Guitar Analyser Manual

The Guitar Analyser Eurorack module does a full analysis of the signal of a typical six string guitar. It measures the individual pitch frequency and amplitude of each of the strings. The results of the analysis are output from the 3.5mm jacks of the module. Additionally, the module has several effects that can be activated to the synthetic signals using the analysis results. The module can be used as an input interface to bring the guitar signal to a Eurorack system, or as a stand alone guitar synthesizer generating signals that are impossible to create using typical filtering based modules. It is 95.5mm wide, occupies 19 HP units.

Inputs

guitar in: ¼” jack for connecting the guitar. Use the neck microphone of the guitar; typically a humbucker mic is more reliable in the note detection than a single coil.
pedal: ¼” jack for guitar effect pedal input. The special effects are adjusted either by a pedal or the front panel “effect adjust” potentiometer. The pedal is only activated if it is connected to the module when power is switched on, and the pedal position is not at zero voltage. Use a pedal that has the output voltage coming from the tip of the TRS plug. The module feeds its voltage to the ring electrode of the pedal potentiometer. E.g. the M-Audio effect pedal works when its select switch is in the “M-Audio” -position.

Outputs

Left: six 3.5mm output jacks for the measured pitch in the **1V/octave** format (default), or alternatively jumper selectable linear voltage output option. The range is adopted to the guitar frequency scale, so that in the 1V/octave case the note C3 (130.81Hz) corresponds to 1Volt, and the lowest 0.167 Volts corresponds to a drop-D2 tuned pitch. The maximum 4.17 Volts corresponds to the 22nd fret D6 (1174.7Hz).

In the **linear scale setting**, C3 stays at 1V, but then C2 is 0.5V, C4 is 2V, and C5 is 4V. The maximum voltage 4.8V is reached a little above D#5, and the notes above that will saturate the output. This is because the D/A converter has 12bit resolution, and a wider span would have resulted too coarse resolution at the low pitch end. Now it is relatively decent 3 cents. The advantage of the linear scale is the fact that linear VCO oscillators are much simpler and cheaper than log scale ones, single chip ones are easy to build even by a beginner hobbyist.

Right: Six 3.5mm output jacks for the synthetic signals that follow the amplitude and pitch of the analysed separate strings (default), or alternatively the string amplitudes only, when the effect adjust potentiometer is at its maximum position. They are DC connected, with a 2.4V offset; therefore the outputs can be used for modulating other modules that need a positive control voltage between 0 and 4.8 volts.

One 3.5mm jack “**out**” for composite signal, i.e. all the six synthetic signals summed together. it is an AC connected low power output to be connected to a power amplifier input or to the inputs of other modules.

Power

The module uses only the +12V and GND connections of the Eurorack bus, the current consumption is 150mA maximum.

Controls

pedal effect select: A three position switch that applies the special effect shown at the right side of the switch individually to each of the synthetic signals that are generated based on the analysis results of the guitar signal. The base signal is pure sinewave. See later sections for detailed description of the available effects.

channel allocation method: A three position switch that determines how the analysis results are output to the six left and right side 3.5mm jacks. See a later section for detailed description of the available methods.

e**ffect hold:** A two position switch that in the “on” position allows stacking of the effects, i.e. when switching from one effect to another, the remaining controls of the previous effects at the switching moment are left as they were. Keep it in the “off” position for outputting of the current effect only.

compressor gain: A potentiometer that determines the string signals' gain before the limiting compressor. If the gain is adjusted high, the signals are limited to constant maximum amplitude even at a low input picking strength. This control is essential when one desires to get dynamically evolving output signals, since the FM and distortion effects are controlled also by the string amplitudes. Each string has a green led next to its output showing when the corresponding string has reached the limiting amplitude that also limits the amplitude dependent effect. So, in order to keep the signal varying dynamically with the picking strength, avoid too early limiting and adjust the compressor gain low enough. Instead adjust the output volume from the center potentiometer. However, the limiter itself does not distort the signal further, the leds just show when the signal remains constant at the maximum level.

effect adjust: The use of this potentiometer depends on the pedal connection status when power is switched on. If the pedal is connected and not at zero voltage at that time, the effect adjust potentiometer controls the gain of the pedal control. If the pedal is not connected or positioned for zero voltage at startup, the selected effect strength is determined solely by this potentiometer. **NOTE: if the effect adjust potentiometer is at the maximum clockwise position, the right side outputs will supply only the string amplitude signals, not the synthetic oscillator signals.** In that case, the effect adjust function is only available from the pedal, and the oscillator signals are only output from the composite out jack.

threshold: this control determines the threshold level of the guitar input signal that enters the analysis SW. Adjust it for proper level for your guitar: if too high, weak picking may be lost, if too low, spurious signals may be detected. You may also need to adjust the guitar volume knob for good detection.

volume: This is the only analog control of the module, a typical final amplifier gain control for the common "out". It has no control on the six voltage outputs, for them you need to use the compressor gain and/or guitar volume potentiometer, if needed.

Jumpers

There are four removable jumpers that can be accessed through the openings made in the backside protective metal shield. When viewed at the backside, the leftmost jumper **JP1** determines the logarithmic/linear characteristics of the CV notes outputs. Keep it installed for the 1V/octave outputs, remove if you want the linear outputs.

Two jumpers determine the transpose multiplier of the output. When both are inserted, the multiplier is 1.0, i.e. there is no transpose. The first, **JP2** is just above JP1. If you remove only it, the pitch multiplier is 2.0, i.e. one octave upwards. The second jumper, **JP3** is on the right side, it is the very lowest position jumper. If you also remove JP3, the multiplier is 0.25, i.e. two octaves down. If you keep JP2 but remove JP3, the multiplier is 0.5, i.e. one octave down. Please note that the CV note output values are not affected by the transpose factor, only the composite out terminal signal and the six outputs on the right side.

Finally, the upper right side jumper **JP4** removal allows the direct guitar signal to be output from the "out" jumper. All the six left side CV note and the six right side outputs are not affected by JP4, they remain as described above. Jumper changes should always be done with power disconnected.

Channel allocation method

The three alternatives how the separate string analysis results are output to the six left and right side 3.5mm jacks are described in the following.

The **“picking order”** alternative allocates the notes starting from number 1 channel in the timely order in which the guitar is picked. There is no pitch or amplitude based ordering. If any of the string amplitudes decays under detection, the corresponding channel is released to be used for any new note entering, otherwise the channel remains unused. The main advantage of this method is that for any of the channels, the amplitude and frequency will encounter no clicks as long as the same note remains active. The disadvantage is that after a while, the order becomes random in terms of any pitch order, therefore it may be difficult to determine a sensible patch connection setup for any subsequent module in the rack. Another disadvantage is that when a low pitch string is picked, the first note typically is detected at one of its harmonic frequencies (although you do not hear it, because it is rapidly damped), so the base note will be allocated to one of the higher numbered channels and remains there.

As the **“pitch ascending”** alternative suggests, it keeps the output pitch values in an ascending order, with the channel 1 for the lowest note, and packs the active channels starting from no 1 without any unused channels in between. The advantage is obvious when connecting the outputs to further processing modules. The order remains deterministic. The disadvantage is that a new note that has pitch lower than any of the present notes will then push all the higher notes to the subsequently higher numbered output channels. However, the transitions are softened by suitable interpolation, and this allocation method therefore is often the most practical when considering patching the signals to other modules.

The **“skip zeros”** alternative is otherwise similar to the **“picking order”** method, but it additionally removes any silent channels between active ones and packs the remaining ones towards the lower numbered end of the six channels. The main advantage is that typically, the maximum number of notes in guitar chords is four, so a practical patch cable setup to other modules may easily leave the channels 5 and 6 unused. The disadvantage is that there are transition periods in case of the packing operation is done, however, not in the case when a new note in any pitch is entered; it will just be allocated to the lowest number available channel.

NOTE: Regardless of the allocation method selected, the left and right side channels keep their correspondence.

Pedal effects The three position toggle switch selects one of three possible effects to be applied to the synthetic signal, **FM, vibrato** or **distortion**.

FM is the classical FM modulation effect used in many synthesizers. The base signal without any effects is pure sinewave without any harmonics. When the FM effect is applied, several harmonics will appear, depending on the modulation strength. The strength depends on two factors. The effect adjust or pedal will give a common modulation strength coefficient to all the outputs. In addition, the amplitudes of the individual strings will also control their individual modulation strength up to the maximum set by the pedal or effect adjust potentiometer. A third important control is the compressor gain potentiometer. When any of the string amplitudes reaches the compressor limit, shown by the led next to its output, then obviously the string amplitude remains constant and it has no further effect on the modulation strength. Therefore, in order to keep the sound dynamic, adjust the compressor gain so that the signal is not overly compressed. There is, however, no clipping when the output is limited, it just remains spectrally constant.

The **distortion** effect is a simple effect that uses clipping and pulse duty cycle adjustment to create more harmonics than the base sinewave. It is useful as an input for additional waveshaping modules in the Eurorack. It has similar amplitude dependent strength control characteristics as the FM modulation method.

The **vibrato** effect needs to have the pedal connected. When the **effect adjust** potentiometer is at zero, the pedal will control the upwards pitch bend, up to a one fourth interval. Otherwise the potentiometer controls the vibrato frequency and the pedal the strength, i.e. the deviation of the vibrato frequency around the base pitch.

Limitations The string pitch analysis has some limitations, because all the six string signals come stacked together in a standard guitar pickup signal. The main limitation is that the software does not recognize such string notes that fall on the harmonics of a lower base note. The practical limitations are mainly relevant only with the octave interval, and the octave + fourth (=3rd harmonic) -interval, i.e. a note blocks all its higher octave notes, as well as notes above the octave + fourth pitch. In order to not unintentionally block higher harmonic notes, the player should dampen those lower strings that are not played, and/or adjust the ”threshold” knob more clockwise.

The frequency resolution vs. amplitude has limitations as well: the string detection software resolution goes down to one seminote, but a strong neighboring note may obscure a weaker note +- one seminote apart. Another limitation comes from the delay of the analysis. A few cycles are needed to find the frequency of each note. The frequency resolution and latency are tied together: higher resolution inevitably means longer latency and vice versa. Therefore we have tried to achieve a good tradeoff between an acceptable latency and good enough note discrimination resolution.