Duophonic Distribution

Quick Specs:

Width: 4hp

Depth: 1.975 inches

Power Draw (max): +12V: 90mA, -12V: 45mA

2 x precision analog sample and holds designed with pitch in mind

Fast acquisition, low droop; perfect for long hold times and audio rate sampling

Momentary track input, available simultaneously

Flip-Flop style clock divider with comparator through

Precision analog full wave rectifier

Internal normalizations for duophonic voice allocation

Schmitt triggers on all gate inputs

Installation:

To install Duophonic Distribution, allocate 4hp of space in your modular case. Before installing the module, examine the ribbon cable and ensure that the red stripe aligns with the **RED STRP (-12V)** indicator on the PCB. Duophonic Distribution comes with a keyed header, making it very difficult to install the ribbon cable backwards. When installing the ribbon cable on the PSU of your modular case, ensure that its red stripe aligns with the **-12V** or **RED STRIPE** indicator on the power supply PCB. Failure to do so can result in permanent damage to Duophonic Distribution, your power supply, and/or other modules in your case. When the ribbon cable is guaranteed to be in its correct position, place the module in your case, screwing the panel to the rails.

What does it do? :

This module features two precision analog sample and holds with momentary track inputs. There is also a flip-flop style clock divider, the input of which is accompanied by a comparator through. A precision full wave rectifier is available for converting bipolar signals to unipolar signals. For maximum flexibility, all of these circuits are available independently, but internal normalizations allow duophonic voice allocation with just a single CV and gate input.

What is a sample and hold? A sample and hold is a utility module that has a signal input, a gate input, and a signal output. The module receives audio or CV at its signal input, but nothing will be present at the output until a gate is received at the gate input. The output signal is equal to the value of the input voltage the moment a trigger is received at the gate input. This output voltage is held indefinitely until another trigger is received at the gate input.

What's the point? This is useful for processing control voltage to create new modulation sources. Imagine applying a saw wave LFO to the signal input. The frequency of the LFO is 1hz. Now imagine applying a clock to the gate input. The frequency of the clock is 4hz. The saw wave will complete one cycle every second, while the clock triggers the sample and hold four times every second. Every time the clock triggers the sample and hold, the voltage level at the output adjusts to the voltage level of the saw wave at that exact moment. This voltage level will be held until the subsequent gate is received from the clock source. By comparing the signal at the input of the sample and hold to the signal at the output of the sample hold, you will see that the smooth and continuous saw wave was transformed into a staircase shaped waveform. The staircase waveform sounds similar to a step sequence when it is used as a modulation source.

How to patch it? :

Duophonic distribution is divided into four sections. The top two sections are identical. They are precision sample and holds with momentary track inputs. Each sample and hold features a signal input, labeled **In,** a signal output, labeled **Out,** a gate input for the sample and hold, labeled **smpl**, and a gate input for the track and hold, labeled **track**. The sample and hold functions exactly how a sample and hold is expected to, as described in the preceding paragraphs. The added track and hold functionality allows the sample and hold circuit to be bypassed. By applying a gate to the **track** input, the signal applied to the signal input is passed directly to the signal output, without being affected by the sample and hold circuitry. When both **smpl** and **track** inputs simultaneously receive gates, the **track** functionality takes precedence.

Imagine that the track and hold was added to the example in the paragraph titled "what's the point?". By applying a gate to the track and hold gate input, you would find the saw wave at the output, rather than the generated staircase waveform. When the gate at the track and hold input goes low, you would again find the staircase waveform at the output.

A basic patch: Find a saw wave or triangle LFO *(a square wave LFO will not work)*. Apply this signal to jack labeled **In.** Find a clock source in your system. It is necessary that the rate of the clock is significantly faster than the rate of the LFO. Apply this clock source to the **smpl** input. Take the output, from the jack labeled **out,** and apply it to either the exponential FM input or the v/octave input of an oscillator. Listen to the output of the oscillator. You will hear its pitch being modulated in a stepwise manner. Adjust the rate of the LFO and of the clock triggering the sample and hold. Listen to how the relative frequencies of these two signals interact with one another and affect the output of the sample and hold.

A little more complex: Now, find a square wave LFO or a clock source that is significantly slower than the clock being applied to the **smpl** input. Another option would be to mult the clock source being applied to the **smpl** input, apply it to a clock divider, and use the output of the clock divider as your second clock source. If you are using a clock divider, try to divide by a factor of eight, if not more. Take this second clock source and apply it to the **track** input. Listen to how it affects the pitch of the oscillator. Whenever there is a gate high at the **track** input, the initial LFO will be unaffected by the sample and hold circuit, resulting in the pitch of the oscillator being modulated smoothly rather than stepwise motion. This example may not be the most musical, but it is useful for demonstrating how the module works. Rather than modulating the pitch of an oscillator, use the final output of the sample and hold to modulate the cut off frequency of a filter or the timbre of a waveshaper.

It is important that the gate signal being applied to the* ***track* *input is truly a gate, rather than a short trigger. The tracking effect is only active as long as the signal at the* ***track*** *input is high, so a short trigger may not hold the* ***track*** *input high for a long enough time for there to be a significant effect**

Duophonic Voice Allocation:

The internal normalizations allow a single pitch sequence to be alternated between two oscillators to create two voice polyphony. Setting this up is very easy to do. You will need a CV pitch sequence with an accompanying gate sequence from either a dedicated sequencer or MIDI to cv converter. You will also need two independent oscillators, two envelopes, and two VCA's or LPG's.

First, tune your two oscillators so that they are the same pitch. Apply your pitch sequence to both oscillators simultaneously. Make any fine tuning adjustments that are necessary so that the oscillators play the sequence in tune with one another.

Second, remove the patch cables from the v/oct inputs on the two oscillators. Now, take the pitch sequence from your sequencer or MIDI to CV converter and apply it to the input (labeled **IN**) of the first sample and hold, at the top of the Duophonic Distribution module. Find the output of the same sample and hold (labeled **OUT**). Use a patch cable to connect this output to the v/oct input of your <u>first</u> oscillator. Find the output of the second sample and hold (labeled **OUT**). Use a patch cable to connect this output to the v/oct input of your <u>second</u> oscillator.

Find the output of the first oscillator and apply it to the audio input of your first VCA or LPG. Find the output of the second oscillator and apply it to the audio input of your second VCA or LPG.

Now, find the third section of the Duophonic Distribution module. You will see four jacks, labeled **IN**, **CMP, FLIP,** and **FLOP.** Take the gate sequence from your sequencer or MIDI to CV converter and apply it to the jack labeled **IN.** Now, take a patch cable and connect it from the jack labeled **FLIP** to the gate input of your first envelope generator. Find the output of the envelope generator and apply it to the CV input of your first VCA or LPG. Now, connect a patch cable to the jack labeled **FLOP**. Connect the opposite end of the patch cable to the gate input of your second envelope generator. Connect the output of this envelope generator to the CV input of your second VCA or LPG.

Take the two audio outputs of your two VCAs or LPGs and mix them together. Then apply this mix to the final output of your modular system. Play your sequence and listen to the final output. Lengthen the decay times of your two envelopes and listen to the notes of your sequence overlap with one another

**** Note*** *- Throughout these instructions it dictates "first oscillator" "second oscillator" "first envelope" "second envelope", etc. It is essential that you are consistent and conscientious of pairing the first output of the sample and hold with the first oscillator, which is then paired with the first VCA/LPG, which is paired with the first envelope, which is triggered from the FLIP output. Do the same for the second oscillator and everything labeled second. **

What is the cmp output? : The input (labeled **IN**) of the clock divider section can be triggered by any signal with an amplitude that is greater than 1.2 volts. Whatever signal is present at this input is internally converted to a gate signal, which is then used to trigger the clock divider. The internally generated gate is supplied at the **CMP** output so that it can be used throughout your modular system. For example, if you apply a triangle wave, LFO or VCO, to the input (labeled **IN**). You will see a square wave at the output. If you apply stepped random voltage to the input (labeled **IN**) you will see randomly generated gates at the **CMP** output.

Why is it labeled CMP?: CMP is short for comparator. The word comparator was too long to fit on the front panel. A comparator is a logic circuit, meaning that it strictly deals with gates. A comparator has two inputs and a single output. If the voltage at the first input is greater than the voltage at the second input, the output provides a gate high (8v). This gate stays high until the voltage at the first input is less than the voltage at the second input. When this happens the output provides a gate low (0v). Internally, 1.2V is tied to the second input of the comparator, meaning that any voltage greater than 1.2V generates a gate high at the **CMP** output.

Patch Tip: *The comparator circuit can be used for PWM. Take an audio rate VCO (do not use a square wave) and mix it with a variable DC offset or modulation source, using a DC coupled mixer. Apply the output of the mixer to the input* ***(IN)*** *of the clock divider section of Duophonic Distribution. Listen to the* ***CMP*** *output to hear a PWM square wave. Using the mixer, adjust the level of your VCO and modulation source to taste. The variable DC offset will act as a manual control for square wave's Pulse Width. This will also provide two square wave sub oscillators at the* ***Flip Flop*** *outputs.*

What is Flip and Flop?: This section is built around a **flip flop** style clock divider. It is a divide by two circuit with two outputs that always maintain an opposing phase relationship. **Flip** and **Flop** are the two gate outputs for the clock divider circuit. Only one of these outputs is high at a given time. They alternate back and forth. Applying a gate at the input (labeled **IN**), causes **flip** to go high (8v) and **flop** to go low (0v). A subsequent gate applied to the input (labeled **IN**), causes **flip** to go low (0v) as **flop** simultaneously goes high (8v). You could think of it as a two step gate sequencer if you want. It flips and then it flops. Flip flop flip flop flip flop flip flop etc.

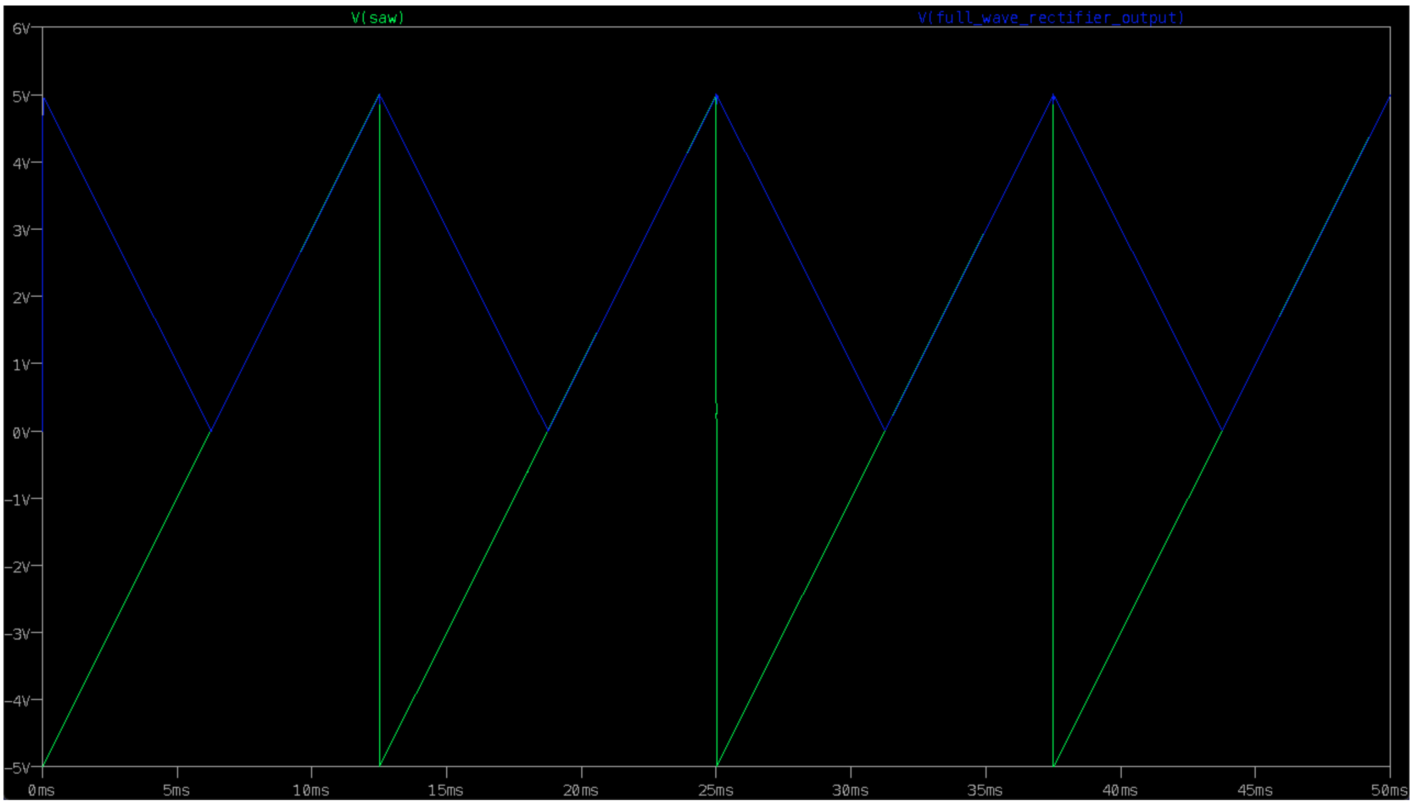
Apply a square, saw, or triangle LFO to the input (labeled **IN**). Patch **FLIP** and **FLOP** to two separate circuits that have LED metering (like a mult, mixer, whatever), or use an oscilloscope if you have one. Watch the LEDs turn on and off in an alternating manner. If you are using an oscilloscope, you will see two alternating square waves. Apply an audio rate waveform to the input to use the **Flip Flop** clock divider to generate sub oscillators.

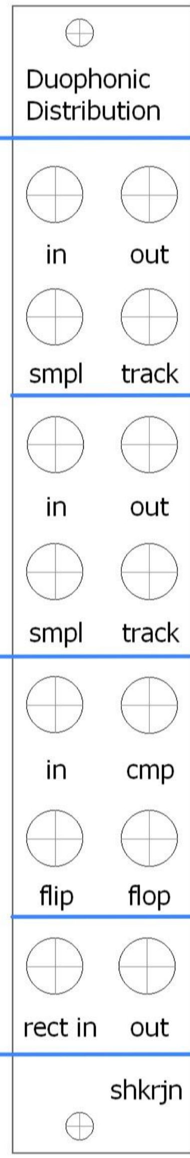
What is a full wave rectifier? At the very bottom of duophonic distribution you will see two jacks, labeled **rect in** and **out. Rect In** is the signal input for the full wave rectifier circuit. Audio or CV can be applied here. **Out** is the output for the full wave rectifier. The best way to think about a full wave rectifier is an absolute value circuit. What's the absolute value of 5? 5. What is the absolute value of -5? Also 5. This circuit allows positive voltage to pass through unaffected, but always inverts negative voltage to its respective positive value.

So what? Some CV inputs only respond to positive voltage. However, most LFO's provide bipolar waveforms that have a range of +/- 5V. Modules that only respond to positive voltage will ignore the negative portion of these LFOs. By first applying these bipolar waveforms to the full wave rectifier, you will enable these modules to respond to the entirety of the LFO.

Furthermore: Full wave rectifiers can have an effect on the waveshape of some waveforms. Apply an audio rate triangle or sine wave to the input of the full wave rectifier. Listen to the output. The output will sound an octave higher than the waveform applied at the input.

Why? Because the negative portions of the waveform have been inverted, there are twice as many peaks within the same amount of time. This translates to the output waveform having twice the frequency of the input waveform. **NOW** apply a saw wave to the input. Listen to the output. You will hear what sounds more like a triangle wave. There is not a concise way to explain why on paper. But drawing it makes it very clear. Full wave rectifiers are great tools that come in handy more often than you expect. They're especially useful when you're modulating the strength and decay parameters of the decay envelopes because those inputs only respond to positive voltages.





Sample and hold 1

In: Input for the sample & hold

Out: Output for the sample & hold

smpl: Gate input for the sample & hold;
Gate high triggers sample functionality

track: Gate input for the momentary track & hold;
When gate is high, the sample & hold output tracks the input

White Noise is internally normalized to in on sample & hold 1

All Gate inputs feature schmitt triggers
Any voltage > 1.2V is gate high

Sample and hold 2

Functionality is identical to sample & hold 1

in on sample & hold 2 is internally normalized to in on sample & hold 1

Comparator and Clock Divider

In: Signal input for comparator and clock divider

Cmp: Output for comparator

Flip: Divide by two gate output for clock divider

Flop: Divide by two gate output for clock divider

Flip is high if flop is low and vice versa

Flip output is internally normalized to smpl input on sample & hold 1

Flop output is internally normalized to smpl input on sample & hold 2

Full Wave Rectifier

Rect in: Signal Input for Full Wave Rectifier

Out: Signal output for Full Wave Rectifier;
Positive Voltages are passed unchanged;
Negative Voltages are inverted to their respective positive value