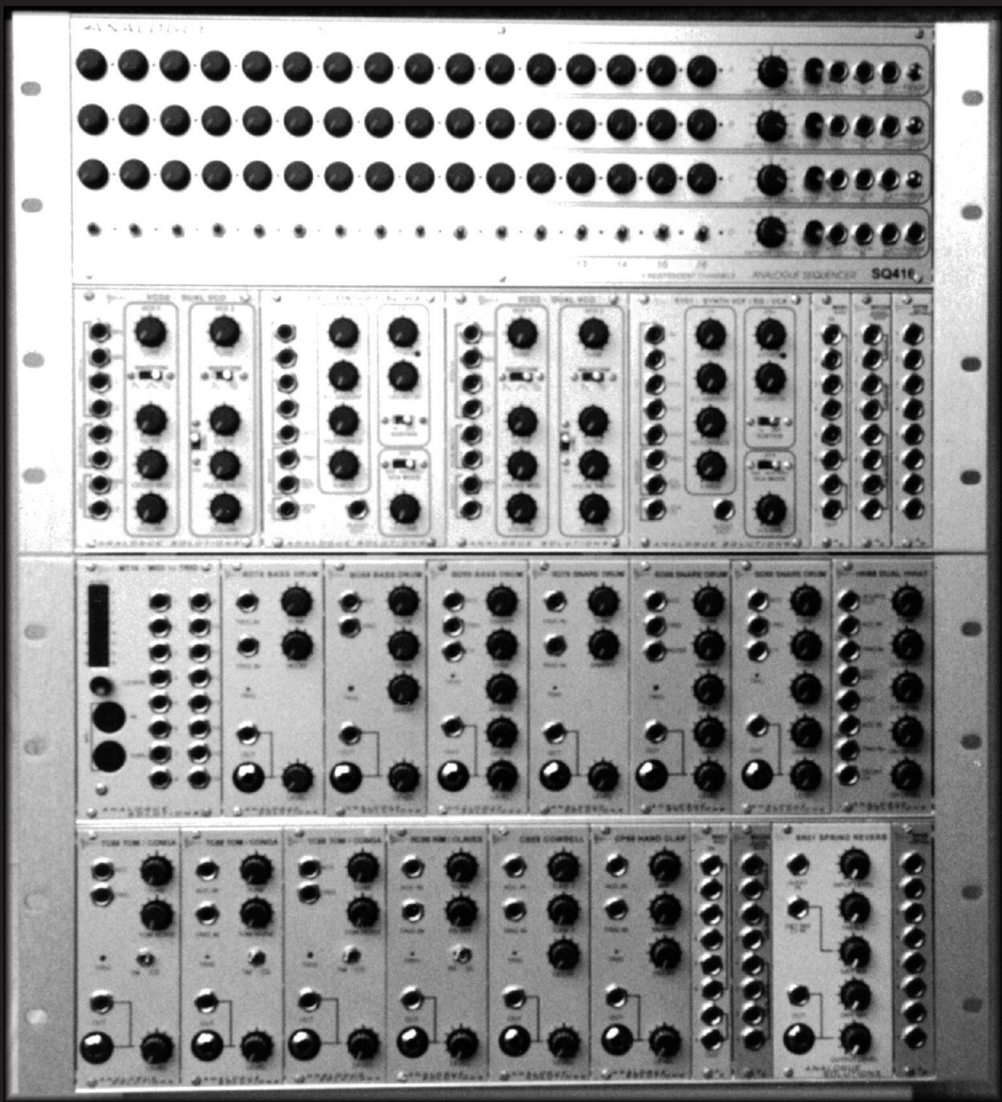
CONCUSSOR & PHOBOS RANGE

synthesis / percussion / sequencing / effects

ANALOGUE SOLUTIONS



modular synthesisers, modular percussion synthesisers, rack effects, oddities

Contents

The Beginning

Before You Start

A modular synthesiser can be daunting for the novice in which case it is a good idea to read through this manual from start to finish. It will give you an idea what this system can do. But due to the limitless possibilities of modulars, not all can be covered. I am sure there are many good books out there or articles on the internet which will give additional tuition. For those already familiar with the principles of modulars, you may just want to flick through the brief outlines of each module, just to familiarise yourself with the less obvious ins and outs of each module.

Safety Instructions

The Concussor should not be operated in the rain or near water and should not be exposed to moisture. If the unit is brought from a cold environment to a warm one, the unit should be left to reach the ambient temperature. This is to allow any possible condensation moisture inside the unit to evaporate. Although any built up moisture will not damage the Concussor, any shorting may be hazardous.

The Concussor does not feature a power switch. One could not really be fitted to the front panel, and it would be impractical to fit one to the rear panel. Most studios have a main power switch or at least a mains switch on each power outlet.

Before connecting the unit to the mains, observe correct voltage supply as indicated on the back panel. 230V for Europe and 115V for USA/Japan.

Always leave the ground connection of the power socket connected to ensure safety.

Always disconnect the mains supply cable from the Concussor when inserting or removing modules. If there are any vacant module slots in the casing, the use of blanking panels is advised to keep straying fingers out of reach of the internal power supply and module electronics.

Refer any servicing to a qualified service personnel.

Introduction

There is no getting away from the digital MIDI dominated world of high-technology. Many write off analogue synth's as being expensive, archaic, and limiting (in terms of MIDI, memories, etc.), and in many ways that is true. But there are two important points that as undeniable. Analogue synth's do still have their place in the modern studio because of the unique way in which they have to be programmed. They force the user to think and create their sounds in a different way (in fact FORCING you to create sounds – you can't rely on presets like so many do with digital gear). Also, the analogue circuitry they use gives more warmth, depth and humanisation to the sounds, a feat DSP constantly tries to mimic, but most would agree never quite cuts it.

Although a good modular system may seem expensive compared to an all sing and dancing digital synth the analogue will usually hold its value and never go out of date. Most digital synth's are out of date within 6 months when the next software update comes out or a newer model is released.

About The Manual

This manual will explain all details about each module, along with some standard and more complex patching examples. With modular synths there are a near limitless range of patching possibilities. We can't possibly tell you all the different patches possible, so if you need further help and information there are many great resources on the web.
The manual is constantly being updated with new patching ideas and other information, so there will never be a final version. Our web site has a copy of the latest manual freely available for download in PDF format.

Unlimted Customer Support

Owners of Concussor receive unlimited phone/fax/email support. We will do our best to help you get the best from your system. If you have a lengthy or complex problem, it is probably best to call rather than email, as some things are easier to explain than type.

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Key To Symbols Used:

Try this, it's good!

Don't try this, not fatal, but not recommened!

Don't try this, could be fatal to yourself or the Concussor!

General note

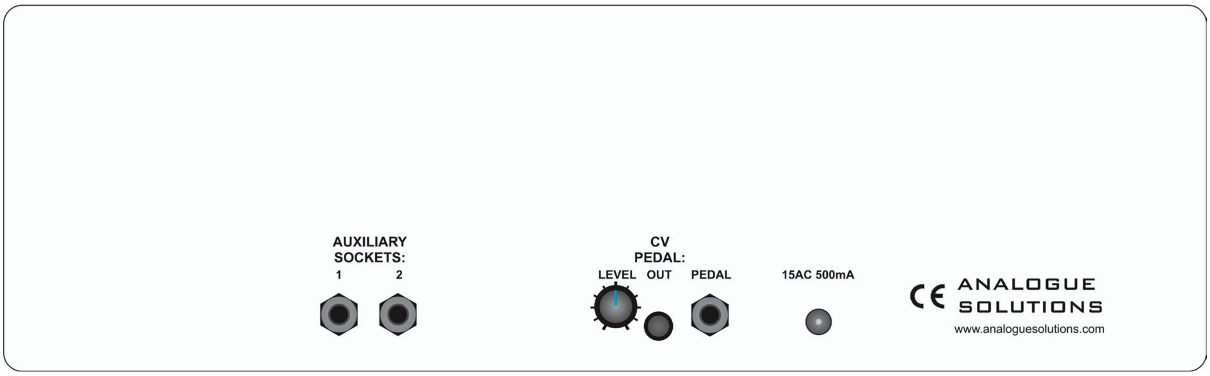
Cases & PSUs

AS0084 84HP Minimodular Case

Module Power Connectors:
9x Doepfer, 3x Integrator

Power input:
5V AC external power supply 500mA (not supplied), 2.1mm DC socket

Width:
84HP



Introduction

This is a lower cost alternative to the pricey aluminium style cases.
The AS0084 will take any combination of Concussor, A100 or Integrator modules with no problems of gaps or lining up because it does not use tapped strips.
Note: Integrator modules are a little wider than Concussor and A100, so it may not be possible to fill the case with only Integrator modules - the total width may be too wide by say 0.5mm.

Power Supply

The power supply board has 9 Doepfer style connectors and 3 Integrator sockets. A 5V power rail is provided for those modules that require it. It uses an external power suppy of 15VAC. This case uses 15V AC (NOT DC!) Use a minimum of 500mA external power supply (not supplied). We can only supply 230V external adaptors with either UK or Euro plugs.
As the power supply requires AC, it does not matter which way round you fit the connector. If you accidentally use a 15DC power supply, it will not cause any damage (but it won't work).
The case will generally supply a full set of modules unless there are too many power hungry modules.

Mounting

The AS0084 case eliminates the use of tapped strips, used in the metal case. This saves a heck of a lot of money, but it does mean you must drill your own mounting holes.
The modules are mounted with self tapping screws into plastic mounting blocks. As plastic is not as strong as metal, these cases are not ideal if you constantly are swapping modules around. Although this can be done many times, the thread in the plastic screw holes can weaken with time with constant screwing and un-screwing.
All that is needed is to position modules in the case, mark the mounting holes, remove modules, drill pilot holes approx 1mm to 2mm diameter, screw modules into position. This is far easier than it sounds, but if it puts you off then pay the full cost for a metal case! (Nearly x2).

Notes: do not overtighten the self tap screws otherwise you will strip the thread in the plastic.

<u>If you are plugging in non Concussor modules - you may void warranty as we cannot be responsible for other manufacturer's faults.</u>

Other Features

The AS0084 is a generic case for use with modules and the Oberkorn sequencer. So the rear panel features may not be wired to or appropriate to what ever module you have fitted in the case.

Apart from the power socket, none of the rear panel sockets and controls are wired to the front panel modules. These are extras that can be wired or patched.

The case comes with A CV pedal function: this is the stereo 6.35mm socket (pedal in), a 3.5mm socket (pedal CV out) and level control knob (pedal CV level).

It also comes with 2 spare (unconnected) mono 6.35mm sockets that can be wired up for your own use.

Optional CV Pedal

Introduction

The AS0084 features a CV pedal input. A special kind of variable level foot pedal can be used to produce a CV. The CV range is 0 to 12V when the Pedal CV control is at maximum. This range can be reduced right from 0-12V to 0-0V.

Use

To use the pedal CV voltage, you must use a CV patch lead (a standard audio lead terminated in 3.5mm jack plugs at both ends). Patch this from the Pedal CV output into one of the front panel CV inputs such as Pitch, Filter Cut-Off or PW.

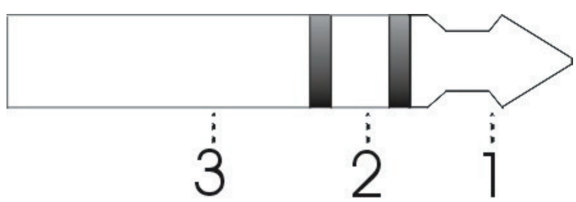
As the pedal is push forwards, the voltage will increase.

The optional pedal we can supply is made of very durable plastic and will suit most musicians. However, we have yet to see whether it will withstand unreasonalbe stomping from heavy metal guitarists!

Any CV pedal that is terminated with a stereo 6.35mm jack plug, and with the correct internal wiring can be used.

The stereo 6.35mm pedal jack plug must have the potentiometer wiper on the tip as shown in the diagrams below.

1 Signal
2 12V
3 Ground



Handling and Inserting Modules

Modules can be placed in any position in the rack. It is a good idea to place one module in at a time and test it before going onto the next. This way if any module causes a problem, it can quickly be identified.

Some modules contain static sensitive circuitry . This circuitry is exposed when not fitted into the rack. When handling any module, try and only handle it by the front panel. Try not to touch any part of the circuitry. It is OK to handle the edge of the circuit board. Always disconnect the rack power supply from the mains when inserting modules or doing anything inside the rack. Note, just a few of the Concussor modules have the power socket up-side-down. This should not cause any confusion as the power cable is always supplied fitted when buying a Concussor module.

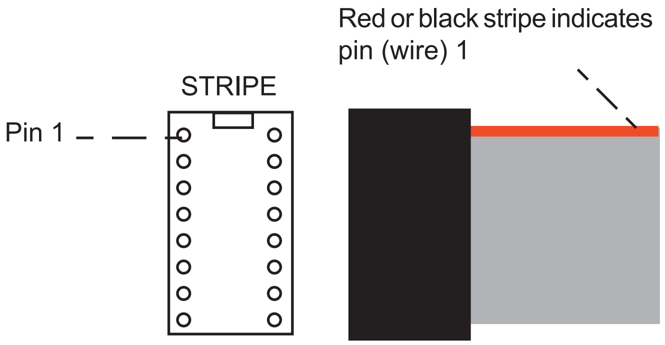
When fitting the SR01 spring reverb, ensure that after mounting the spring unit, that it's lead will reach where you have positioned the SR01 module.

Concussor modules feature both Doepfer and Integrator power sockets. Please state which system you intended to use the Concussor module with so we can supply the correct power lead. Default is Doepfer. The Concussor power supply also features sockets for both. If you run out of sockets on the power supply board, you can purchase 'bus power cables' from us. These are ribbon power leads that feature multiple power sockets - buss cables - enabling many modules to be plugged into only one power supply socket.

It is up to you to connedt modules to PSUs correctly. Whether you have bought one of our cases, power boards or adaptor leads, we cannot guarantee your handy work! Also we certainly won't guarantee our modules if you are mixing and matching manufact-ers. To those who have bought our adaptor cables to connect Integrator modules to a Doepfer PSU, or vice versa:- Our adaptor cables are fully tested. However, we do not guarantee the modules or the power supply under ANY circumstance (including faulty cable).

Please refer to the following diagrams for details of connecting the power lead to the power sockets:

Power socket as used by the Integrator



16 pin power socket
(as featured on the Concussor module, Concussor PSU, and Integrator PSU)

16 pin ribbon power lead - same at both ends.

When connecting the power lead to the socket, ensure correct orientation (stripe to pin 1), and that none of the pins are bent. Make sure all pins seat correctly into the socket.

On the Concussor module, the Integrator power socket will be marked 'AS-POWER' and pin 1 is located by the word 'STRIPE'.

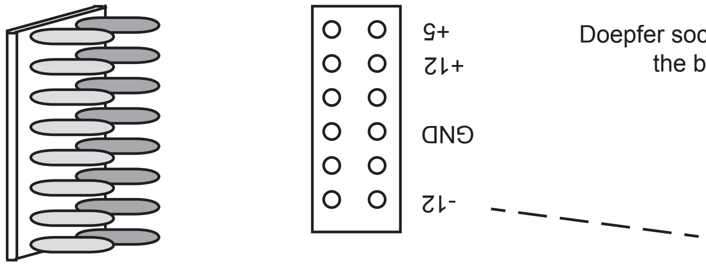
we CAN tell when a module is blown due to incorrect fitting of power lead. Get it right!

Power socket as used by Doepfer.

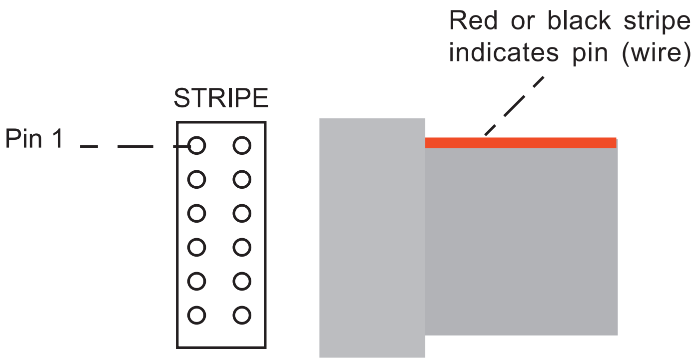
The red stripe on the cable indicates pin 1 (-12V). You may think of pin 1 as being at the top, but Doepfer have place it at the bottom, so Doepfer sockets appear UP SIDE DOWN!

Each power line (+12, -12, & +5) has 2 pairs of pins. The Ground (GND) line uses 2 pairs.

Note: With Doepfer systems, +5 power is an option. Some Concussor modules require +5, so check you have this option installed! With Concussor power supplys, +5 is always supplied as standard.



The socker on the lead may have more holes than there are pins on the PCB. On allConcussor modules, there are 12 pins on the board, and either 14 or 16 holes in the socket. The same applies though - match the striped end of the lead/socket to the work 'STRIPE' on the board. In this case the socket will over hang the board on the opposite end to the stripe end.



16 pin IDC header (as featured on Concussor or Doepfer PSU/Bus board) - this will be 14 pin on the Concussor module.

14 pin IDC socket at module end.
14 pin IDC socket at Doepfer PSU/Bus end.
12 pin IDC socket at Concussor PSU end.

When connecting the power lead to the socket, ensure correct orientation (stripe to pin 1). When the socket has more holes than the header has pins, make sure the pin 1s of the connector line up.

On the Concussor module, the Doepfer power socket will be marked 'DOEPFER' and pin 1 is located by the word 'STRIPE'.

Warning:

The pin outs of our sockets are the same as Doepfer and Integrator. So if you plug them in correctly you will have no problems. Do not expect our sockets to necessarily look the same, or be in the same oreintation as either Doepfer or Integrator. Do not assume anything. Always put the red stripe of the cable towards the word 'STRIPE' on the board. Stripe indicates pin 1. Pin 1 on the cable must go to pin 1 of the socket. Look at what you are doing! Many people pay little attention and just plug the power lead in without checking it is being plugged in correctly.

About Minijack Sockets

The sockets are manufactured in accordance with European DIN dimensions. Many products of Far East origin incorporate plugs that do not meet DIN standards. The main difference is the shape of the jack plug tip. Plugs with very tapered tip profiles may not operate the socket efficiently. To further complicate matters, the US also use their own slightly differing version of the jack plug. These slight variations, though subtle, are enough to cause possible problems, though this happens rarely.

We test all sockets before despatch and they work with the several types of jack plugs we use here.

Percussion Voices

General note about all drum voices:

Trigger input:

a +5 to 12V trigger or gate signal will trigger the voice. The higher the voltage, the louder the voice will play, unless there is a plug in the Accent socket.

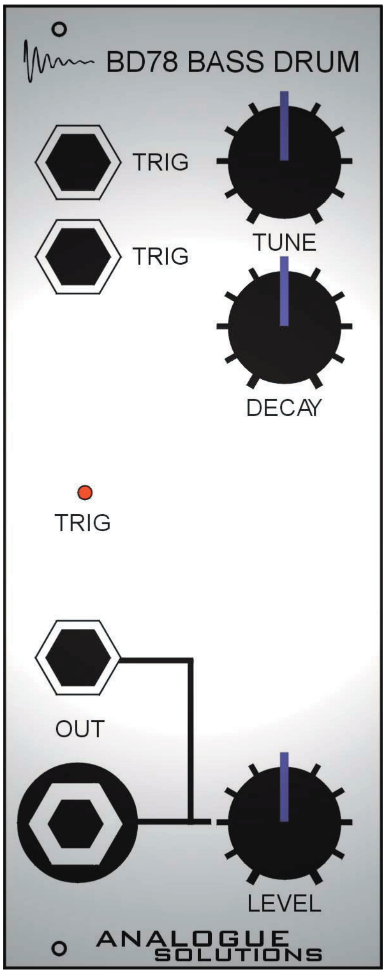
Trigger with our analogue Gate sequencers, or use our MIDI to Trigger (MT16) device for MIDI control.

Accent input:

a 5-12V CV signal controls the overall loudness of the drum sound each time it is triggered.

3.5mm and 6.35mm jack sockets:

with the exception of the HH88 hihat module, all drum voices feature both sizes of jack socket on the audio output. This enables the signal to be re-routed back to the system, or taken straight out to an external mixer or effects unit, without the need for jack adaptors. Jack adaptors are fine, but contacts can be bad, and with 6.35mm socket to 3.5mm plug adaptors, the extra length of the adaptor with the cable plug, combined with cable weight puts extra strain on the module socket.



BD78 Bass/Conga/Bongo Drum Module

Inputs:	Outputs:	Power:
Trigger socket 5-12V x2	3.5mm audio out 6.35mm audio out	+/-12V

Controls:	Indicators:	Size:
Tune Decay Level	Trigger LED	10HP

Introduction

The BD78 is an accurate replica of the CR78 bass drum sound. The module is entirely discrete analogue circuitry, based as closely as possible on the original. The CR78 bass drum circuitry is identical to the CR78 Bongo and Conga sounds, so these can also be produced with the BD78.

The BD78 can produce more than just bass drum sounds...
The BD78 has the addition of a tune control. This enables the pitch to be brought right up. This in conjunction with short decay times allows CR78 style Bongos and Congas to be produced.

The BD78 can be tuned way low with long decays... watch your bass bins mate!

Use

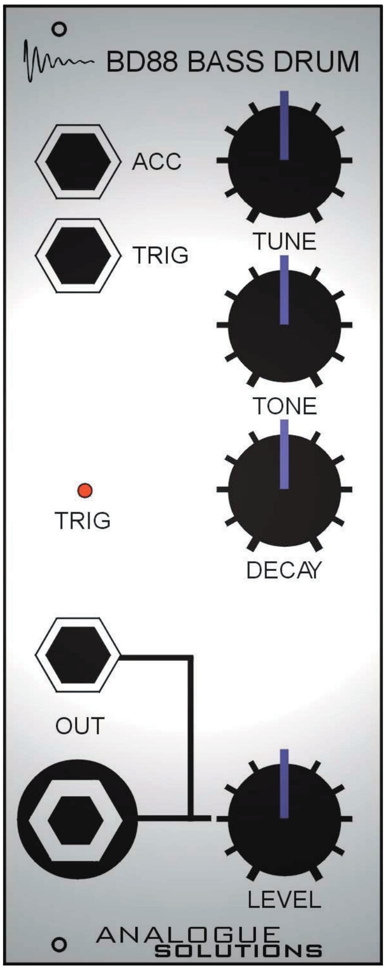
The trigger input has a range of about 5 to 12V. A trigger or gate signal of at least 5V will trigger the sound. A larger trigger pulse will increase the level of the sound (accent the notes). Note, the BD78 does not have a separate accent input.

The trigger LED will light each time a trigger pulse is received.

Controls

Tune Tunes the bass drum up or down
Decay Sets decay time
Level Sets output level

Due to the way the CR78 bass drum circuitry works (using a tuned-resonance ciruit), the Tune and Decay controls affect each other. At certain combinations of settings, the BD78 may go into self-oscillation, with loud results! The BD78 circuitry was never designed to have manually variable Tune and Decay controls, and as we wish to retain the original sound, this by-product has been left in and cannot be deisgned out anyway without such a re-design it would no longer be the same drum sound. Also, you may get some small noise when you rotate Decay pot. This is normal and not a fault with the control.



BD88 Bass/Tom/Conga Drum Module

Inputs:	Outputs:	Power:
Accent socket 0-12V	3.5mm audio out	+/-12V
Trigger socket 5-12V	6.35mm audio out	

Controls:	Indicators:	Size:
Tune	Trigger LED	10HP
Tone		
Decay		
Level		

Introduction

The BD88 is an accurate replica of the TR808 bass drum sound. The module is entirely discrete analogue circuitry, based as closely as possible on the original.

The BD88 can produce more than just bass drum sounds...
The BD88 has the addition of a tune control. This enables the pitch to be brought right up. This in conjunction with short decay times allows TR808 style Toms and Congas to be produced.

The BD88 can be tuned way low with long decays... watch your bass bins mate!

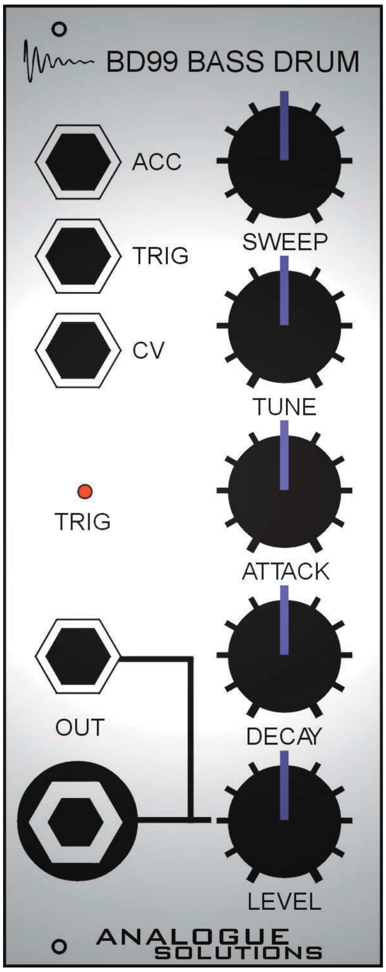
Use

The trigger input has a range of about 5 to 12V. A trigger pulse of around 1mS is recommended, although the gate output of a synth can be used. A larger trigger pulse will increase the level of the sound (accent the notes). When the accent socket is used, the trigger input socket only triggers the sound and the level of the sound is controlled by the voltage of the CV at the accent socket. The accent voltage can be from another trigger pulse generator, or from an LFO or Envelope. When using for example an LFO, you can have the level changing with the LFO sweep.

The trigger LED will light each time a trigger pulse is received.

Controls

Tune Tunes the bass drum up or down
Tone Controls 'click' level (like stick hitting drum skin). The click you hear is actually the trigger pulse.
Decay Sets decay time
Level Sets output level



BD99 Bass/Elec.Tom Drum Module

Inputs:	Outputs:	Power supply:
Accent socket 0-12V	3.5mm audio out	+/-12, 5V
Trigger socket 5-12V	6.35mm audio out	
Tune CV 0-12V		

Controls:	Indicators:	Size:
Tune1 (sweep)	Trigger LED	10HP
Tune2		
Attack		
Decay		
Level		

Introduction

The BD99 is an accurate replica of the TR909 bass drum sound. The module is entirely discrete analogue circuitry, based as closely as possible on the original.

The BD99 can produce more than just bass drum sounds...

The original TR909 has a tune control. This really sets envelope sweep on the pitch, so we have named this control Sweep on the BD99 for accuracy. We have added a true tuning control which enables the pitch to be changed. The pitch can be brought right up, enabling electronic Tom and Conga style sounds to be produced.

Also, by modulating the pitch via the Tune CV input with a high frequency signal (CV or Audio), new and excinting waveforms can be produce. This radically changes the sound of the BD99, effectively turning it into a drun synth.

The BD99 can be tuned way low... watch your bass bins mate!

Use

The trigger input has a range of about 5 to 12V. A trigger or gate signal of at least 5V will trigger the sound. A larger trigger pulse will increase the level of the sound (accent the notes). When the accent socket is used, the trigger input socket only triggers the sound and the level of the sound is controlled by the voltage of the CV at the accent socket. The accent voltage can be from another trigger pulse generator, or from an LFO or Envelope. When using for example an LFO, you can have the level changing with the LFO sweep.

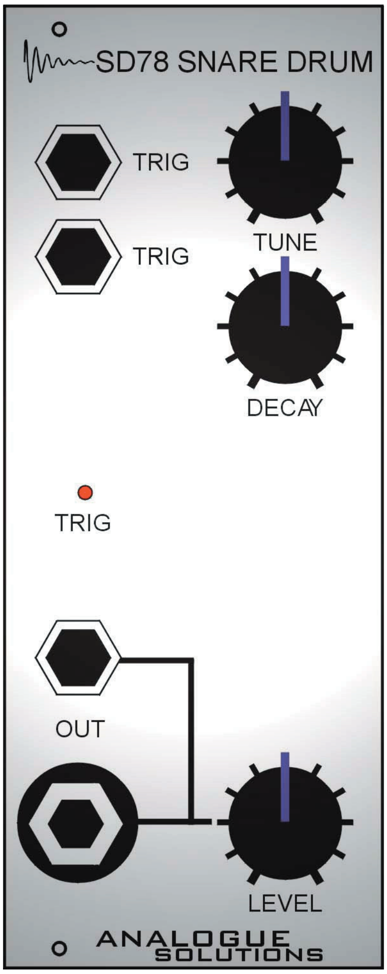
The Tune CV input can be used to alter the tuning with any voltage. E.g. use LFO for slow pitch sweeps. Using audio or any other high frequency signals to modulate tuning allows wild cross-modulated drum sounds to be produced. This sort of modulation of pitch provides an extreme range of sounds to be produced, aside from the standard TR909 style kick.

The trigger LED will light each time a trigger pulse is received.

Controls

Sweep Called 'Tune' on original TR909. Sets the amount of envelope sweep on the pitch.
Tune Sets the initial pitch of the drum sound
Attack Controls 'click' level (like stick hitting drum skin). The click you hear is actually a burst of filtered noise.
Decay Sets decay time
Level Sets output level

Note: the BD99 12pin power header is mounted up-side-down. Always ensure power cable is fitted correctly.Doefper power socket is upside down on this module, though this should not cause confusion as the module is always supplied with the power cable fitted.



SD78 Snare Drum Module

Inputs:	Outputs:	Power supply:
Trigger socket 5-12V	3.5mm audio out 6.35mm audio out	+/-12V

Controls:	Indicators:	Size:
Tune Snappy	Trigger LED	10HP

Introduction

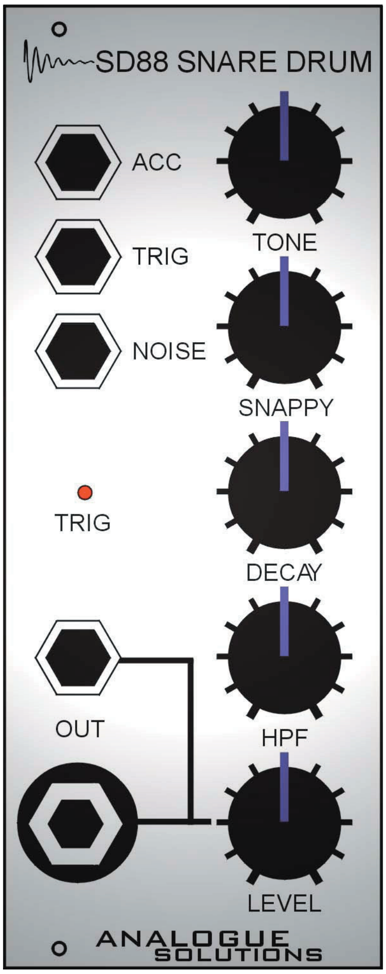
The SD78 is an accurate replica of the CR78 snare drum sound. The module is entirely discrete analogue circuitry, based as closely as possible on the original.

Use

The trigger input has a range of about 5 to 12V. A trigger or gate signal of at least 5V will trigger the sound. A larger trigger pulse will increase the level of the sound (accent the notes). Note, the SD78 does not have a separate accent input.

Controls

Tune Changes the tuning of the snare sound.
Snappy Sets the noise level.
Level Sets output level



SD88 Snare Drum Module

Inputs:	Outputs:	Power supply:
Accent socket 0-12V	3.5mm audio out	+/-12V
Trigger socket 5-12V	6.35mm audio out	
	Noise out	

Controls:	Indicators:	Size:
Tone	Trigger LED	10HP
Snappy		
Decay		
HPF (High Pass Filter)		
Level		

Introduction

The SD88 is an accurate replica of the TR808 snare drum sound. The module is entirely discrete analogue circuitry, based as closely as possible on the original. The snare sound is made up of two oscillators mixed with filtered noise.

The SD88 can produce more than just snare drum sounds...

The SD88 has additional controls to change noise decay (Decay) and High Pass Filter cut-off (HPF) on the filtered noise. This enables a wide range of snares, noise hits, and bongo/conga type sounds

Use

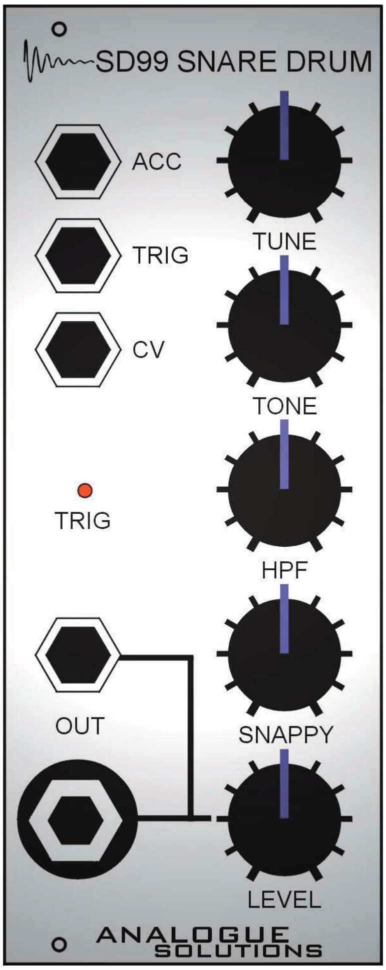
The trigger input has a range of about 5 to 12V. A trigger or gate signal of at least 5V will trigger the sound. A larger trigger pulse will increase the level of the sound (accent the notes). When the accent socket is used, the trigger input socket only triggers the sound and the level of the sound is controlled by the voltage of the CV at the accent socket. The accent voltage can be from another trigger pulse generator, or from an LFO or Envelope. When using for example an LFO, you can have the level changing with the LFO sweep.

The white noise which is the source of the 'snappy' part of the sound is available at the NOISE output directly and unprocessed.

The trigger LED will light each time a trigger pulse is received.

Controls

Tone	Sets the mix between the 2 oscillators that make up part of the sounds.
Snappy	Sets the noise level.
Decay	Sets decay time of the noise.
HPF	Alters filter cut-off of the noise.
Level	Sets output level



SD99 Snare Drum Module

Inputs:	Outputs:	Power supply:
Accent socket 0-12V	3.5mm audio out	+/-12V
Trigger socket 5-12V	6.35mm audio out	
Tune CV 0-12V		

Controls:	Indicators:	Size:
Tune	Trigger LED	10HP
Tone		
HPF		
Snappy		
Level		

Introduction

The SD99 is an accurate replica of the TR909 snare drum sound. The module is entirely discrete analogue circuitry, based as closely as possible on the original.

The SD99 can produce more than just snare drum sounds...

Use

The trigger input has a range of about 5 to 12V. A trigger or gate signal of at least 5V will trigger the sound. A larger trigger pulse will increase the level of the sound (accent the notes). When the accent socket is used, the trigger input socket only triggers the sound and the level of the sound is controlled by the voltage of the CV at the accent socket. The accent voltage can be from another trigger pulse generator, or from an LFO or Envelope. When using for example an LFO, you can have the level changing with the LFO sweep.

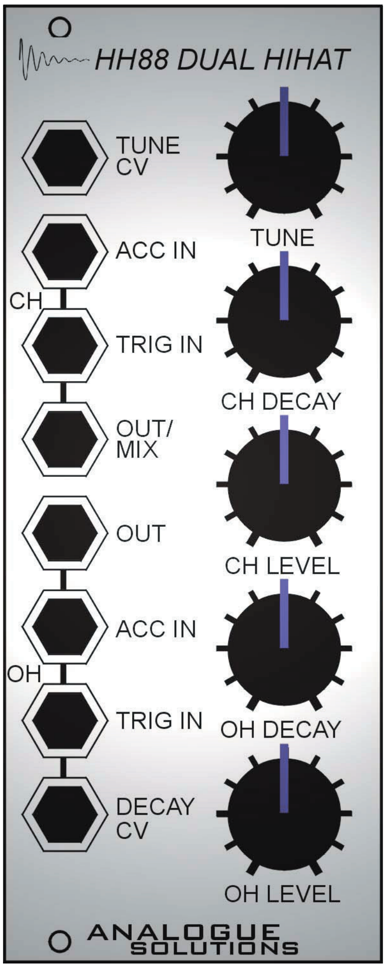
The Tune CV input can be used to alter the tuning with any voltage. E.g. use LFO for slow pitch sweeps. Using audio or any other high frequency signals to modulate tuning allows wild cross-modulated drum sounds to be produced. This sort of modulation of pitch provides an extreme range of sounds to be produced, aside from the standard TR909 style snare.

The trigger LED will light each time a trigger pulse is received.

Note; when the HPF control is set very high, some noise may low level hiss through. This is because the HPF is being set far beyond the range originally intended. We have allowed the range to be extended in this way to allow a much wider range of sounds to be produced. Remeber, the original 909 sound never allowed you to manually adjust the HPF cut-off!

Controls

Tune Tunes the two oscillator ‘tone’ sounds.
Tone Sets the amount of noise. Higher setting increases the hold and decay time of the noise.
HPF Sets the cut-off level of the filter on the noise.
Snappy Sets the noise level.
Level Sets output level.



HH88 Dual HiHat Drum Module

Inputs:	Outputs:
CH Accent socket 0-12V	CH audio out
CH Trigger socket 5-12V	OH audio out
OH Accent socket 0-12V	Source
OH Trigger socket 5-12V	
OH Decay CV 0-12V	**Power supply:**
	+/-12, 5V

Controls:	Indicators:
Tune	None
CH Decay	
CH Level	
OH Decay	**Size:**
OH Level	10HP

Introduction

The HH88 is an accurate replica of the TR808 hihat drum sound. The module is entirely discrete analogue circuitry, based as closely as possible on the original.

The HH88 has addition of a Tune control for both, and Decay on the closed hihat sound.

The two hihat sounds are made up of six filtered square wave oscillators, fed through separate VCA's controlled by envelopes (a VCA and EG per hihat sounds).

The HH88 can produce more than just hihat drum sounds...
The HH88 has the addition of a decay control for the closed hihat, and the decay time for the open hihat has been extended. This enables TR808 cymbal style sounds to be produced.

Use

The trigger input has a range of about 5 to 12V. A trigger or gate signal of at least 5V will trigger the sound. A larger trigger pulse will increase the level of the sound (accent the notes). When the accent socket is used, the trigger input socket only triggers the sound and the level of the sound is controlled by the voltage of the CV at the accent socket. The accent voltage can be from another trigger pulse generator, or from an LFO or Envelope. When using for example an LFO, you can have the level changing with the LFO sweep.

When both hihat sounds receive a trigger at the same time, a muted hihat sound is produced.

The HH audio source - which is six detuned square wave tones - is available at the SOURCE output directly and unprocessed. The Tune control changes the tuning of one of the six square wave tones.

The OUT sockets are the audio outputs for the CH and OH sounds.

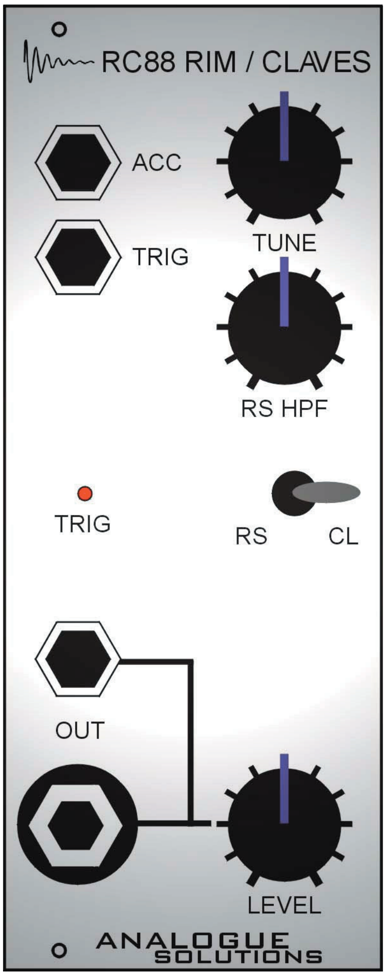
When using the (OH) Decay CV input, turn the OH Decay pot fully clockwise. The CV input subtracts from the control.

The SOURCE output is the source sound of the hihats, unfiltered. Use this in conjuction with a modular system to create new drum sounds.

Controls

Tune	Sets tuning of 1 of the 6 oscillators that makes up the sounds.
CH Decay	Sets the closed hihat decay time.
CH Level	Sets closed hihat output level.
OH Decay	Sets the open hihat decay time.
OH Level	Sets open hihat output level.

Note: The 'MIX' feature of the CH Output is no longer utilised. This is because the mixer is a passive type and it causes a significant drop in signal level when the CH and OH were mixed through this output. Many customers prefer the higher signal lever obtained by omiting the mixer.



RC88 Rimshot / Claves Module

Inputs:	Outputs:	Power:
Accent socket 0-12V	3.5mm audio out	+/-12V
Trigger socket 5-12V	6.35mm audio out	

Controls:	Indicators:	Size:
Tune	Trigger LED	10HP
Rimshot tone (HPF)		
Rimshot / Clave selector switch		
Level		

Introduction

The RC88 is an accurate replica of the TR808 rimshot and claves sounds. The module is entirely discrete analogue circuitry, based as closely as possible on the original.

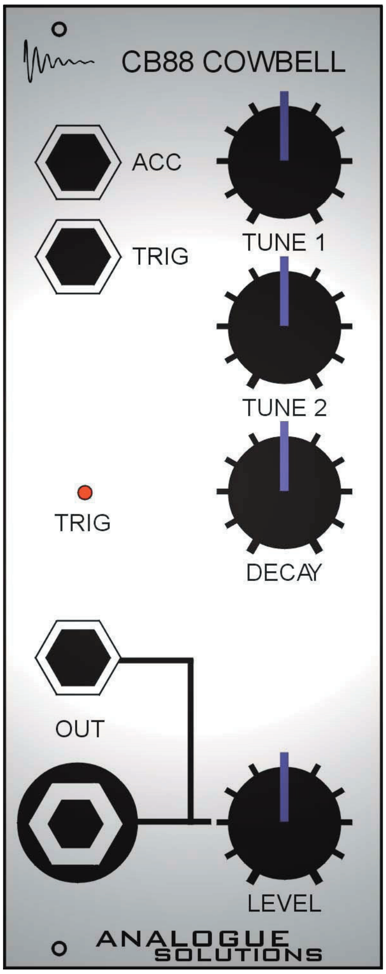
Use

The trigger input has a range of about 5 to 12V. A trigger or gate signal of at least 5V will trigger the sound. A larger trigger pulse will increase the level of the sound (accent the notes). When the accent socket is used, the trigger input socket only triggers the sound and the level of the sound is controlled by the voltage of the CV at the accent socket. The accent voltage can be from another trigger pulse generator, or from an LFO or Envelope. When using for example an LFO, you can have the level changing with the LFO sweep.

The trigger LED will light each time a trigger pulse is received.

Controls

Tune	Tunes the sound up or down
RS HPF	Changes the tone of the rimshot sound by adjusting the cut-off of an internal HPF.
Switch	This selects either the rimshot (RS) or the claves (CL) sound.
Level	Sets output level



CB88 Cowbell Module

Inputs:	Outputs:	Power:
Accent socket 0-12V	3.5mm audio out	+/-12V, 5V
Trigger socket 5-12V	6.35mm audio out	

Controls:	Indicators:	Size:
Tune 1	Trigger LED	10HP
Tone 2		
Decay		
Level		

Introduction

The CB88 is an accurate replica of the TR808 cowbell sound. The module is entirely discrete analogue circuitry, based as closely as possible on the original. The sound is made up from 2 square wave oscillators whose pitch can manually be adjusted. To create the cowbell sound though, they are tune close together so they beat.

The CB88 can produce more than just cowbell sounds...
By tuning the 2 oscillators to equal pitch a synth sound is produced, which can be tuned to the melodies in your music.

Use

The trigger input has a range of about 5 to 12V. A trigger or gate signal of at least 5V will trigger the sound. A larger trigger pulse will increase the level of the sound (accent the notes). When the accent socket is used, the trigger input socket only triggers the sound and the level of the sound is controlled by the voltage of the CV at the accent socket. The accent voltage can be from another trigger pulse generator, or from an LFO or Envelope. When using for example an LFO, you can have the level changing with the LFO sweep.

The trigger LED will light each time a trigger pulse is received.

Controls

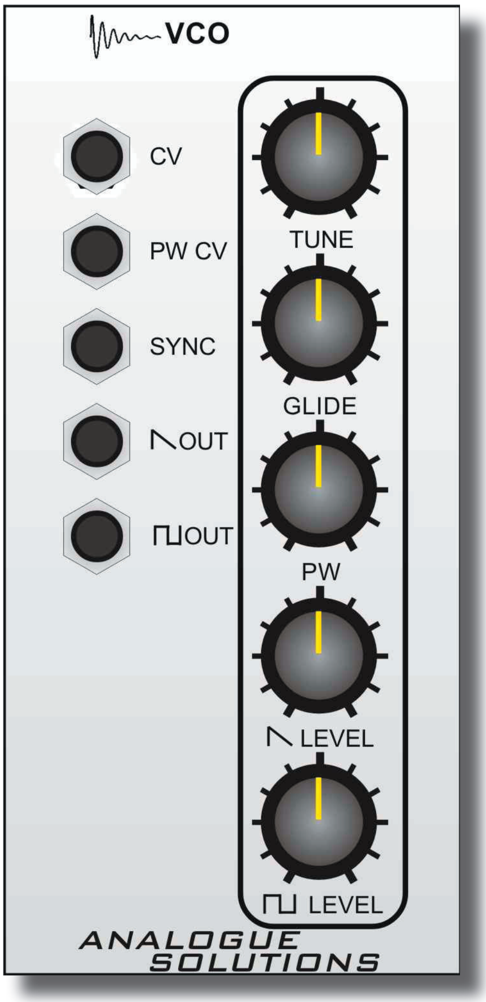
Tune 1	Tune oscillator 1 up or down
Tune 2	Tune oscillator 2 up or down
Decay	Sets decay time
Level	Sets output level

Synthesizer Modules

VCO - Voltage Controller Oscillator

Introduction

VCO is a single VCO laid out in a similar way to the VCO section you would find on a monosynth. VCO includes a glide (slew) function, so this is a great bonus. The VCO can go down to low frequencies so they can be used as an LFO.



Voltage Controlled Oscillators (VCOs) in general

The VCO or signal generator, commonly found on most music synthesizers consist of an exponential amplifier, a resetable integrator and waveshaping circuitry. 'Voltage Controlled Oscillator' means an oscillator whose frequency can be controlled by an external voltage.

A VCO is responisble for the pitch of a sound and its basic timbre. Most VCOs will offer a variety of wave forms. Some VCOs offer the facility to alter the pulse width of a square wave either manually or with a voltage (pulse width modulation).

Controls In Detail

Tune

The VCO has a wide ranging tune control. Turning Tune alters the basic pitch of the oscillator.

Glide

Turning up Glide adds portamento to the oscillator, so the pitch will bend (slide) up or down to each note (as opposed to a sudden change). The higher the setting, the longer the oscillator takes to reach its new note. Not many VCO modules give you the all important Glide control as standard!

Pulse Width

You can alter the pulse width (duty cycle) of the square wave. Turning this gives a sound similar to chorusing. In the centre position a square wave is produced.

Sawtooth Level

This sets the output volume of the sawtooth waveform output.

Square Level

This sets the output volume of the square waveform output.

Sockets in Detail

CV in

This socket is the pitch CV control input and is used to control the pitch of the oscillator.

PW

This is a pulse width control CV input. Use a -12 to +12V voltage in this socket to alter the pulse width of the oscillator square wave.

Sync In

This is an oscillator Sync reset input.

Sawtooth Out

This is the audio output for the sawtooth waveform. Its level is affected by the sawtooth level pot.

Square Out

This is the audio output for the square waveform. Its level is affected by the square level pot.

General Specification

Inputs
- CV
- PW
- SYNC

Outputs
- SAW
- SQUARE

Controls
- Tune
- Glide
- Pulse Width
- Saw Level
- Square Level

Size
- 12HP

Power
- +/-12V, GND using Doepfer power socket.

VCO-RM - Ring Modulation / VCO

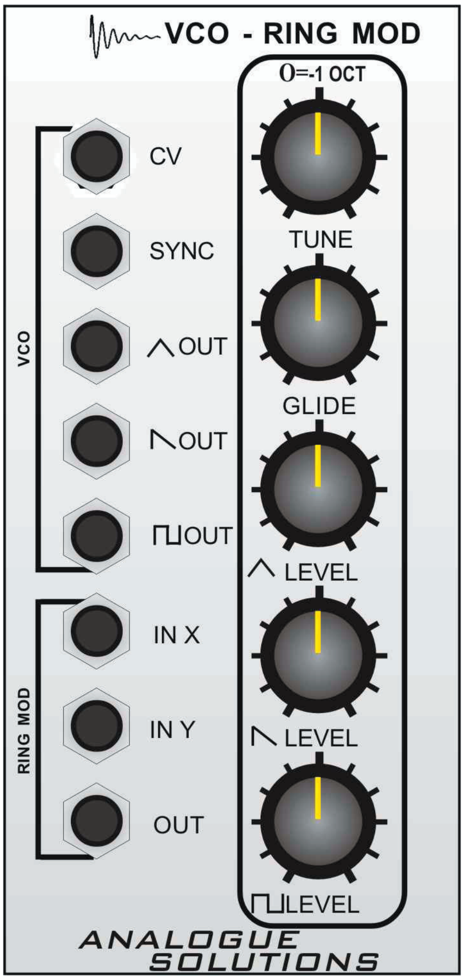
Introduction

VCO-RM is a single VCO-RM laid out in a similar way to the VCO section you would find on a monosynth. VCO-RM includes a glide (slew) function, so this is a great bonus. VCO-RM can go down to low frequencies so they can be used as an LFO

It also has a complete and independent Ring Modulator. This can be used on its own, or you can patch the oscillator into the RM, and along with another audio source gives you great Ring Mod'd sounds.

Ring Modulation

VCO-RM features a high quality Ring Modulator very similar to those found in the Roland System 100m, EMS VCS3 and ARP2600.



What the RM does is take 2 audio inputs, which we will call X and Y, then produce an output that contains the original frequencies X and Y, as well as the sum and differences of X and Y (X+Y, X-Y).

In non-technical speak, the RM best produces metallic sounding bell and clang sounds. The output is rich in harmonics and can often sound quite abbrasive. So filtering is necessary and the sound is best used for effects. The RM is a vital processor producing it's own unique sound vital in an system.

RM In Use

Basically experiment with any two audio sources. The type of input signal (waveform) and pitch will alter the output. By slowly changing the pitch of one input some dramatic sounds can be produced. Note, the RM is completely independent from the VCO.

Controls In Detail

Tune

The VCO has a wide ranging tune control. Turning Tune alters the basic pitch of the oscillator.

Glide

Turning up Glide adds portamento to the oscillator, so the pitch will bend (slide) up or down to each note (as opposed to a sudden change). The higher the setting, the longer the oscillator takes to reach its new note. Not many VCO modules give you the all important Glide control as standard!

Triangle Level

This sets the output volume of the triangle waveform output.

Sawtooth Level

This sets the output volume of the sawtooth waveform output.

Square Level

This sets the output volume of the square waveform output.

Sockets in Detail

CV in

This socket is the pitch CV control input and is used to control the pitch of the oscillator.

Sync In

This is an oscillator Sync reset input.

Triangle Out

This is the audio output for the Triangle waveform. Its level is affected by the Triangle level pot.

Sawtooth Out
This is the audio output for the sawtooth waveform. Its level is affected by the sawtooth level pot.
Square Out
This is the audio output for the square waveform. Its level is affected by the square level pot.
RM X input
The X audio input to the Ring Modulator
RM Y input
The Y audio input to the Ring Modulator
RM output
The RM output

General Specification

<u>Inputs</u>
CV
PW
RM X
RM Y

<u>Outputs</u>
TRIANGLE
SAW
SQUARE
RING MOD

<u>Controls</u>
Tune
Glide
Pulse Width
Saw Level
Square Level

<u>Size</u>
12HP

<u>Power</u>
+/-12V, GND using Doepfer power socket.

VCO-SUB - Voltage Controller Oscillator

Introduction

VCO is a single VCO laid out in a similar way to the VCO section you would find on a monosynth. VCO includes a glide (slew) function, so this is a great bonus. The VCO can go down to low frequencies so they can be used as an LFO.

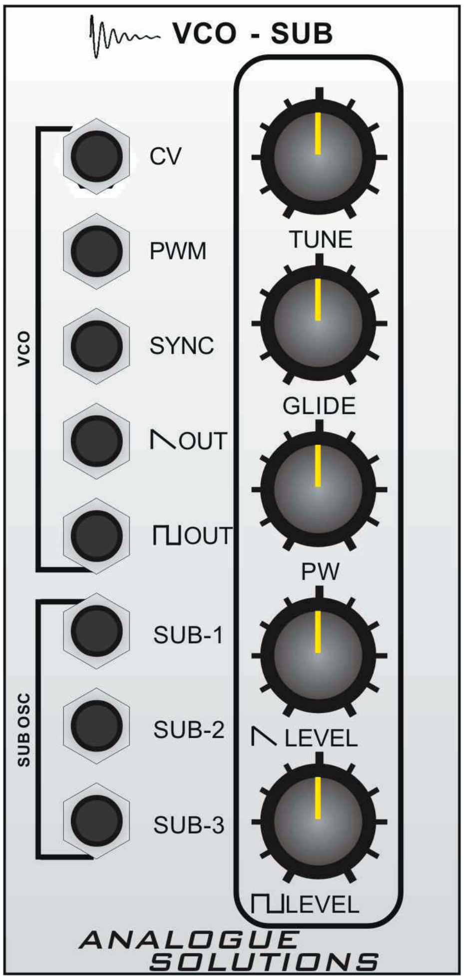
Controls In Detail

Tune

The VCO has a wide ranging tune control. Turning Tune alters the basic pitch of the oscillator.

Glide

Turning up Glide adds portamento to the oscillator, so the pitch will bend (slide) up or down to each note (as opposed to a sudden change). The higher the setting, the longer the oscillator takes to reach its new note. Not many VCO modules give you the all important Glide control as standard!



Pulse Width

You can alter the pulse width (duty cycle) of the square wave. Turning this gives a sound similar to chorusing. In the centre position a square wave is produced.

Sawtooth Level

This sets the output volume of the sawtooth waveform output.

Square Level

This sets the output volume of the square waveform output.

Sockets in Detail

CV in

This socket is the pitch CV control input and is used to control the pitch of the oscillator.

PW

This is a pulse width control CV input. Use a -12 to +12V voltage in this socket to alter the pulse width of the oscillator square wave.

Sync In

This is an oscillator Sync reset input.

Sawtooth Out

This is the audio output for the sawtooth waveform. Its level is affected by the sawtooth level pot.

Square Out

This is the audio output for the square waveform. Its level is affected by the square level pot.

Sub 1 Out

This is a Sub-Osc output 1 octave below the main saw/square outputs.

Sub 2 Out

This is a Sub-Osc output 2 octaves below the main saw/square outputs.

Sub 3 Out

This is a Sub-Osc output 3 octaves below the main saw/square outputs.

Note; the Sub outputs run off the main square wave. If PWM is applied to the square wave, it may affect the Sub outputs!

General Specification

Inputs
- CV
- PW
- SYNC

Outputs
- SAW
- SQUARE
- SUB1
- SUB2
- SUB3

Controls
- Tune
- Glide
- Pulse Width
- Saw Level
- Square Level

Size
- 12HP

Power
- +/-12V, GND using Doepfer power socket.

VCO Calibration

It is fairly easy to tune in the VCO to another sound as you just have to alter the front panel VCO Tune control. There may be instances though were the octave spacing (scaling is out). The VCO could go out of tune as you play up the keyboard. As mentioned earlier, this does not neccessarily mean the VCO itself is at fault. The VCO responds to 1V per octave, it could be the control source is not giving out exactly 1V per Octave.

With most control sources the output scaling can be altered. This is the case with all MIDI to CV converters. It is better and far easier to alter the output scaling of the source than the scaling of the VCO.

If you do need to calibrate the VCO scaling for some reason follow the guide lines below:

Firstly, adjusting the trimmers inside the VCO may result in you voiding your warranty. Electrical safety (to yourself and to the module) must be obsereved. If you are not 100% confident and competant - do not attempt!

You will need to let the VCO circuits to ‘warm up’ for a period of time. I usually leave the systems on for 30 minutes to allow all the analogue ciruits to settle.

VCO: Turn Glide zero. Set the Tune pot to centre.

Connect a MIDI to CV converter to the VCO Pitch CV in. The MIDI keyboard you use to control the VCO will need to have audio too so you can tune the VCO to your MIDI keyboard.

On the VCO circuit board are 2 trimmer pots. Tune and Scale.

1/ Play a low key on the MIDI keyboard. Adjust Tune so that the VCO is in tune with the MIDI sound source.
2/ Play a high key (about 3 octaves up). Adjust the Scale so that the VCO is in tune with the MIDI sound source.

3/You will need to repeat steps 1 and 2 several times till tuning is spot on.

Your VCO should now be tuned to the outputs of your MIDI to CV converter.

Note, unless your MIDI to CV converter is also setup correctly, the VCO may sound out of tune when played with another CV source.

For details on how to use your MIDI to CV converter, contact the manufactuer, not us!

Oscillator Sync

The familiar and favourite oscillator sync sound is achieved by using two oscillators. One is the master oscillator and each time it begins a new wave cycle, it will reset the slave oscillator. Full details of the ins and outs of osc sync can be found various web resources.

To get this feature working on the VCO

- patch a 2nd VCO’s waveform out into the 1st VCO's Sync In
- Take your main audio out from the 1st VCO (this is where your sync sound will be)
- To control the Sync effect, control the either VCO's pitch.

Note; it is important to balance the relative pitches of the 2 oscillators to get an effective sound. With certain settings you may not get an effect or even any audio!

SY01 Synthesizer / Drum Synth - [VCF / EG / VCA]

Inputs:
2x Audio
2x Cut-Off CV
1x Resonance CV
1x Trigger/Gate
1x VCA CV

Outputs:
EG CV
Audio (VCA)

Power:
+/-12V, GND using Integrator or Doepfer power sockets.

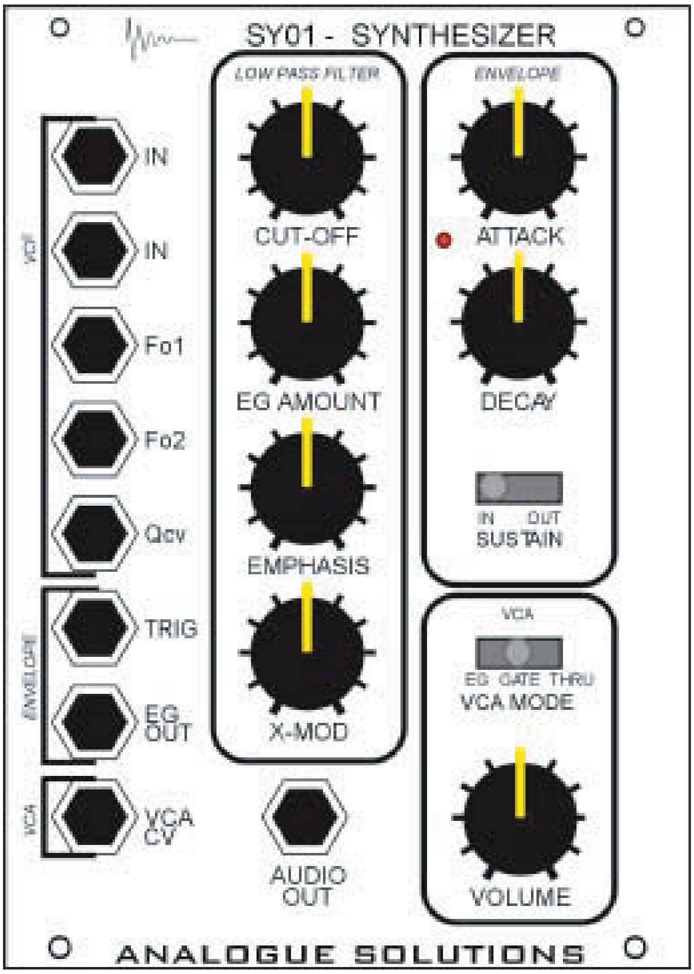
Indicators:
Trigger LED

Size:
18HP

VCF Controls:
Cut-Off
Resonance
EG Amount
Cross-Mod

EG Controls:
Attack
Decay
Sustain In/Out Switch

VCA Controls:
Mode Switch: Gate/EG/Bypass
Volume



Introduction

The SY01 is basically a whole synth in one module, minus the oscillators (we recommend using the VCO2). The SY01 comprises a Moog-type ladder voltage controlled low pass filter (VCF), envelope generator (EG) and voltage controlled amplifier (VCA). These are all internally hardware patched in the traditional monosynth configuration. This reduces an enormous amount of patching in what must be the most common patching between a VCF, EG and VCA, it also gives a huge saving compared to buying the VCF/EG/VCA separately.

Although hardware pre-patched, the module is totally patchable in the traditional modular way. Some additional less seen features are, cross-modulation control and CV control of the resonance.

The filter type used is a 24dB roll-off ladder filter, sounding very much like a Moog filter.

The SY01 can do more than just filter external audio...

By combining the VCF/EG/VCA into one module, it can act as a complete drum voice/drum synth module producing kicks, sweeps, pulse hits, toms and other electronic noises with no other modules required. With additional CV/audio control to cross modulate the filter cut-off (with the resonance set to self-oscillation and using X-MOD) the SY01 can produce many new types of sounds metallic and other useful sound effects.

Voltage Controlled Filters (VCFs) in general

The VCF receives an audio signal from the VCO and also accepts CV signals from other modules. The VCF is a filter whose cut-off frequency can be controlled by an external control voltage.

The filter processes the waveforms from the VCO and Noise Generators. VCFs on most analogue modules change the timbre of a waveform by removing harmonics - a method called subtractive synthesis. The most common form of filter is the Low Pass Filter (LPF). This works by removing the higher harmonics above the cut-off frequency, only letting the lower waves through. This filter will produce a softer and more rounded sound compared to the original un-filtered sound. The VCF will contain a cut-off frequency (Fc) control, and usually a resonance control too. The resonance control emphasises the harmonics at the cut-off point by using feedback. The resonance control is what can make a singal sound more 'electronic'. With some filters, the resonance can be set so high, that the fitler will ring, or self-oscillate at the cut-off frequency.

Controls

VCF

Cut-Off
This alters the cut-off frequency of the LPF.

Resonance (Q)
This alters the resonance of the LPF. Set to maximum it will self-oscillate producing *a* continuous sinewave. In this way the SY01 can be used as a VCO.

Cross Modulation (X-Mod)
This sets the amount of cross modulation between the audio input number 2, and the cut-off frequency. To hear the effect, the resonance control should be set to maximum. Any audio source going into inpur 2 can be used for cross modulation, but I find modulated pulse width waveforms work especially well. Cross modulation can be performed by feeding audio into the cut-off CV inputs, but this method offers no attenuation.

Envelope Generator

Attack
Sets the rate at which the EG goes from zero to maximum level.
Decay
Sets the rate at which the EG goes to zero after reaching maximum.

Sustain Switch
In: when a key is held down (i.e. the gate input is active with +5V), the envelope will remain at it's maximum level, (after the attack section of the EG cycle), i.e. it will sustain. In this mode, the Decay control becomes a Release control, setting the rate at which the EG takes to get from maximum to zero, after the key is released.
Out: There is no sustain, so the length of the envelope is wholly dependant on the attack and decay rates, and not on how long a key is pressed.

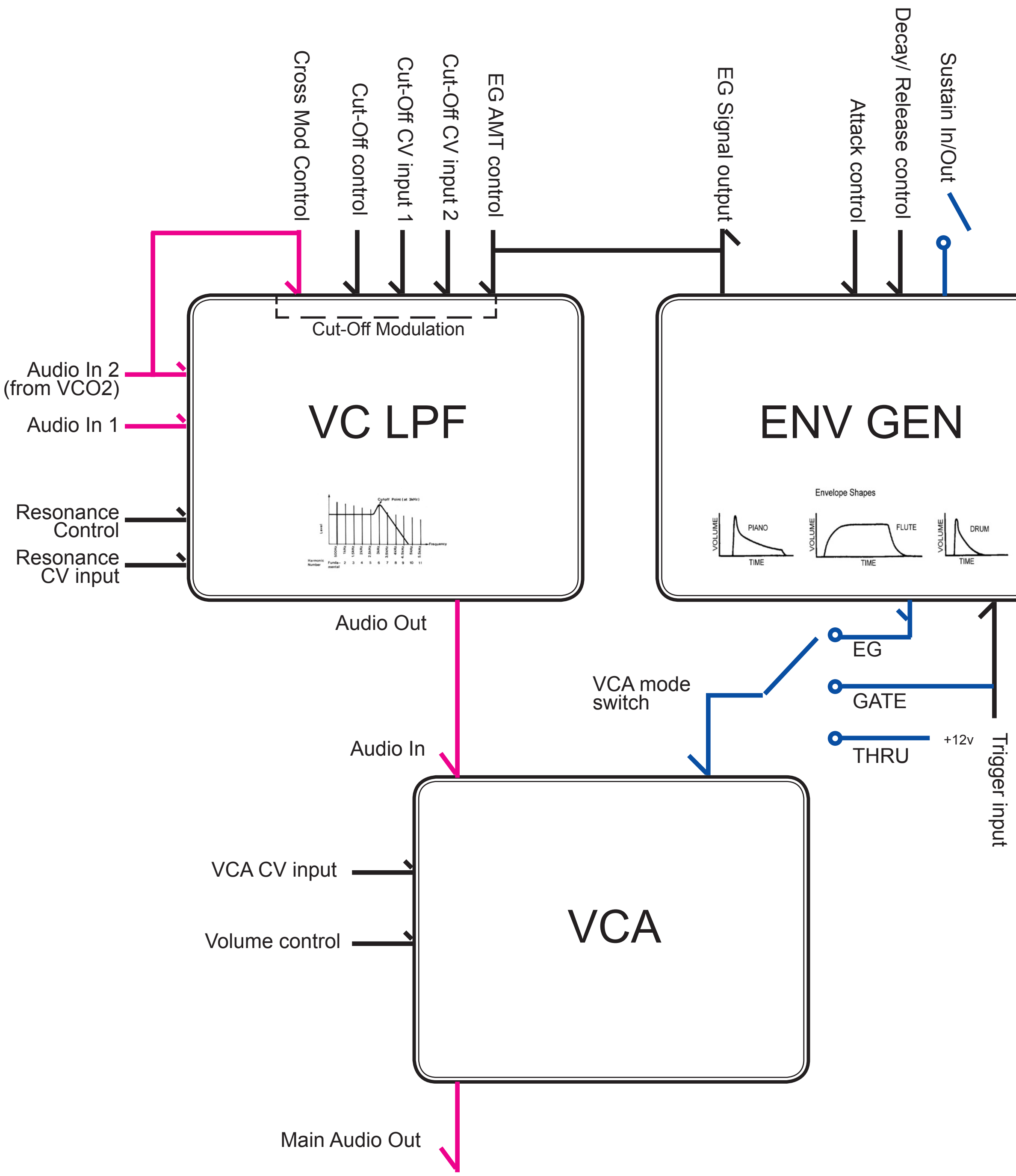
VCA

Mode Switch
Gate: The gate/trigger input signal is used to control the VCA, i.e. the VCA will either be on or off fully.
EG: The envelope contour signal will control the VCA
Bypass: The VCA will be permanently on, at maximum level. In this mode the SY01 can be used like traditional filter modules, without the need to gate the VCA.

Volume
Sets the output level from the VCA.

Sockets

Audio In 1 / 2
Audio inputs to the VCF. These are also the source for the X-MOD. Audio in 2 will provide a higher level signal to the X-MOD control than input 1.
Cut-Off CV In 1 / 2
-12 to +12V CV input for control of cut-off.
Resonance CV In
0 to +12V CV input for control of resonance level.
Gate In
Gate/trigger input to activate the EG. 0V off, +5V on.
EG Out
The EG signal is output here. EG range is 0 to +5V.
Audio Out: Audio out from the VCA (i.e. main audio output from SY01).
VCA CV In
-12 to +12 CV input for control of the VCA level.



SY01 Signal Flow

SY02 Multimode Filter / VCA- [HPF / LPF / VCA]

Inputs:
2x Audio
2x HPF Cut-Off CV
2x LPF Cut-Off CV
2x VCA CV

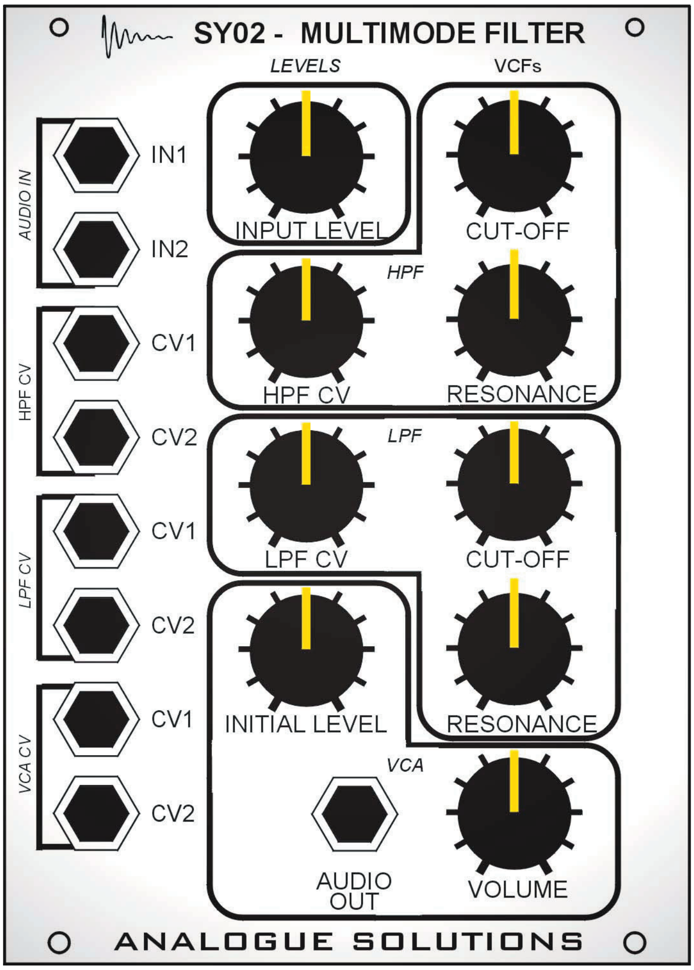
Outputs:
Audio

Power:
+/-12V, using Doepfer power socket.

Size:
18HP

VCF Controls:
HPF Cut-Off
Resonance
HPF Input CV Level
LPF Cut-Off
LPF Resonance
LPF Input CV Level

VCA Controls:
Input Level
Initial Level
Volume



Introduction

The SY02 is a multimode filter with a difference. There are separate cut-off and resonance controls for both the HPF and the LPF. This means you can use the SY02 as a LPF, a HPF, or by combining the two you get a BPF. When used as a BPF, it has the advantage over others in that you have control over the band width and the added advantage of two separate resonance controls.

The SY02 can be used on its own as a filter (by turning up the Initial Level), but it includes a VCA, this means straight away it can be used with envelopes, without the need of a separate VCA.

The filter types used in the SY02 are 12dB roll-off filters very similar to those used in the classic Korg MS20.

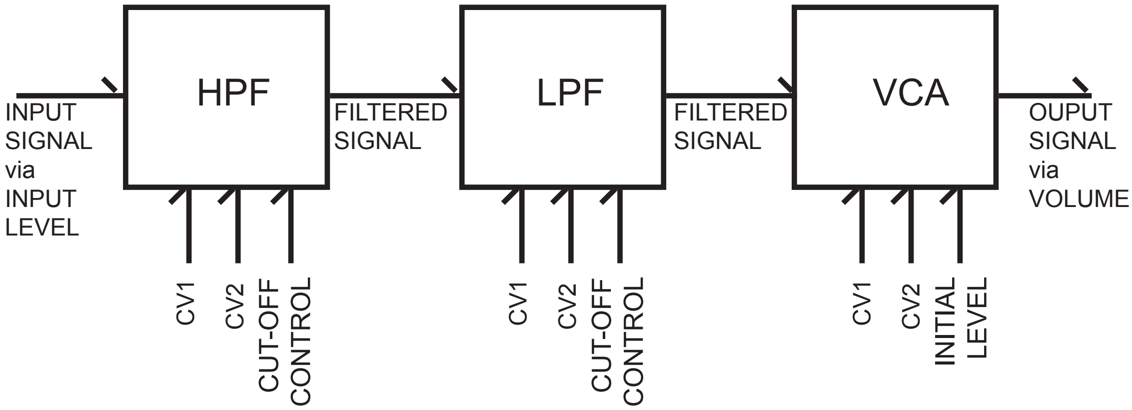
General Use

Overdrive

The controls have not been limited, so with certain level settings, especially Resonance, overdriven sounds are possible. This is great for a nice dirty distorted sound. If overdriven sounds are not required, simply decrease the Input level or Resonance level.

No Sounds

There are certain settings that will result in no audio output. Apart from the obvious (initial, input or output level at zero), this may occur when HPF cut-off is set very high and LPF cut-off set very low. This is basically due to ALL the audio being filtered out!



AUDIO AND CONTROL SIGNAL FLOW

Controls

Input Level
This sets the level of the audio signal fed into input 1. Input 2 has a fixed level.

HPF CV
This sets the level of external CV fed into HPF CV input 1 that affects the HPF cut-off. Set at minimum, the external CV has no effect, set at maximum, the external CV has full effect. HPF CV input 2 has a fixed level.

LPF CV
This sets the level of external CV fed into LPF CV input 1 that affects the LPF cut-off. Set at minimum, the external CV has no effect, set at maximum, the external CV has full effect. LPF CV input 2 has a fixed level.

Initial Level
This sets the initial level of the VCA. It applies a voltage to the VCA that opens it up. Set to maximum, the VCA will pass signals through at full volume. This would be the normal way of using this control. As the control is turned to minimum, the level of the VCA decrease finally to zero. In this case, an external CV is used to control the VCA level (such as an envelope signal).

HPF Cut-Off
This control sets the cut-off frequency of the HPF. As the controls is increased, more lower frequencies are filtered out (only high frequencies are allowed to pass through).

HPF Resonance
This alters the resonance of the HPF. Set to maximum it will self-oscillate producing *a* continuous sinewave. In this way the SY02 can be used as a VCO.

LPF Cut-Off
This control sets the cut-off frequency of the LPF. As the controls is increased, more higher frequencies are filtered out (only low frequencies are allowed to pass through).

LPF Resonance
This alters the resonance of the LPF. Set to maximum it will self-oscillate producing *a* continuous sinewave. In this way the SY02 can be used as a VCO.

Volume
This sets the final output volume of the SY02.

Sockets

IN1, IN2

These are both audio inputs to the filters. Input 1 goes via the Input level pot.

HPF CV1, CV2

These are external CV inputs to control the HPF cut-off frequency. Input 1 goes via the HPF CV pot.

LPF CV1, CV2

These are external CV inputs to control the LPF cut-off frequency. Input 1 goes via the LPF CV pot.

VCA CV1, CV2

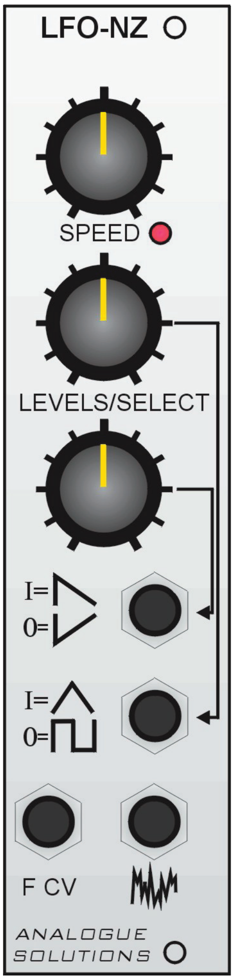
These are external CV inputs to control the VCA level.

Audio Out

This is the audio output from the SY02.

Calibration

There are 2 trimmer pots on the main board marked HPF Q and LPF Q. These are used to adjust the intensity of the resonance feed back. With these trimmers set at maximum (fully anti-clockwise), it is possible to really self-oscillate the filters (when the resonance pot on the front panel is also at a high/maximum setting. Note however that if the resonance self-oscillates at too higer level, distortion may result (although this is not always a bad thing. Set the Q trimmers to a setting that gives you the amount of resonance you want balanced with an acceptable level of distortion. This is done before SY02s are sent out.



LFO-NZ - VCLFO / NOISE

Inputs:	Outputs:	Power:
Freq CV	Ramp/Rev.Ramp	+/-12V
	Tri/Square	
	Noise	

LFO Controls:	Size:
Ramp/Rev.Ramp Level/Select	6HP
Tri/Square Level/Select	

Indicators:
LFO Speed

Introduction

An LFO module with 2 separate waveform outputs, out of a choice of 4. Use an LFO to control a VCO (to create vibrato), a VCA (for tremelo) or to control a filter's cut-off frequencer (wah-wah /filter sweep). Use it any where you want to change a module's performance over time.

2 modules in one:
Separate LFO and Noise features

Controls

Speed
This sets the LFO frequency

Ramp / Reverse Ramp Level / Select Switch
Sets the output level to the Ramp out jack of the Ramp signal. Pull the control out to select Reverse Ramp.

Triangle / Square Level / Select Switch
Sets the output level to the Triangle out jack of the Triangle signal. Pull the control out to select Square.

Sockets

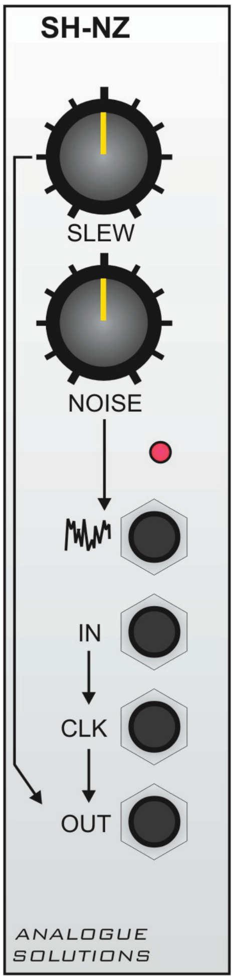
Ramp / Reverse Ramp
Outputs the Ramp or Rev.Ramp signal depending which is selected.

Triangle / Square
Outputs the Triangle or Square signal depending which is selected.

Noise
Direct output of a white noise signal. Use this as a control source for interesting effects, or as an audio source for wind or percussive sounds when used with a filter/EG/VCA (such as our SY01)

Frequency CV In
This is a control voltage input. The level of the input voltage controls the speed of the LFO. This is good for creating even more interesting LFO effects. Feed the output of an envelope, like our EG02, into here for an LFO speed that changes over time. Or use a CV sequencer such as the SQ8 to change the speed on each beat.

SH-NZ - Sample+Hold / Noise



Inputs:	Outputs:	Power:
SH Signal	SH Signal	+/-12V
SH Clock	Noise	

Controls:	Size:
SH Slew rate	6HP
Noise Level	

Indicators:
SH Clock

Introduction

2 modules in one:

Sample and Hold and Noise features
The S+H and Noise units are independent so can be used entirely of each other.

Controls

S+H Slew
This adds slew (or portamento) between each changing voltage held by the S+H. So as each voltage is sampled, the changes in old to new voltage level will be smoothed out.
Noise Level
Alters the noise level.

Sockets

S+H In
This is the signal source for the S+H. Normally the noise output would be used for a random output.
S+H Clock In
Each clock or gate pulse will make the S+H sample the input signal then hold it till the next clock signal. Normally a square wave signal would be fed into here, like the LFO-NZ square wave output.
S+H Sig Out
This is the output from the S+H unit.
Noise Out
This is an audio output from the white noise generator. This can be fed into the S+H signal input for random S+H.

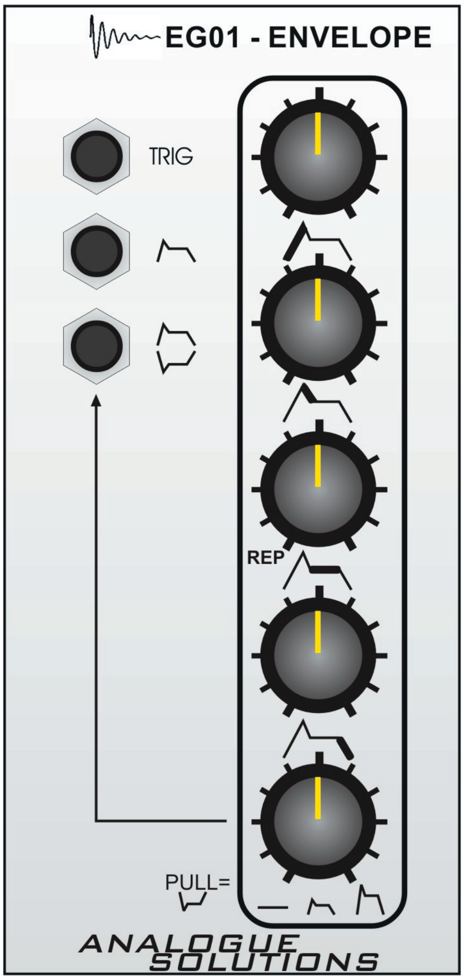
Obtaining a stepped random signal (using the Sample and Hold Facility)

To create a stepped random voltage (as indicated on the first position of the LFO wavform switch):
Patch a Square wave LFO output into the S+H Clock input, the Noise output to the S+H Signal input. Turn the Noise level up to about 30-50% (A higher setting causes strange - though not neccessarily - output). The S+H output signal will appear at the S+H Out socket.

EG01 - Envelope

Inputs:	Outputs:	Power:
Trig In	+out	+/-12V, GND
	+/- out	

Controls:	Indicators:	Size:
Attack	Trig LED	12HP
Decay		
Sustain		
Release		
Level with integrated push/pull switch for invert		



Envelope Generators (EGs) in general

The EG produces a CV that varies over a period of time. It's start is triggered by a Gate or Trigger signal.

The main use for an EG is to vary the volume of a sound, (when used to control a VCA), to reproduce the way natural instruments sound. E.g. a piano sound starting loud when the key is struck, then gradually dieing away. It can also be used to change the timbre of a sound over time, by controlling the cut-off frequency of a VCF.

The most common EGs are ADSR, AD and AR (Attack, Decay, Sustain, Release). The attack time adjusts the rate at which the envelope will rise to its peak value. This is initiated with a Gate signal. The Decay adjusts the rate the signal takes to fall to the Sustain level. The Sustain level adjusts the level that the EG signal will sustain from the end of the Decay time till the end of the Gate signal (i.e. when the key is release). The Release time adjusts the rate the signal takes to fall from its current level to zero, after the end of the Gate signal (when the key was released). If a short Trigger signal is used to initiate the EG, there will be no Decay or Sustain portion to the EG signal. The signal will rise to peak level (Attack), then imidiately fall from peak to zero at the Release rate.

Use

Sockets

Trigger In	A positive gate or trigger voltage in here will activate the ADSR sequencer.
+ EG Out	A full level positive envelope signal from this socket. Level pot has no effect.
+/- EG Out	The EG signal is positive. By pulling out the EG knob, the signal is inverted (negative).

Controls

Attack	After receiving a Gate signal, this sets the time is takes the envelope to reach full level.
Decay	After reaching full level, this sets the time it takes the envelope to decrease to the Sustain level.
Sustain	The sets the Sustain level. The envelope signal will remain at the Sustain level (after decreasing from the maximum level) as long as there is a Gate signal present at the Trigger in socket (i.e. as long as a key is held down).
Release	After the key is released (the Gate signal no longer present), this sets the time it takes for the envelope signal to decrease to zero.

Special EMS-Style Repeat Function

The EG01 has an EMS-style repeat function which is easy to use and adds powerful modulation capabilities.

When the sustain level is set to zero the repeat function will be active. The envelope will repeat (re-trigger itself) so as soon as the envelope finishes its cycle, it starts all over gain. In this mode the EG becomes an LFO or Oscillator. The speed is set by the Attack and Decay time. (The Release control will have no effect). By setting very short times an audio signal will be present!

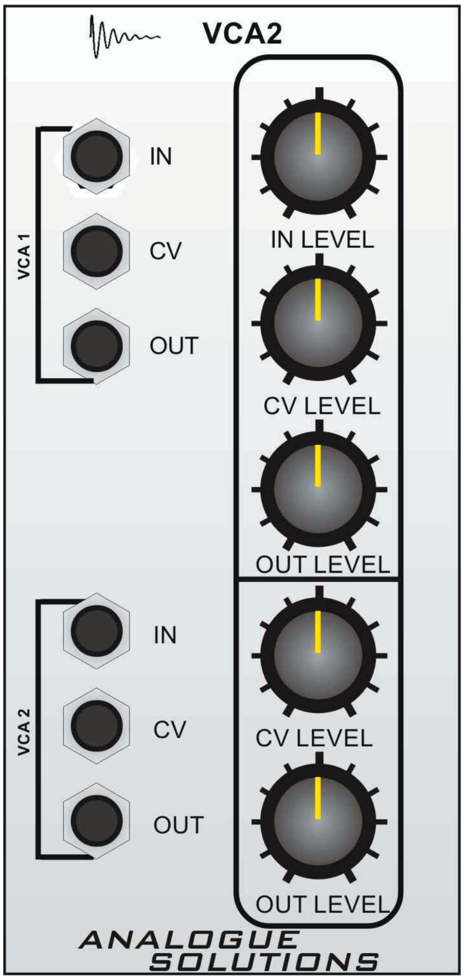
Whether the repeat is short giving high frequencies (audio signals) or slow (like an LFO), by varying the Attack or Decay times you are changing the wave shape of the output. Anything from Saw to Triangle to Reverse Saw wave froms can be created.

The unique way in which the EG01 repeat works is that it will only repeat as long as the Gate voltage is at the Trigger input (i.e. the key is held down/sustained). This is a great bonus compared to other repeating envelopes in that rather than repeating indefinately, it will only repeat as long as you want it to (by controlling the length of the Gate signal). One application of this is by setting the repeat function to create an audio signal. The output will only produce audio when a Gate signal is present. This gives you a gated audio signal - effectively an Oscillator through a VCA.

VCA2 - Dual (2x) Voltage Controlled Amplfier

Inputs:	Outputs:	Power:
Signal In 1	Signal Out 1	+/-12V, GND
Signal In 2	Signal Out 2	
CV In 1		
CV In 2		

Controls:	Indicators:	Size:
Signal Level 1	-	12HP
CV Level 1		
CV Level 2		
Signal Out Level 1		
Signal Out Level 2		



Introduction

2 modules in one:

Two independent VCAs in one module. VCAs are amplifiers whose gains are under voltage control. This means you can use a modulation CV, such as an LFO or EG signal, to modulate the level of another signal.

A typical use would be the output from an envelope generator altering the volume level of a signal, such as a fiilter output.

Controls In Detail

The 2 VCAs are identical, except that VCA 2 does not have a Input Signal level control.

IN LEVEL
Alters the input signal level

CV LEVEL
Alters the CV level, to adjust the amount of VCA modulation.

OUT LEVEL
Alters the output signal level.

Sockets In Detail

IN
VCA Signal input. Normally the signal would be audio, but CV signals can also be modulated.

CV
CV signal input for VCA modulation.

OUT
Output from the VCA

General Specification

<u>Inputs</u>
IN 1
IN 2
CV 1
CV 2

<u>Outputs</u>
OUT 1
OUT 2

<u>Controls</u>
IN LEVEL 1
CV LEVEL 1
CV LEVEL 2
OUT LEVEL 1
OUT LEVEL 2

<u>Size</u>
12HP

<u>Power</u>
+/-12V, GND using Doepfer power socket.

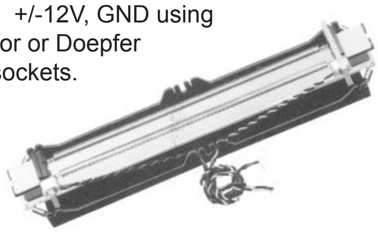
Effects Modules

SR01 Spring Reverb with VCA

Inputs:
Audio In
Wet Mix CV In 0-12V

Outputs:
3.5mm Audio out
6.35mm Audio out

Power:
+/-12V, GND using Integrator or Doepfer power sockets.

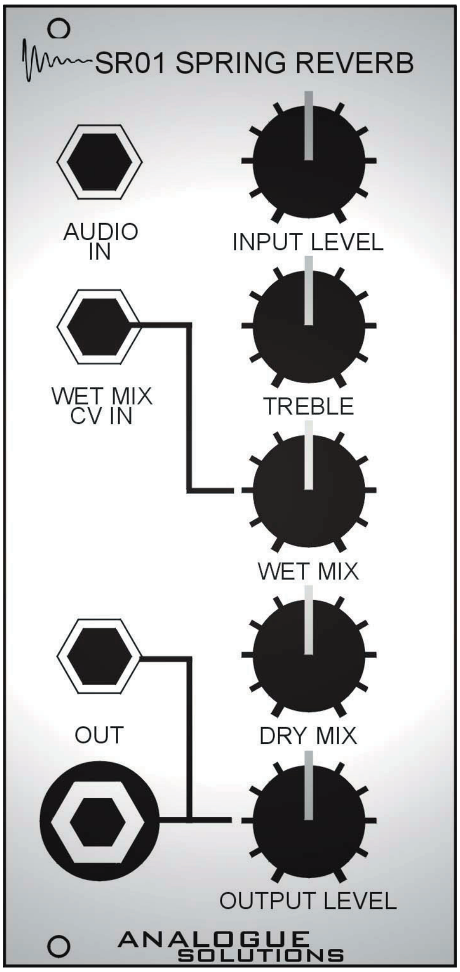


Controls:
Input Level
Treble
Wet Mix Level
Dry Mix Level
Output Level

Indicators:
None

Size:
12HP

Dual Spring Specification:
Dual spring, length: 160mm x2. Feedback/shock reducing mountings.
Rever time approx. 2 to 4 seconds (frequency dependent).
(Triple spring spec similar to above, but also has shielded case with shock mounting and connection wires).



Introduction

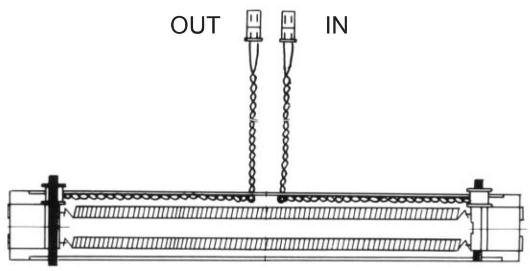
Natural reverbaration is caused by reflected sounds, and in a normal room most of these reflections come from the walls and Ceiling. It is important to differentiate between reverb and echo; an echo is heard as clear repetition of the original sound, whereas reverb consists of numerous reflected sounds which normally come only marginally behind the original sound.
Reverbaration tends to enhance sounds by giving depth and colour, and artificial reverbaration is understandably a very popular effect. There are several non-digital ways of producing artificial reverb, but one of the best methods that produces excellent results at a low cost is the spring-line system as used in the SR01. Spring reverb is the same type of reverb found built into the EMS VCS3 and ARP2600.
A spring-line normally consists of 2 long springs mounted side by side, and having an input transducer at one end and an output transducer at the opposite end. A signal applied at the input transducer results in sound waves being transmitted down the springs, and these are desinged so that there is a significant delay time. Also, in order to produce a more realistic effect it is normal for the springs to give different delay times. The sound waves produce an output signal when they reach the output transducer, but some of the energy is reflected back to the input transducer, and then back to the output transducer where further output signal is produced. In fact, the energy will be reflected up and down the springs numerous times, gradually diminishing until no significant output is produced.
This is obviously analogous to sounds being reflected around a room, and for such a simple system, can give a very realistic effect.

Mounting

☠ Disconnect the power supply from the mains!

The spring is small enough to fit inside most 3U and 6U cases. To keep mains hum to a minimum, mount spring as far away as possible from the mains transformer. For AS1600/ 1900 case we recommend vertically on the right hand side panel. For any other case, try the top or bottom panel. Make sure there is enough clearance for the module PCBs, and that it is clear from the power supply. Ensure no cables interfere with the springs.

Note: with the triple spring line that also has the shielded case, space may be more of a problem. It should fit inside most 3U cases, and almost definately in any 6U case (it partly depends on how deep the modules you have are). As it is boxed, external mounting should be OK. The triple spring case has 4 mounting holes.



Connections

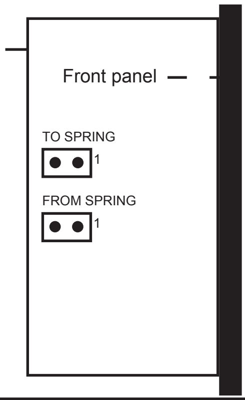
The Green/Black cable is the spring input; connect these wires to the connector marked 'TO SPRING' on the SR01. The Red/Black cable is the spring output; connect these wires to the connector marked 'FROM SPRING'.
Pin 1 is the coloured wire, pin 2 is the black wire (ground). Be sure to fit the connector the right way round! Pin 2 is towards the edge of the board. See diagram to the right for visual details.

PCB viewed from component side

To spring:
Pin 1 is green wire

From spring:
Pin1 is red wire.

For both connectors:
Pin 2 is black



If neccessary, use cable ties to gather cables away from spring. Make sure spring cables will reach the module connectors.

The spring assembly fits to the base with the self-adhesive strip supplied. Although this will give rock solid support, screw holes are provided should you wish to screw mount it.

General Use

Everything self explanatory really. Feed audio into the spring reverb, take it out into your mixer. All controls self-explanatory. The Wet Mix CV input has a 0-12V range to vary the wet mix level. When using the wet mix CV input, the wet mix control becomes a level offset.

The treble cuts the top end before the signal enters the spring.

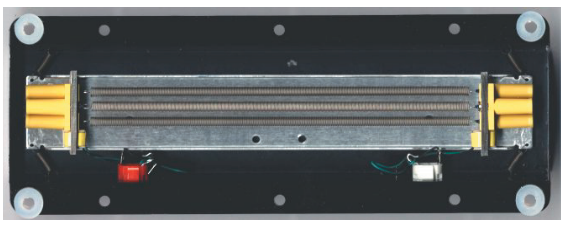
Feed you VCO straight into the Reverb, with dry mix to minimum, and wet mix up high. If the pitch is kept constant the VCO takes on a new timbre - strange and metallic. With subtle triangle LFO modulation on the VCO pitch, you start to get a chorusing effect.

Echo Effect

Set-up the reverb as normal. Feed a gate/clock or square LFO waveform signal into the wet mix CV input, and turn the wet mix to a low value. This way the reverb affect sound will be gated as it decays away, creating an echo style effect. (Thanks to Tim Hannigan for this patch).

Weird Effects

Bashing the casing or scraping the springs (not a good idea when spring is mounted inside new to the power supply) is very avant garde.



A triple spring line with a shielded case and phono connectors is available for improved noise reduction and a smoother sound.

Sequencer & Control Modules

MT16 MIDI to Trigger / Gate converter

Inputs:
MIDI In

Outputs:
16 independent trigger outs
MIDI Thru

Power supply:
+5V, GND using Integrator
or Doepfer power sockets.

Controls:
Learn button

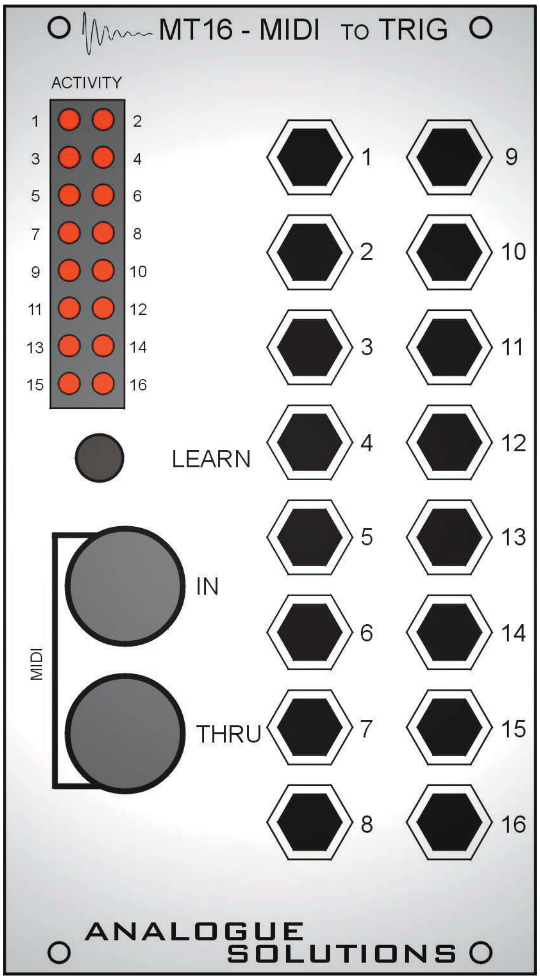
Indicators:
16x Actuivry LEDs

Size:
14HP

Introduction

The MT16 is an advanced MIDI to triggering module. It allows triggering of a range of analogue devices; drum voice modules, envelope modules, analogue sequencers, analogue drum machines, and anything else with a gate, trigger or clock input.

It has 16 separate outputs, so 16 devices can be independently triggered via separate MIDI keys.



Use

Connecting

Connect the MIDI sequencer or other MIDI device to the MIDI input of the MT16. Connect MT16 trigger outputs to device you wish to trigger.

Programming

Programming the MT16 is a very simple proceedure.

Press the LEARN button. All LEDs will light. Then press a MIDI key.

The MT16 will set its MIDI receive channel (which will be the same for all outputs) to the MIDI channel the keyboard is transmitting on.

The outputs are pre-assigned, output 1 starting at middle D.

Uses

Triggering Drum Voices

The most logical use for the MT16 is for easy programming of Concussor drum voice modules via a MIDI sequncer. Electronic percussion patterns can be quickly created using for example Cubase's grid editor or a rhythm sequencer such as the Akai MPC2000.

Gating Envelopes

The length of the trigger pulses is determined by how long the MIDI key is held down for (sustained). When you hold a MIDI key down, the trigger output will stay on (+5V) until the key is released. This makes the MT16 ideal for controlling envelopes.

Devices with Clock Inputs - MT16 is more advantagous than using a MIDI-Sync device!

The MT16 offers a much more flexible way of synchronising analogue sequencers and drum machines than using a MIDI-Sync device. MIDI-Sync devices have the problem that they run continually when your MIDI sequencer is running. Therefore you cannot stop and re-start your analogue sequencers mid-song. Using the MT16 this problem can be overcome. By programming a bar of 16th notes and looping the bar, straight away you have created a clock signal that your analogue sequencers can lock on to. But there are several important advantages - you can have up to 16 clock signals, they can all be different divide ratios by changing the note pattern (so you could say have one sequencer running half the speed of the other). The string of notes does not have to be a steady pattern, it can be a syncopated rhythm, clocking you analogue sequencers in a particular changing pattern you decide.

M2CV - MIDI to CV Converter

Introduction

The M2CV is a quick and easy to use module that allows control of analogue devices from a MIDI controller. The M2CV is straight forward MIDI to CV conversion - easy to use.

Use

Connecting

Connect the MIDI sequencer or other MIDI device to the MIDI input of the M2CV.

Normally CV1 (PITCH) goes to the pitch CV input of your synth. CV2 goes to any other CV input, like filter cut-off CV. Gate goes to the Gate or Trigger input o your synth.

Make sure the M2CV is on the right MIDI channel and away you go!

Details

The MIDI activity light will be lit whenever there is a gate voltage present (i.e. whenever 1 or more MIDI keys are held).

The Pitch CV out (CV1) conforms to 1V per Octave.

The controller CV out (CV2) responds to MIDI controller 1 (mod Wheel). It has a 0 to 10V range. Controller number cannot be changed. Resolution is 7 bit.

Legato

When 1 or more notes are overlapped, this output will change from 0 to 5V. It will return to 0V when no notes are overlapping. This output is ideal for activating devices like portamento, LFO reset, Clock inputs, VCF cut-off etc.

Accent

When a MIDI velocity of over 99 is received, the Accent output will go from 0 to 5V. It will not return back to 0V till it receives a MIDI note velocity value of less the 100. This output is ideal for accenting notes by connecting it to a VCA or filter cut-off input. It could also clock a sequencer or other device.

Note, there is no Pitch bend.

MIDI Channel Selection

Use the DIP switches to select MIDI Channel

CV2 Control Source Assignment:

Use the toggle switch to select CV2 source, left is Velocity, right is Mod Wheel

The Push button is a manual Gate trigger button.

SQ8 8 Step CV and Gate Sequencer

Inputs:
Reset In
Clock In

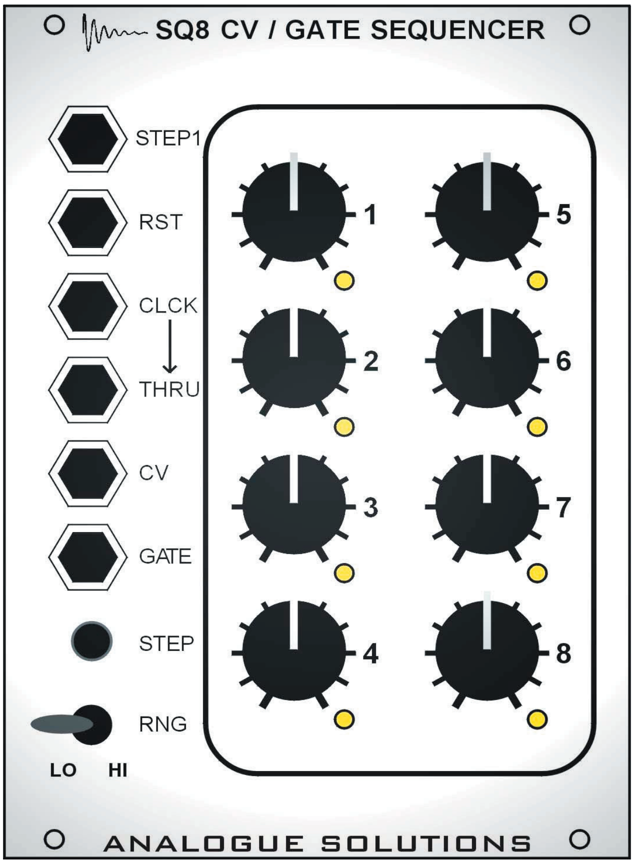
Outputs:
CV
Gate
Step 1
Clock Through
Additional 8 step outputs with optional expansion

Power Supply:
+/-12V, +5V, GND using
Integrator or Doepfer
power sockets.

Controls:
8 x CV knobs with
built in Gate Switch
Step Button
Range Switch (5/10V)

Indicators:
8x Step LED's

Size:
18HP



<u>Introduction</u>

Analogue sequencers allow you to programme a set of voltages and Trigger events (8 in this case), and allow you to step through them by using a clock signal or LFO square wave. With each step, the next voltage as set by the control is present at the CV output. This way repeating melodic lines can be produced. The sequencer need not control pitch, it could be used to change the filter cut-off, volume, or LFO speed.

The advantage analogue sequencers have over hardware sequencers is firstly their immediacy. Having all the controls, switches and sockets in front of you allows quicker programming. Secondly, as the output is an analogue control voltage and not a digital MIDI signal, the output of analogue sequencers can easily be mixed with other voltages, or processed in some way.

The SQ8 is a single channel sequencer with 8 steps. As it is single channel, larger sequencing systems can be built up by doubling up on sequencers. As they are all separate, you can clock them at different speeds or clock divide ratios.

Controls In Detail – standard operation

Clock In

The SQ8 requires a +5V clock signal. Each clock signal will advance the sequencer 1 step. When the last step is reached, it resets back to the first step. Take the clock signal from a square wave LFO, Clock module or MIDI to Clock converter.

Clock Thru

The clock signal appearing at the CLOCK IN socket is buffered and transferred to the CLOCK THRU socket, so it can be daisy chained to another sequencer.

Reset In

A 5V signal in here will reset the sequencer to step 1.

Step 1 Out

Each time step one is played, the step 1 gate socket will go high (output +5V). This socket can be used to trigger other devices, such as a cymbal sound at the start of each bar, or to trigger another sequencer. The step 1 out will give a clock signal that is the main clock in divided by out.

CV Out

This socket will output a control voltage set by the pot for the current step. It can be changed in real time.

Gate Out

This socket will output a 5V gate signal at each step where the CV control is not at the zero position (i.e. Gate 'on').

Step Button

This allows the sequencer to be manually stepped by one step at a time whilst the clock is not running. It can also be used to reset the sequence. As with most analogue sequencers, step it to the last step (8 or whatever is patched) so that when you start the clock, the sequencer starts on position 1.

Note: The step button will only work when any clock input signal is at a low state (0V). So it will note work when the clock input is high. This situation will note arise in normal use, as one would only normally use the step button when the master clock is stopped (making the clock signal low). If the sequencer is being clocked by a clock divider, even when the clock is stopped, it is possible that the clock signal may still be high until reseted.

CV / Gate pots / Range switch

The CV and Gate functions are built into a single combined pot/switch. Turning the pot fully left will turn the Gate off. You will fell the switch click off.

There are 8 CV pots, 1 for each step. When the range switch is HI the range is approximately 0 to 10V, when LO the range is approximately 0-5V. LO is best for use when controlling a VCO as this gives better abillity to fine tune each step.

The CV pots have a built in Gate switch. Pull the knob out to turn off the gate for that step, push it in to turn the gate on.

LED Indicators

There are 8 LED's. These will light in turn to show step position.

Alternative Applications Of Controls:

Clock In

The sequencer does not have to be stepped through at normal regular intervals as is usual. It can be clocked from any source, such as the gate from a MIDI-CV converter or monosynth, the trigger out of a drum machine such as the TR606/808, or from the gate outputs of other analogue sequencers. This allows it to step through the sequencer rhythmically, as and when you want.

If a sine, triangle or sawtooth wave (that goes positive and negative in polarity) is used, the sequencer will step randomly (backwards and forwards), and also skip beats! This can produce interesting musical results or is good for sound effects.

Reset In

It can be taken from an source, such as Gates from MIDI to CV converters, synth's or other analogue sequencers.

Step Out

It can be used individually to clock other analogue sequencers, to trigger analogue percussion modules, gate monosynths.

The step out is basically a divide by 8 clock divide (relative to the clock input signal). Any step output (normally step one) can be used to clock something else 8x slower.

CV Out

If a very high frequency clock pulse is used to step the sequencer (in audio frequencies), the CV pots can be used as a waveform generator. Because of the quantised steps, it will sound digital in form, unless an external slew rate generator (portamento) module is used to smooth the waveform.

More SQ8 Example Patching

Standard connection to a monosynth to control pitch and gate;

CV out to Synth CV in.
Clock Thru to Synth Gate in.

Filter control of monosynth;

CV out to Filter cut-off input.
Connect CV and Gate of synth to another sequencer or MIDI to CV converter.

Alternative connections to a monosynth;

Plug CV out to Synth CV in
Step 1 out to Gate in
The synth will be triggered on the 1st step. If the release time is long, you can still hear the pitch changing with each step, but without the EG re-triggering.

Miscellaneous Connections;

Step 1out is a divide by 8 clock divider.

Take step 1into the clock of another sequencer.

Use a quantiser if you want to precisely control to pitch of a synth.

Use a voltage scaler to scale down the control voltage to a smaller range when necessary.

When using a clock other than our MC01 master clock module, (as with all analogue sequencers), you must manually step to the last step of the sequence before you start the clock. This is done automatically when using the MC01.

To control transpose:
Use 2 SQ8s (we'll call them A and B) and a mixer.
Clock the SQ8 A 1 step every bar. Clock SQ8 B as normal. Feed the output CVs of both through a mixer (like the MX61), then take the output of the mixer to a synth as normal. SQ8 A will control the transpose level, changing every bar. SQ8 B sets the pattern of notes as normal. This is a handy and easy way of shifting the pitch of all 8 steps up or down with only one pot.

Using the SQ8 as a waveform generator

By clocking the SQ8 at audio frequencies, the output of the sequencer can be used to generate audio waveforms.

The simplest example is to create a square wave. Simply set to 1st 4 pots to full and the following 4 to zero. To double the frequency, set the 1st two pots to full, the next two to zero, and repeat.

Now take the 1st example with 4 pots at full and 4 at zero. Now if you start to change the the levels away from their setings, you will hear the timbre change. All combinations of settings are useable, so just play around live as the sequencer plays.

GT8 8 Step Dual Gate Sequencer

Inputs:
Reset In
Clock In

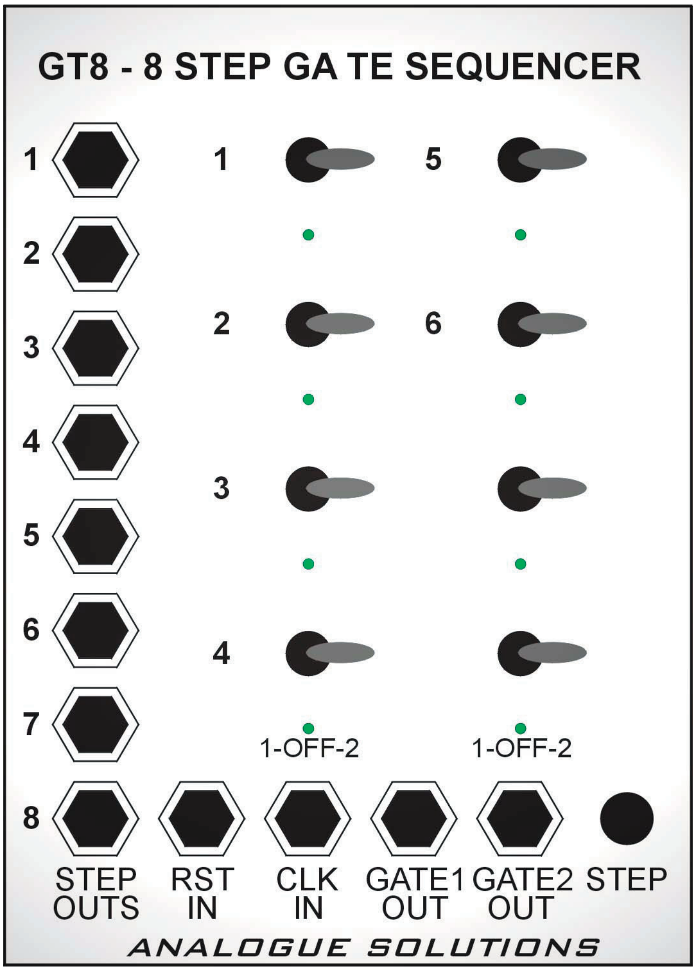
Outputs:
2 x Gate. Range 0-5V
8 x Step Outs (on=5V).

Power Supply:
+/-12V, +5V, GND using Integrator or Doepfer power sockets.

Controls:
Step Button
8 x Gate switches
(left=channel 1, middle=off, right=channel 2)

Indicators:
8x Step LED's

Size:
18HP



Introduction

Analogue sequencers allow you to programme a set of voltages and Trigger events (8 in this case), and allow you to step through them by using a clock signal or LFO square wave. With each step, then next voltage as set by the contol is present at the CV output. This way repeating melodic lines can be produces. The sequencer need not control pitch, it could be used to change the filter cut-off, volume, or LFO speed.

The advantage analogue sequencers have over hardware sequencers is firstly their immediacy. Having all the controls, switches and sockets in front of you allows quicker programming. Secondly, as the output is an analogue control voltage and not a digital MIDI signal, the output of analogue sequencers can easly be mixed with other voltages, or processed in some way.

The SQ8 is a single channel sequencer with 8 steps. As it is single channel, large systems can be built up by doubling up on sequencers. As they are all separate, you can clock them at different speeds or clock divide ratios.

Main Applications

For triggering envelopes, clocking other devices, or controlling CV's. The GT8 comes with a 5V range as standard. This range is ideal for triggering most monosynths and envelopes.

If requested when ordering, the CV8 can be given a 10V range. It is possible to change to range between 5 and 10V by adding or removing resistor R13 (100K) on the main board. Having this resistor in place makes the range 10V. Changing this yourself is safe to do but will invalidate the warranty.

Controls In Detail – standard operation.

Clock In

The GT8 requires a +5V clock signal. Each clock signal will advance the sequencer 1 step. When the last step is reached, it resets back to the first step. Take the clock signal from an LFO, Clock module or MIDI to Clock converter.

Step Outs

There are 8 step out sockets. As each step 1 to 8 is reached, the corresponding step out socket will go high (output +5V). By patching one of these outputs into the Reset In socket, the sequence length can be altered between 1 to 7 steps long. For example, to obtain a five step sequencer, patch the next number up step output into the reset input, i.e. step out 6. To have an eight step sequencer, no reset patch is needed.

Gate 1/2 Outs

Either of these two sockets will output a +5V gate when at a particular step, the corresponding switch is in either position Gate 1 or Gate 2. Centre position will result in not Gate from either socket.

The pulse width of the clock input decides the pulse width of the Gate outputs.

Step Button

This allows the sequencer to be manually stepped by step at a time whilst the clock is not running. It can also be used to reset the sequence. As with most analogue sequencers, step it to the last step (8 or whatever is patched) so that when you start the clock, the sequencer starts on position 1.

Note: The step button will only work when any clock input signal is at a low state (0V). So it will note work when the clock input is high. This situation will note arise in normal use, as one would only normally use the step button when the master clock is stopped (making the clock signal low). If the sequencer is being clocked by a clock divider, even when the clock is stopped, it is possible that the clock signal may still be high until reseted.

Gate Switches

There are 8 Gate Switches, 1 for each step. They have three positions. When switched left, Gate 1 is active, right, Gate 2 is active, and centre is no Gate. The voltage range is 0 to 5V.

LED Indicators

There are 8 LED's. These will light in turn to show step position.

Alternative Applications Of Controls:

Clock In

The sequencer does not have to be stepped through at normal regular intervals as is usual. It can be clocked from any source, such as the gate from a MIDI-CV converter or monosynth, the trigger out of a drum machine such as the TR606/808, or from the gate outputs of other analogue sequencers. This allows it to step through the sequencer rhythmically, as and when you want.

If a sine, triangle or sawtooth wave (that goes positive and negative in polarity) is used, the sequencer will step randomly (backwards and forwards), and also skip beats! This can produce interesting musical results or is good for sound effects.

Reset In

The reset signal does not need to be taken from the step outputs. It can be taken from an source, such as Gates from MIDI to CV converters, synth's or other analogue sequencers.

Step Outs

These can be used individually to clock other analogue sequencers, to trigger analogue percussion modules, gate monosynths, or control analogue switching modules such as our Fill In Module.

Each step out is basically a divide by 8 clock divide (relative to the clock input signal). Any step output (normally one) can be used to clock something else 8x slower.

More GT8 Example Patching

Any step out is a divide by 8 clock divider.

Take step 1 (or any other step output) into the clock of another sequencer.

GT8 controlling open and closed hihats (or 2 monosynths);
Gate out 1 to CH, Gate out 2 to OH.

Use a pulse shaper on the clock input of the GT8 to change the Gate output pulse widths.

Take different step outputs (from same or other GT8 sequencers) and mix them with a multiple to get another gate sequencer.

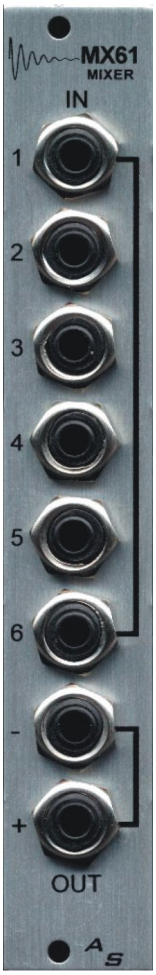
When using a clock other than our MC01 master clock module, (as with all analogue sequencers), you must manually step to the last step of the sequence before you start the clock. This is done automatically when using the MC01.

Using a SQ8, GT8 & MC01;
Run one at 1:1 clock ratio, and the other at a divided down clock ratio, say 1:4 for interesting effects.

Utility Modules

MX61 6 Input Audio / CV / Gate Mixer with Inverter

Inputs:	**Ouputs:**		**Power:**
6x ins	3.5mm out	(normal)	+/-12V, GND
	3.5mm out	(inverted)	
Controls:	**Indicators:**		**Size:**
none	none		4HP



Details

The MX61 is a 6 input unity gain mixer. It can mix audio, control voltages and gates. It allows you to bring many signal source together to be sent to another module, such as several VCOs into one filter, or to be used to mix audio sources and bring them together to an external recodring devce or mixing desk. It has 2 outputs, a normal (positive) output, and an inverted output.

Use

Audio Mixer

Use this to mix audio in instances where you have many sources you wish to mix to one mono channel, e.g. several percussion modules. When mixing audio, you would normally use the normal output.

CV Mixer

Several CV sources can be mixed to make a more complex control signal. E.g. mix an LFO with an envelope signal.

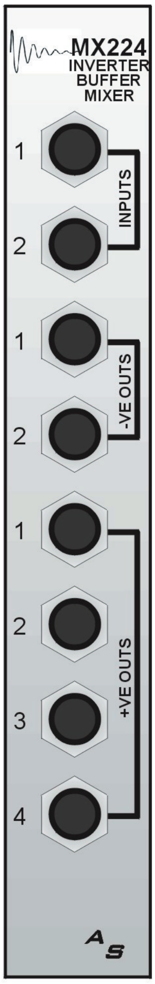
Inverter

When the inverted output is used, the signal is inverted so it becomes a negative signal (or a negative signal becomes positive). For example, put 5V through, and you get -5V out. An example of a useful application of this is to create auto-pan. Feed an LFO into the mixer. Take the normal output to 1 VCA, and the inverted output to another VCA. Pan the 2 VCAs hard left and right. Feed the same signal into both VCAs and it will pan between the speakers!

Gate Mixer

By taking different step outputs from a CV8 or GT8 and mixing them with the MX61, new step patterns can be created.

The MX61 has a -10 to +10V range.



MX224 Mixer / Inverter / Buffer

Inputs: 2x ins	**Ouputs:** 2 inverted 4 normal	**Power:** +/-12V, GND
Controls: none	**Indicators:** none	**Size:** 4HP

Details

The MX224 is a 2 input mixer/inverter/buffer. It can mix audio, control voltages and gates with multiple normal and inverted outputs. I deal for splitting signals with no signal drop.

Use

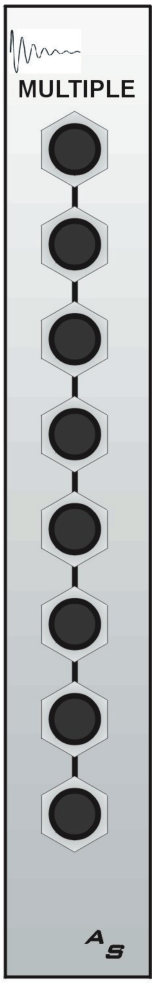
Audio Mixer
Use this as a simple 2 input audio mixer.

CV Mixer
Two CV sources can be mixed to make a more complex control signal. E.g. mix an LFO with an envelope signal.

Buffer
When a signal is continually split, like with a multiple, the signal level can drop. A buffer is a device that will maintain the voltage so no drop is seen at the output.

Inverter
When the inverted output is used, the signal is inverted (polarity is inverted) so it becomes a negative signal (or a negative signal becomes positive). For example, put 5V through, and you get -5V out. An example of a useful application of this is to create auto-pan. Feed an LFO into the mixer. Take the normal output to 1 VCA, and the inverted output to another VCA. Pan the 2 VCAs hard left and right. Feed the same signal into both VCAs and it will pan between the speakers!

The MX224 has a -10 to +10V range.



Multiple

Inputs/Outputs: Size:
2x 4 parallel connected sockets or 4HP
1x 8 parallel connected sockets
(state when ordering - 2x4 is default)

Introduction

This is a device that is a row of jack sockets all wired together in parallel.

It can be provided as two independent rows of 4 if required. If you have an 8 way multiple, you only have to remove the tinned wire link on the PCB to make this 2x 4 way multiples.

Use

The main use of a multiples is to split signals, making it available on more than 1 socket. For example, to make the output of an envelope which only has one output socket, available to lots of inputs. Plug the EG output into the multiple, then the seven multiple sockets will also carry that EG signal.

The multiple is a passive module. It takes no power.

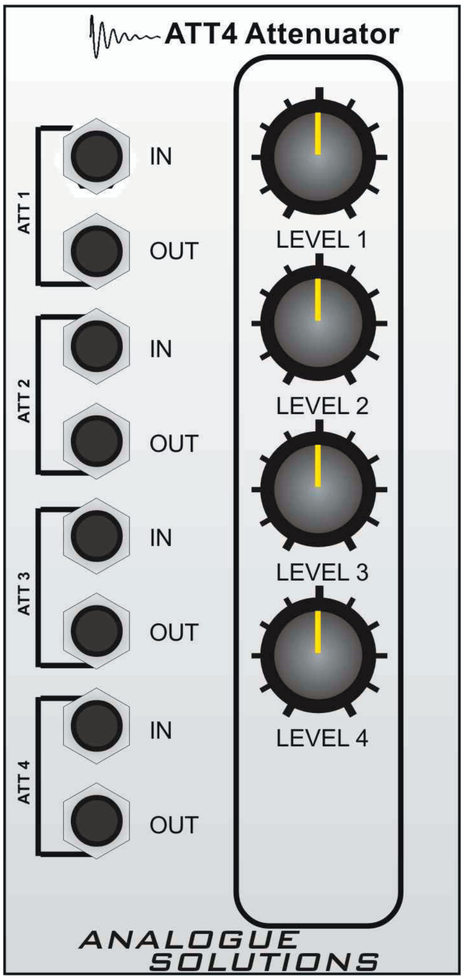
When a signal is split by a passive device such as a multiple, the output power level will be reduced. is generally will not cause a problem as most modules' inputs are buffered.

A multiple can be used as a simple passive audio mixer, but there may be problems of impedance and level matching. It can also be used in some instances to mix control voltages, but this again depends on the design of the modules. With both audio and CV's it is a case of try and see. If it fails to work as desired, no damage will occur.

ATT4 - Quad (4x) Attenuator

Inputs:	Outputs:	Power:
Signal In x4	Signal Out x4	n/a

Controls:	Indicators:	Size:
Signal Level x4	-	12HP



Introduction

4 attenuators.

Four independent attenuators (level controls) in one module. Attenuators are passive controls (they use no power) that are used to decrease the level of an audio or CV signal.

The signal can be changed from full level right down to zero. Use anytime you need to decrease the level of a signal.

A typical example is in conjunction with our MX61 6 input mixer. As this mixer does not have level controls, you can use the attenuators of the ATT4 to adjust the levels before they are mixed with the MX61.

Controls In Detail

The 4 Attenuators are identical.

LEVEL (1-4)
Alters the input signal level

Sockets In Detail

IN (1-4)
Attenuator input.

OUT (1-4)
Output from attenuator.