Working with the minimatrix and the node processors.

A matrix is just a convenient way to connect any of a given number of output signals to any of a given number of signal inputs.
The minimatrix module has six input connectors on the left vertical column and four output connectors on the top row. Input signals are *always* plugged in the left vertical column. The input and output connectors are mono connectors and the crosspoint connectors are stereo tip-ring-sleeve types. These crosspoint connectors or nodes work exactly similar to the insert connectors that you can find on the bigger mixing desks.

When a signal is connected to the input connector of a row this signal is first buffered and then routed to the tip connections in the corresponding horizontal row node connectors. The ring connections in the node connectors in a vertical row are connected to an internal mixing amplifier and the mix of the ring signals in this column is output on the top summing output connector. So, to connect an input signal to a summing output on top of a column all that you have to do is insert a 'shorting jack' in the corresponding node field. This shorting jack has to be a stereo jack where the tip and the ring of the jack are soldered together within the housing of the jack. Your module came with a couple of black shorting jacks with a red coding ring, these are the shorting connectors you normally use to connect a row input signal with a column mix output.
When a second signal is connected to another row input and a jack is inserted in this row and in the same column as the first jack the two input signals will be mixed 1:1 with each other and their mix is available on the column summing output.
To experiment just connect the outputs of two oscillators to two input jacks. Then connect two shorting jacks in the rows of the two input signals, and in the same vertical column. Then connect a cable into the column summing output on top of the column and route it to e.g. your amplifier. You will now hear the mix of the two oscillators.

The simplest use of the matrix is to add signals together in 1:1 ratios and route them to the summing outputs. But there are many more things you can do with the matrix. You should always keep in mind that the tips of the connectors in the node fields are outputs and the rings are inputs. Lets see what happens when a mono jack cable is inserted into a node field connector. The row input signal will be present on the tip, but the ring connector is shorted to ground through the sleeve of the mono connector. This will short the column mixer input for this particular connector to GND, and thus the row input signal will nót be added to the mix output. This will not influence the other nodes in the column because the matirx is fully buffered. Still, you can now plug the other side of the cable into any module input connector and the row input signal will be routed to that module input at the original signal level. This way you can 'recycle' any node connector that is not used to mix to a column mix output as a 'multiple' output. This means that if you do not use the mixing outputs on top of the columns the minimatrix is still a set of six multiples with an input and four fully buffered outputs. You can even route a mono cable from a node connector to the row input connector of another row to expand from four to seven multiple outputs. And of course you can route a mono cable from a column mix output to the row input connector of an unused row to ‘chain’ mixes plus the later described node processor functions. Because, though simple in concept, a matrix allows for the most complex signal routings preceivable.

Now lets see what happens when a stereo cable is connected to a node field connector. At the stereo jack on the other side of the cable the row input signal will be present on the tip. The ring will be connected to an input of the column summing mixer. When shorting the tip and the ring the signal will be passed on to the column mix output. You can test this yourself by using a bit of metal to shorten the tip and the ring on the jack while listening to the column mix output. So, if you would have a footswitch, with a stereo connector that connects the tip and the ring when tapped on, you could switch the input signal in and out of the mix with your foot. Alternatively you could use what is called an ‘insert cable’. This is an ‘Y’ cable with a stereo jack on one side whereas at the other side the cable splits in two mono cables, each with a mono jack. The red (hot) mono plug on an insert cable carries the signal, so it is connected to the tip of the stereo plug on the other side, and the black mono plug connects to the ring, and thus to an internal ‘channel’ of the column mix output. You can connect the red mono plug to the input connector of a volume pedal and the black one to the volume pedal output connector, which will enable you to control the amount of the row input signal in the final mix with the footpedal. If you would now remove the stereo jack on the insert cable from the matrix and plug it into an insert connector on your mixing desk you would hear that the volume control effect on the mixing desk is exactly the same.

Alternatively you can use an insert cable to ‘apply’ e.g. a filtering function to the node. To do so you connect the red mono plug into the input of a filter module and the black mono plug into its output connector. Then you insert the stereo plug into a node that would connect e.g. an oscillator output to a column mix output that goes to your amplifier. You will hear that the oscillator signal is filtered by the filter.

The ‘node processors’ make use of the insert function of the node field connectors. However, to make cabling easier they use cables with stereo plugs on both sides instead of Y-cables. Below the minimatrix you see ten connectors, eight of these are stereo connectors that can be readily connected to a node connector with a short stereo patch cable. However, the TRANSPOSE, VCA A MOD and VCA B MOD connectors are mono connectors that do not connect to a node field connector, but are CV inputs to modulate some of the node functions with a CV signal.

The node processors always connect to a node connector in the node field with a stereo patch cable, don’t use them in any other way as that simply won’t work (unless your experienced enough to realize that with an Y cable you could swap the red and black connectors as inputs and outputs). The node processor connector inputs and outputs are internally protected for short circuiting, so if you ever accidentally happen to connect them in a ‘not so sensible way’ no harm is done.

The available node processor functions are:

1) Two passive audiotaper pots.
2) Two DC-coupled bipolar VCAs or Ringmodulators
3) Adding or subtracting a voltage level to or from a signal
4) Gain control between unity gain and +20dB amplification
5) Realtime note quantizer running at a sample rate of 14kHz.

The audiotaper pots work exactly as in the volume footpedal example, what they can do is set the amplitude of an input signal on a column mix output. E.g. if you connect the outputs of two oscillators to two row input connectors and use two short stereo patch cables to connect two nodes in the same column to the AUDIO A and AUDIO B connectors, you can use the AUDIO TAPER A and AUDIO TAPER B knobs to set the volumes of the oscillators in the final mix signal at the top column mix output of that column.

The bipolar VCAs can be used as normal VCAs, but they can also 'invert' or 'phase reverse' a signal. Both VCAs are DC-coupled and can process both audio and slow CV signals like those from LFOs or envelope generators. For example, connect an oscillator output to a row input connector, connect a node in the row to the VCA A connector with a stereo patch cable and listen to the column mix output above the node the VCA A is plugged into. With the VCA A LVL knob you can set the volume of the oscillator. Note the silence at the twelve o'clock position. Turning the knob towards the right will increase the volume, but turning towards the left will also increase the volume. However, when turned to the left the signal will also be inverted, basically becoming a negative copy of the input signal. You can modulate this knobs 'position' by a CV signal, e.g. plug the output of an envelope generator with a mono cable into the VCA A MOD connector and open the VCA A MOD knob to the twelve o'clock position while setting the VCA A LVL knob also to the twelve o'clock position. This will sound like the normal VCA behavior. But now turn the LVL knob slightly to the left and turn the MOD knob slighly to the right. You will hear that the VCA start to behave differently, almost like there is a slight echo. Then play slightly with the attack and release knobs from the envelope generator and listen to all the slight envelope effect nuances you can thus create.
When a bipolar VCA is modulated by an audio rate signal it effectively turns into a ringmodulator. Just unplug the envelope generator output from the row input connector it is in and instead connect it to the output of a second oscillator. Play with the frequencies of the oscillators and with the LVL and MOD knobs on the bipolar VCA. When the LVL knob is in its twelve o'clock position you get true ringmodulation and you can finely tune the suppression of the original oscillator pitches. However, when the LVL knob is not in its center position you can subtly change the sound towards unipolar amplitude modulation. But forget the technical blahblah and just set the knobs by ear to your taste.
Another nice effect is when you use an envelope signal or LFO instead of the first oscillator and use a LFO to modulate the bipolar VCA. Then connect the column mix output to the cutoff modulation input of a filter. Playing with the LVL and MOD knobs of the VCA and the rates of the envelope and/LFO and modulating LFO will travel you through a whole range of varying filter modulation rhythms.

The OFFSET function can be used whenever you need to lift or sink a CV signal by a manually set amount. There seems to be no apparent main musical use but many times while patching up a sound you may find it a convenient function for one purpose or another. It can also be used as a voltage source by plugging it into a row that has no input signal and getting the CV voltage from the column mix output. To get the clean CV no other plug should be present in this column.

The GAIN function will amplify a signal by a factor set by the GAIN knob. The main function is to amplify the level of an external line level signal to a synth level signal. For instance, you can connect the output of a CD player to a row input connector. This signal level will be well below the output signal of e.g. an oscillator module. Now you connect the GAIN connector with a stereo patch cable to a node connector and you can amplify the signal, the result is present on the top column mix output. Then you can connect the top column mix output to the input of e.g. a filter and the filter output will be similar in level as when you would have filtered an oscillator signal.
You can also use the GAIN function to boost a signal before it goes into e.g. a filter to overload the filter input to cause distortion. The GAIN function soft clips when it is overloaded itself, but at a level that will still slightly overload a filter input.
The GAIN function will also allow the use of stompboxes in your patch. The input level of a stompbox is much lower as the output level of e.g. an oscillator module, and the same goes for the stompbox output signal. If a stompbox would be connected with an 'Y'-type insert cable directly into a node the signal levels would not be compatible and the stompbox input would overload heavily. So, this needs to be done in a different way. For example, lets try to use e.g. a delay stompbox on an oscillator signal. First we need to attenuate the oscillator output signal. To do so you can use the AUDIO TAPER A or B knob in the first column and connect the column mix output of this first column with a mono cable to the input of the delay box. Then you connect the delay output to another row input connector and use the GAIN insert on the node in this row that is in the second column. If you now listen to the second column mix output you can set the stompbox input level with the AUDIO TAPER knob so it doesn't clip, and use the GAIN knob to crack up the stompbox output level to normal synth level again.
You module comes with two black plugs with white coding rings. These two plugs have an attenuating resistor inside that will attenuate a synth level signal by roughly –16dB, producing a level that we found is quite compatible with e.g. Boss brand stompboxes. If you use a white coded plug instead of using an AUDIO TAPER knob you can conveniently save the AUDIO TAPER knob for another purpose in your patch.

The QUANTIZER function will force an input signal to a note in a 1V/Oct note scale, and will do so in realtime. The matrix has one node field connector marked with a white bezel ring that has 0.1% precision resistors to give an accurate unity gain node that won't strech the 1V/Oct curve. It does however have a small offeset voltage and you will have to use the fine tune knob on an oscillator to get back in tune.
Lets plug in a slow triangle LFO signal into the bottom row input connector. Now connect the white QUANTIZER connector to the white bezeled node connector. Then turn the QUANT RANGE and QUANT SCALE knobs fully to the left and connect the column mix output signal from the first column to a 1V/Oct input of an oscillator. You might hear a slight detune in the oscillator which you correct with the fine knob. If you want to use two oscillators you can first connect the column mix output to the 1V/Oct input of the first oscillator, retune it to match the second

oscillator, then connect the chain output of the first oscillator to the second oscillator 1V/Oct input and retune the second oscillator to the first one.
Now slowly open the QUANT RANGE knob, you will hear an ascending and descending arpeggio using all the notes from the keyboard. Opening the RANGE knob fully will arpeggiate up and down over the full ten octave range of the quantizer. There are sixteen quantizer tables, fourteen of them are musical scales and the last two are actually used for audio waveshaping. The note scales are:

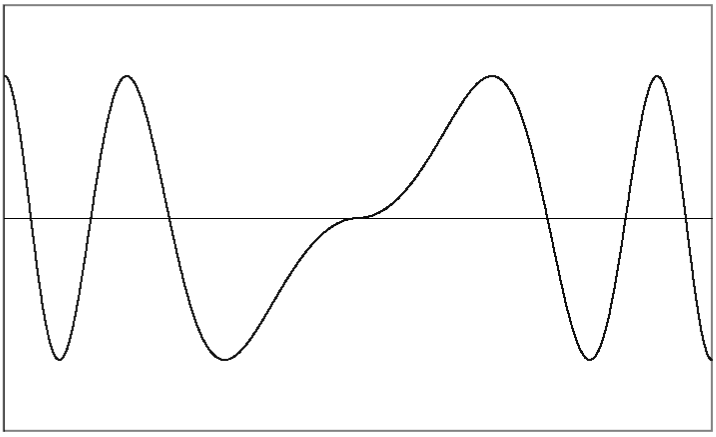
HalfTone	- basically all the white and black keys
Chrom	- only the white keys
C7	- C, E, G, Bes
CM7	- C, E, G, B
Pentatonic	- C0, D0, F0, A0, C1, D1, F1, G1 (repeats every two octaves)
Caug	- C, E, Ais
Cm7	- C, Es, G, Bes
Cm7-5	- C, Es, Fis, Bes
Cdim	- C, Es, Fis, A
WholeTone	- C, D, E, Fis, As, Bes
AugHex	- C, Es, E, G, A, Bes,
Blues	- C, Es, F, Fis, G, Bes
Oriental	- C0, D0, Fis0, G0, A0, Bes0, C1, Es1, Fis1, G1, A1, Bes1 (repeats every two octaves)
Oct	- C0, C1, C2, ..., C10

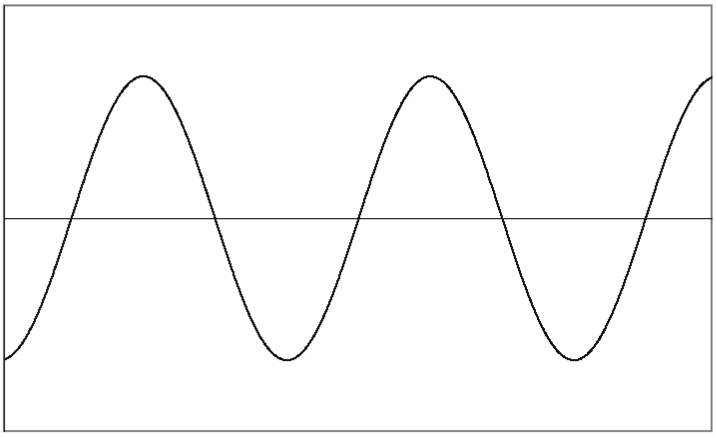
Note that all scales are based on C. Using a 1V/Oct signal on the Transpose input will push the quantizer input up or down by half notes *before* quantization, and thus the quantized notes will remain in the chosen C scale. Adding a 1V/Oct signal *after* a signal is quantized will force the melody in the key played on e.g an analog 1V/Oct keyboard. Note that when using only the TRANSPOSE input in the halftone scale you can correct the CV from an analog 1V/Oct keyboard that has become slightly out of tune.
We did put dots on the frontpanel for the reason that we can burn user specified scales in the scale tables (on user request and with some costs), as long as they are based on whole and half note scales.
The last two 'scales' are actually wavetables to create FM-type sounds, though with a considerable amount of quantization noise. The first table produces a brighter sound while the second produces the more common FM spectrum. You can safely overload the quantizer input for waveshaping purposes by first using the GAIN processor to boost the signal before being waveshaped. To use these waveshaping tables just use e.g. an audio sinewave from an oscillator output as input for the quantizer and open the Quantize Level knob.

Quantizer specs:
Samplerate is approximately 14kHz. Resolution is 8bit. An modern AD converter with a precision ladder network is used to give good monotonicity over the total 10 octave output range.
Input range is between –5V and +5V. Output range is also between –5V and +5V. Zero volt input defaults to zero volt output.

The wavetables for waveshaping are a four period sine tabel (the last scale setting) and a 'fish-eyed' and point symmetrical four period table. Table endpoints soft clip to the minimum and maximum values.

The following graphs are the wavetables used for the waveshaping function:





You can use the waveshaper function with both audiorate and LFOrate signals, or combine them by e.g. connecting an audiorate triangle signal into the TRANSPOSE input and a slowly varying LFO triangle signal into the row input you use for the Quantize function. By opening the QUANT RANGE knob the waveform created by the trianglewave on the TRANSPOSE input will 'glide' through the wavetable at the slow rate of the LFO, thus creating dynamic changes in timbre. If you find the aliasing annoying just route the output through a lowpass filtering function and set the cutoff frequency and resonance amount on the filter to a setting that agrees with you.

Geek talk:
Input impedance on all row and node inputs is 30kOhm. High precision 0.1% tolerance resistors are used and there is a full set of trimmers at the back to precisely cancel offset voltages and gains in all matrix channels. Output impedances are 100 Ohm and everything is short circuit protected.

Create your own shorting plugs by soldering the tip and ring solder lugs in a stereoplug together. Soldering a resistor between the tip and ring lugs will make a node attenuate:

Value	Attenuation
15kOhm	approx. –3dB
30kOhm	-6dB
91kOhm	approx. –12dB
270kOhm	approx. –20dB

There are several trimmers in the module, but don’t touch them. If you think your module needs retrimming ask us directly for the proper trimming procedures.