock bus support. This is the factory-default. It’s safe to connect the jumper to only one pin.

System Settings Mode

System Settings Mode allows you to change the way some features of the **Looping Delay** behave. For novice users, these are not necessary to modify, but advanced users may wish to explore.

It's recommended that new users get familiar with the **Looping Delay** *operation using default system settings before making changes!*

To enter *System Settings Mode,* flip the **Time** switch up and hold down all three buttons for two seconds. Release the buttons when you see all the buttons lights turn off.

To exit *System Settings Mode*, repeat the process (hold down all three buttons while **Time** is flipped up). If you just briefly press the buttons (releasing them in less then two seconds), any changes you made will stay in effect until you power the module down. The next time you power the module on, the previous settings will be restored.
If you wish to save your changes such that they will persist even after powering down, hold down the three buttons while **Time** is flipped up for at least two seconds. Release them when you see the lights flash rapidly.

Note: In order to keep your tap tempo **Ping** time when entering *System Settings Mode*, depress either of the other two buttons before pressing the **Ping** button. If you press the **Ping** button first before the other buttons, the **Looping Delay** will register that as a tap and the tempo will change.

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).

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

Time Center: *Auto Mute, Soft Clipping, and Crossfade Time*

Auto Mute:

*Tap **Inf. Hold** to toggle Auto Mute on and off. When the light is on, Auto Mute is enabled.*

Auto Mute is a noise gate that silences the input when a signal is very low. This prevents runaway feedback if **Feedback** is turned up and the module is allowed to run for a long time with no input signal. Turning it off will introduce extra noise, including small amounts of clicking when the gate output jacks fire. It's highly recommended to leave *Auto Mute*.

Soft Clipping:

*Tap **Ping** to toggle Soft Clipping on and off. When the light is on, Soft Clipping is enabled.*

Soft Clipping enables compression when the output signal exceeds 75% of the clipping point. Below this point, the signal is unaffected. This saturation distortion is often more pleasing than harsh clipping.

Crossfade Time:

*Hold down **Reverse** while turning **Time** to 1, 4, 6, 8, 10, 12, or 16 to set the Crossfade Time (CFT).*

- ***Time** at 1: CFT will be 0ms (cross-fading disabled). **Reverse** will periodically flash once.*
- ***Time** at 4: CFT will be 2ms. **Reverse** will periodically flash twice.*
- ***Time** at 6: CFT will be 4ms. **Reverse** will periodically flash three times.*
- ***Time** at 8: CFT will be 8ms. **Reverse** will periodically flash four times.*
- ***Time** at 10: CFT will be 25ms. **Reverse** will periodically flash five times.*
- ***Time** at 12: CFT will be 100ms. **Reverse** will periodically flash six times.*
- ***Time** at 16: CFT will be 250ms. **Reverse** will periodically flash seven times.*

Any time the **Looping Delay's** read and write "heads" jump from one address to another, there is a short cross fade created in order to smooth out the splice. This occurs whenever the **Time** setting is changed and whenever **Reverse** or **Inf. Hold** is toggled. Also, in **Inf. Hold** mode, when the audio reaches the end of the loop it cross-fades back to the start. In most cases, the default timing (8ms) of the cross fade is sufficient and will make the module operate seamlessly. However, advanced users may wish to experiment with other settings.

Disabling *Crossfade Time* allows for instant movement between points, at the expense of creating clicks and pops. This can be ameliorated by enabling *Quantized Change Mode* (see section page 11) because QCM forces the clicks to only occur on the beat. On the other hand, using a longer *Crossfade Time* makes all transitions very smooth and natural sounding, while limiting how quickly the parameters can change. In any setting of CFT and when **Inf. Hold** is on, if the loop size is shorter than the *Crossfade Time*, then cross-fading will be disabled in order to preserve the loop size.

***Time Down:** Reverse/Hold Gate/Trigger and External Clock De-jitter*

Reverse/Hold jacks Gate/Trigger input:

*Gate/Trigger input for **Reverse** jack: Tap **Reverse** to toggle mode (lit=gate, unlit=trigger)*
*Gate/Trigger input for **Hold** jack: Tap **Inf. Hold** to toggle mode (lit=gate, unlit=trigger)*

The input jacks for **Reverse** and **Inf. Hold** can be set to toggle the state when they receive gates or triggers. In *Trigger* mode, it will toggle every time a trigger is received. For example, the first time the **Reverse** jack receives a pulse, **Reverse** will turn on. The next time it receives a pulse, **Reverse** will turn off.
In *Gate* mode, the feature will toggle states every time the jack toggles states. For example, if you send a gate into the **Reverse** jack, **Reverse** will turn on when the gate goes high and stay on until the gate goes low (at which point **Reverse** will turn off). Pressing one of the buttons in either mode will always toggle the state. Additionally, in *Gate* mode, pressing the button will toggle between whether a high gate = On or if a low gate = On. This effectively inverts the gate signal.

Default is *Trigger* mode.

External Clock De-jittering:

*Hold down **Ping** while turning **Time** to 1, 4, 8, 12, or 16 to set the ECD algorithm:*

*   **Time** at 1: <u>Ignore clock pulses that deviate by 0.2ms</u> or less. **Ping** will periodically flash once.*
*   **Time** at 4: <u>Ignore 1% deviation</u> or less. **Ping** will periodically flash twice.*
*   **Time** at 8: <u>Accept all clock pulses</u> (ECD Disabled). **Ping** will periodically flash three times.*
*   **Time** at 12: <u>Moving average</u> the past 2 clock periods. **Ping** will periodically flash four times.*
*   **Time** at 16: <u>Linear average</u> of the past 4 clocks. Ping time only updates every 4 clocks. **Ping** will periodically flash five times.*

The **Looping Delay** has a very precise and jitter-free internal clock (less than 0.0001% at 120BPM). If ever possible, it's recommended to use the **Looping Delay** as the master clock. However, using external gear to clock the **Looping Delay** is also possible. One problem with using external clocks is that lots of equipment generates clocks with a lot of jitter. Sometimes the tempo may vary by as much as 2-3BPM. Since the **Looping Delay** is always recording things to be played back in the future, if the tempo changes from when the audio was recorded to when it plays back, it will sound out of time. In order to compensate for this, the **Looping Delay** has five algorithms that each work with different types of external clock jitter.

If you need to use an external clock that has jitter, it's recommended you patch a simple patch and try each of the five algorithms to see which one suits the particular type of jitter and your patch. Note that in the case of the two Averaging ECD settings, adjusting the external clock speed will cause the **Looping Delay** to slowly “catch-up” before “locking-on”.
The default setting is *Ignore 1% deviation.* With drifting analog clock sources, *Linear Average of 4* is recommended.

Audio Bootloader

The **Looping Delay** contains a bootloader that is used to update the firmware by playing an audio file the **In** jack on the left side of the module. Firmware audio files can be downloaded from the 4ms website at https://4mscompany.com/ld

1. To enter bootloader mode, power off the **Looping Delay** and connect a computer or smart phone audio output to the **In** jack. Either a stereo or mono cable is fine. Connect the **Send** jack to an amp/speakers so you can listen.

2. **Set the computer/phone's volume to 50%**. You may need to adjust it up or down if this is too loud or quiet (see step 5a). Unlike other 4ms modules, the **Looping Delay's** bootloader is designed to work with consumer line-level signal (a peak-to-peak voltage of a little under 1V is ideal, or -10dbV). Turn off all audio notifications that might interrupt playback.

3. Depress the **Reverse** and **Inf. Hold** buttons while powering on the **Looping Delay**. When you see the either button blink, release the buttons.

4. If **Reverse** is blinking, press the **Reverse** button. **Inf. Hold** will start blinking to indicate the module is now ready to receive firmware.

5. Begin playing the file. Immediately you should see **Inf. Hold** and **Ping** lights flash. Do not interrupt the process!

6. The **Inf. Hold** light blinks in a way that tells you if the audio is too loud or quiet. If it blinks erratically, then the audio is too loud or too quiet. If it blinks regularly with equal times on and off, then the audio level is ideal. A good way to set the level is to start with the volume all the way down and notice the the light is off. Then slowly bring it up until it flashes regularly (about four times per second with equal times on and off).

7. If there's an error, the **Reverse** light will start blinking about three times per second, and the **Ping** light will be off. The **Inf. Hold** light will continue to indicate the audio level, so this is a good time to verify the level is ideal and adjust the output level as needed. 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). Stop the audio file, reset it back to the start, and tap the **Reverse** button to reset. The **Inf. Hold** button should blink slowly. Play the file from the beginning again.

8. If the file loads successfully, the **Reverse** button will blink once every two seconds and the **Ping** button will blink every second. Press **Inf. Hold** to start playing.

The open-source licensed source files (in C++, for compiling with arm-none-eabi-g++) can be found at https://www.github.com/4ms/looping-delay

Firmware Version

To view the firmware version, hold down **Reverse** and **Ping** while powering up. After you release the buttons the red light will flash to indicate the firmware version. Power off.

Firmware v1.0 (released June 2023)	Flash once

Hardware Test Mode

The Hardware Test Mode is a way to verify your hardware is working correctly. This procedure is run on every unit at the factory. If you built the **Looping Delay** as a DIY Kit, or if you suspect damage occurred to your unit, you can run this procedure to verify the module is working properly.

The full procedure is described in the *DIY Looping Delay Kit Build Guide* found at https://4mscompany.com/ldkit?manualtab

Factory Reset

A factory reset is not necessary unless you have changed System Settings or recently upgraded firmware versions.
To perform a factory reset, hold down **Reverse** and **Ping** while powering up. After you release the buttons the red light will flash to indicate the firmwar version. The Ping light will then flash. Hold down the Ping button for five seconds until it turns on solid.
The **Looping Delay** has now been reset to its factory default settings. Power off.

Technical Specifications

- 16 HP Eurorack format module
- 0.98" (25mm) maximum depth with power cable

Power consumption:
- +12V rail: 125mA max
- -12V rail: 45mA max
- +5V rail: not used

Audio Inputs
- 20Hz to 20kHz
- 20V peak-to-peak maximum before clipping (when AC coupled)

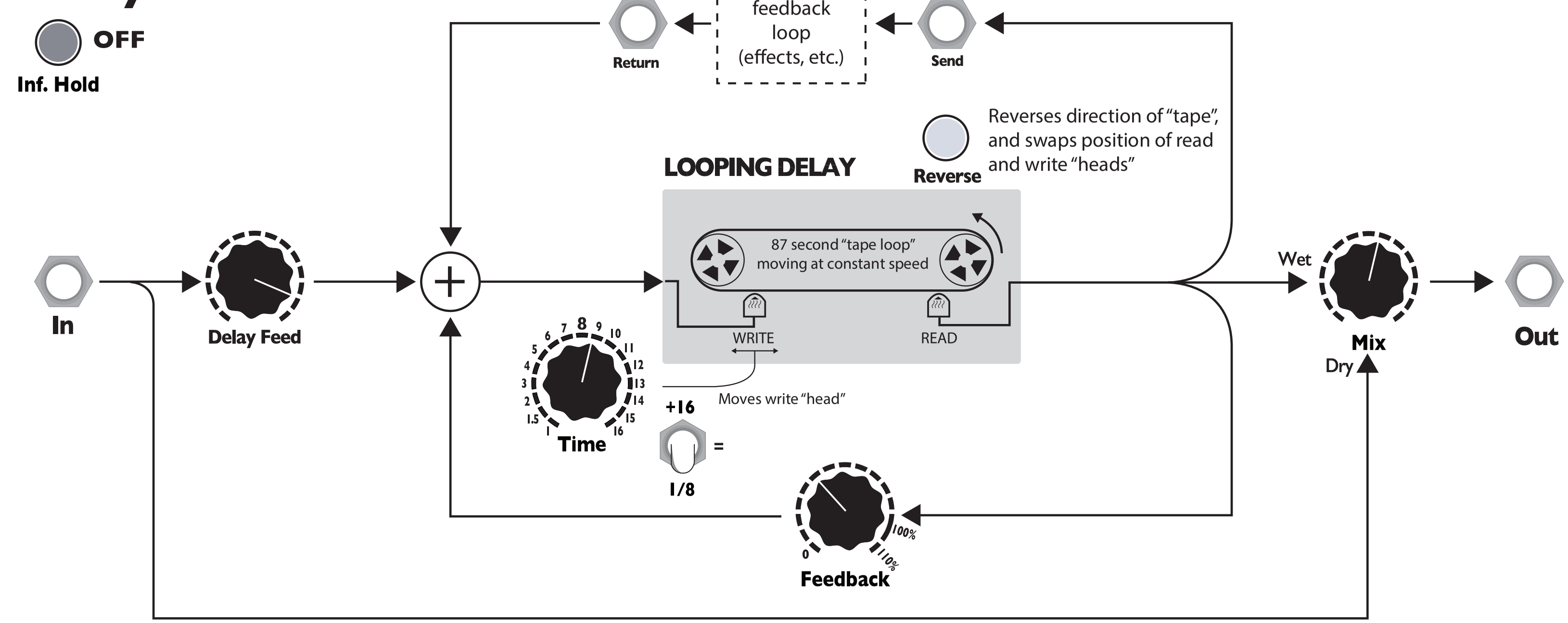
Audio Outputs
- 0Hz (DC) to 24kHz with maximum -1.7dB difference between input and output
- +10V to -10V maximum output

Clock Out:
- 0V to 8V
- +/- 2.4μs maximum jitter (0.001% at 120 BPM)
- Rise/Fall time (10% to 90%): 350μs

Loop Clk Out:
- 0V to 8V
- Rise/Fall time (10% to 90%): 350μs

Sampling
- 24-bit sample at 48kHz, 32-bit processing, 16-bit storage in RAM
- 64Mbit volatile SDRAM chip
- Mono mode: Maximum of 87.38 seconds
- Stereo mode: Maximum of 43.69 seconds

Delay Mode:



Inf. Hold/Loop Mode: