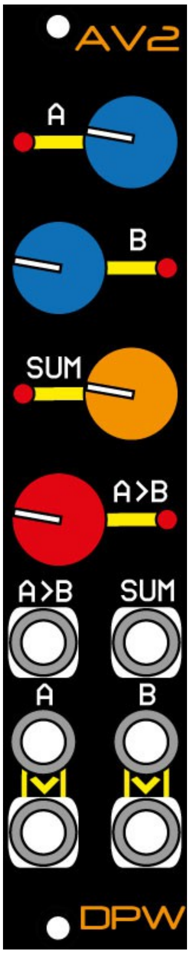
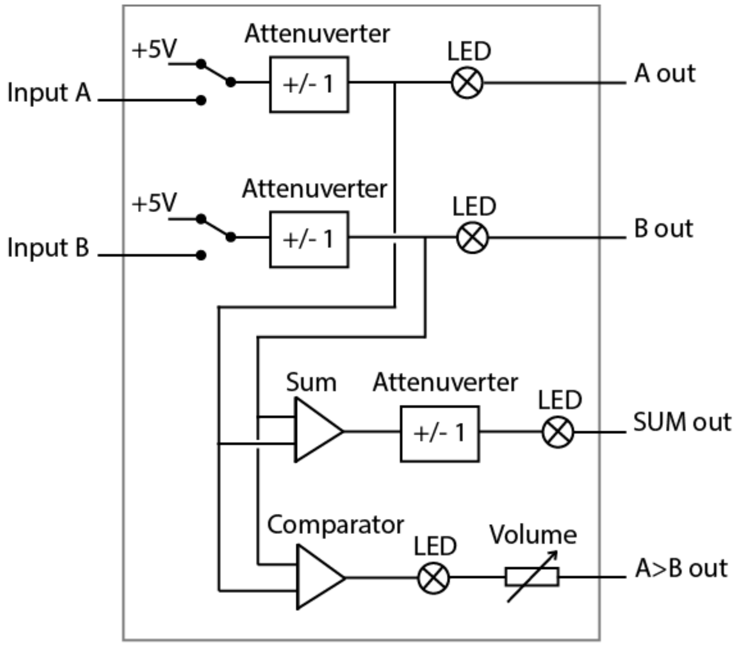
AV2
Attenuverter

The base-function of the AV2 is a dual attenuverter. To the two attenuverters a mixer with an attenuverter on the output and a comparator is connected.

The combination of simple functions makes this a very useful module to have in your toolbox.

The unit can be used as a utility module for control voltages or to run audio through. It is DC coupled.





Input A and B

When nothing is connected to A or B the input of the attenuverter is normalled to +5V.
This gives the possibility to set the output from -5V to +5V DC out via the knob.

When something is connected to A or B that signal can be attenuated or inverted and attenuated via the knob. In the middle position the output is zero. Max clockwise you get 1 times amplification through the module and any signal passes through unaffected. Max counter clockwise the amplification is -1 times. That is 180 degrees out of phase with the input.

The LEDs shows the output voltage. Green is positive and red is negative voltage.

SUM

SUM is the sum of A and B after their attenuverters.
The knob on SUM is an attenuverter as on A and B.

The LED shows the output voltage. Green is positive and red is negative voltage.

A>B

A>B is a comparator that is comparing A and B after the attenuverters.
If A is more positive than B the output A>B will go high. If B is the highest the output will go low.

The output is -5V to +5V and has a volume control.
The output can be used as a logical control signal or as a squarewave audio output that can be PWM modulated depending on A and B.

The LED shows the output voltage. Green is positive and red is negative voltage.
The LED shows the level before the output volume control so you can see how the comparator is reacting before turning up the volume out.

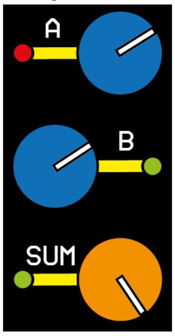
Device specs

Module size:	5 hp wide, 30 mm deep with power cable connected.
Input impedance:	50 kohm
Output impedance:	1 kohm
Bandwidth:	More than 300 kHz.
Power requirements:	+/-12V. Max power consumption 45 mA Connect the power cable with the red stripe down towards the -12V marking. The unit is protected for reverse power.

Use case examples

A few examples of use cases more than just as dual attenuverter, a 2 channel mixer och a comparator. Something to get your imagination going.

Bipolar to unipolar



A normal LFO has an output of +/-5V, that is 10V peak to peak. If you have a module that requires a unipolar signal in you can do that with the AV2. The peak to peak needs to be reduced to 5V and the signal needs to be moved to be just possitive voltages.

Connect your LFO to A and set the potentiometer to halfway between the zero and max clockwise as shown in the picture. You have now set the voltage to half of whats in on A.
Don't connect anything on B and set the potentiometer to the same possition as A. This gives a 2,5V DC voltage out on B.

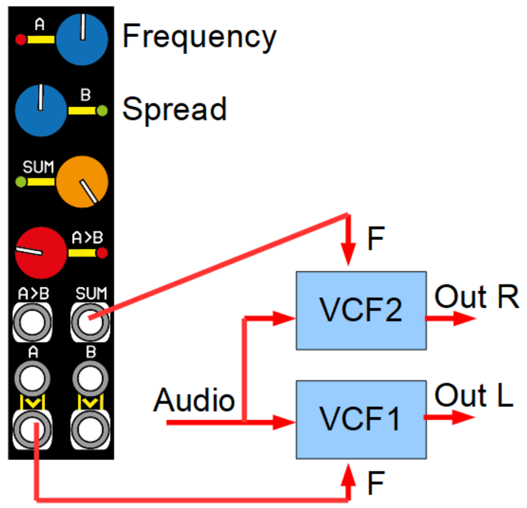
Turn SUM max clockwise and use SUM as your output. Here is your unipolar signal.
If you look at the SUM LED. It shall only be green. If it flashes red when the FLO signal is low, just turn B up a little more to move the signal more away from the negaive side.

PWM modulation

Connect a sine or triangle audio signal to A. Set the A knob to max clockwise position.
Connect a sine or trangle LFO to B. Set the amount of PWM with the B knob.
A>B is the audio out in this case.

If you want an extra fat and wide PWM stereo sound you can use two AV2 modules connected to the same sources and set the controls of the two a bit different.

Wide stereo filter control with spread



This example shows how to filter a mono sound into stereo.

Connect a mono audio signal into two filters.
Connect A out to the cutoff frequency control of VCF1 and SUM to the cutoff frequency control of VCF2.
Set the knobs as shown in the picture.

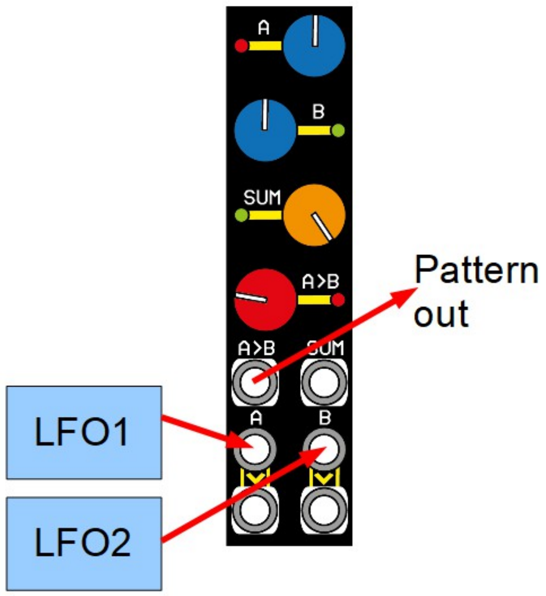
A now controls the cutoff of both filters and B will be the frequency spread control.

To expand the patch:
Turn SUM max counter clockwise. Now when you turn A up the frequency of VCF2 will go down and you will get a stereo effect.

To add movement. Connect an LFO to A in and an other LFO to B in and you can control both stereo filter sweep and spread sweep at the same time.

To make the patch even more interesting. Connect A>B to the resonance control of both filters and set the amount to taste with the knob.

Rhythmic pattern generator



The AV2 can be used to produce pseudo-random like rhythm patterns to trigger other modules. For example drums.

Connect a sine or triangle LFO to A and an other not synced sine or triangle LFO to B.
Use the A>B output as a trigger to some other module.

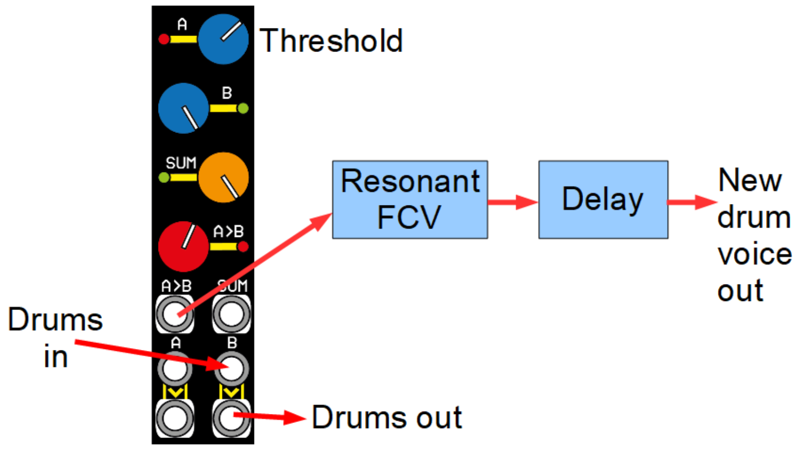
By just varying the setting of A and B you can get at number of interesting rhythmic patterns with variable length that has a base length in multiples of the two LFOs fed to it.
This may sound a bit technical but try it to expirence what it does.

If one of the LFOs are synced with for instance a drum machine this pattern will make rhythmic sense together with the drum machine but still have what seems to be a life of its own.

When set up this way don't forget that you also have the outputs A, B and SUM to modulate other parts of your system.

“Bit reduction” to 1 bit

This is basically a way to use the comparator to turn any aodio into a qsuare wave. The example here is to build an extra voice for the drums by audiorate pinging a resonant filter.



The drums are just passing though the modules B channel here.
The reason for using channel B is that it is easier to access the A knob during a performance.

A sets the threshold voltage for the comparator.
Turn it up a bit so it only reacts to the peaks of the drums.

Set the resonance of the filter just below self oscillation so you can get a good ping sound.

The A>B output will now be used to audiorate ping the filter to create the new voice.

Note that by setting the volume out by the A>B knob you can decide how hard you will ping the filter. The roundness or klickiness of the pinged sound is determined by the amplitude of the ping. This is one of the reasons why there is a volume control on A>B.

The delay after the filter is just to make it a bit more interesting and make tha voice fuller.

You can also connect a sequencer to the frequency cutoff of the filter and get melodic patterns.