HG-16® Instructions for use

Our HG-16® synthesizer gets its name from its principal function where 16 harmonic frequencies are generated atop the base note. However, it has a total of 14 different modes of operation, mostly using additive synthesis, but some modes also apply filtering operations for subtractive synthesis. Additionally, four different variations of the classical channel vocoder are included. HG-16 follows the Eurorack compatible modular synthesizer standard and is not a stand-alone module, but requires external power from the rack, note determining external voltage, gate trigger, and preferably several external envelope generators. These instructions cover the power requirements, external connections, common operating characteristics, and then instructions for each of the 14 synthesis modes. These instructions are quite detailed, and for experienced users, there is a more compact section at the end for quick setup of different operating modes, see "Quick mode setup".

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Power requirements

+5V, 220mA (steady state), 250mA (flash memory operations)
+12V, 40mA
-12V, -40mA

All these power lines have to be supplied from the Eurorack bus using the flat cable of the unit. Please observe the correct polarity of the cable to avoid short-circuiting the power supply. When connecting to a standard Eurorack bus, the ribbon cable should not be twisted: the red marking should correspond to the -12V bus line that is the bottom terminal in the bus connector. The +5V in the bus is not a standard feature in many racks, so please check that your rack has this option installed.

An important detail is related to the module connection order. Because the power return ground and signal ground lines are connected in the Eurorack bus, you need to connect HG-16 closest to the power supply in the rack, and the note determining (typically MIDI) interface to the next slot. Otherwise the ground voltage differences may cause tuning errors.

The system diagnostics is run at power up, and to show that everything is ok, the two red leds (one right to the **volume** knob, and the other left to the **mode select** knob) will blink successively, and then both should go off.

Dimensions

The HG-16 front panel dimensions are 121.6mm x 128.4 mm, so it occupies 24 HP units horizontally. The depth is 25.0 mm from the front panel bottom level, and from that level the highest knob extends 16mm upwards.

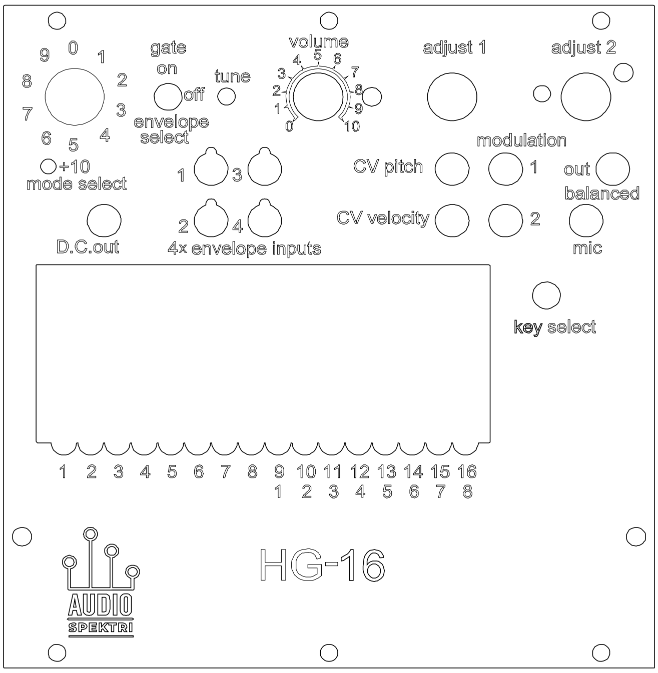

Figure 1: The HG-16 front panel texts and outlines.

Front panel texts

The front panel texts are shown in the picture. To make it easier to find the controls in these instructions, all the words that can be found on the front panel of the synthesizer are emphasized in **bold** throughout the whole text.

A short list of the front panel texts and associated controls follows. Because several controls have different functions in different operating modes, you need to go to the actual mode to learn their specific detailed functions.

mode select: a 10 position rotary selector for selecting one of the 14 operating modes. The light emitting diode under the **+10** opening shows modes over 9.
gate on/off/envelope select: 3 position toggle switch, routes the bus-connected gate signal to the synthesizer, also used in the virtual switchboard programming. Mostly used in the **off** position, see the mode instructions and virtual switchboard instructions for details.
tune: fine tuning; typically not needed for user adjustment.
volume: main volume adjustment.
adjust 1 and **adjust 2**: mode specific adjustments, multi-turn digital encoders.
D.C. out: constant voltage output, useful in many modes as a constant “envelope” output.
4 x envelope inputs: inputs for external envelope control, can be routed to any of the 30 sliders for dynamic parameter control using the virtual switchboard, positive voltage ranges.
CV pitch: input for the pitch bend control voltage signal; maximum +2.5V, minimum -2.5V.
CV velocity: input for the key pressure velocity control voltage signal.
modulation 1 and **2**: mode specific modulation inputs, positive voltage range, typically connected to external envelopes.
out balanced: main output, mono, balanced output. The output D/A has 24bit accuracy, and working with the standard 44.1kHz sample frequency, except Mode 2 has a nonstandard frequency.
mic: microphone input, used only in the vocoder modes.
1 – 16 optical potentiometer sliders, typically for harmonic amplitude adjustment, but also for mode specific functions.
For some modes, the arrangement uses two separate 8 slider sets, therefore there are also duplicate numbers for the sliders starting again from the 9th slider onward.
key select: spring loaded switch, used both for selecting the key to be played and +1 octave transpose (power-on default is C major/A minor, and base note C1). Simultaneous keyboard key activation and switch pulling is needed for its activation. For +1 octave press any key in the current highest octave; for resetting back, press any key in the lowest octave, or do power cycling. The transpose does not work in the vocoder modes.

External connections and additional required modules not included

Most of the input jacks are compatible with mono 3.5mm patch cable jacks typically used in modular synthesizers. The only exception is the microphone input **mic** that accepts a differential 3.5mm TRS (i.e. “stereo”) jack, and used only in the vocoder operating modes.

The main output has a balanced 3.5mm TRS jack. Only connect to an amplifier that also has a balanced (=differential) input, otherwise the noise level is unacceptable. Non-balanced stereo or mono inputs are not compatible.

The external required Eurorack modules that are not included with HG-16 package are:

- A MIDI interface module, with bus-connected 1V/octave note control output and gate on/off signal. There are no dedicated jacks in the front panel for these, so you need a MIDI interface that has the bus-connected gate and note control voltages output, or use a bus breakout module for inputting them to the bus. For the VC note input there is also an alternative path: you can use **envelope 4** if you close a backside PCB switch. The switch can be found through an opening in the metal grid (a single horizontally operated switch). If the switch is closed, then envelope 4 cannot be used for other input voltages.
  The MIDI modules typically also have pitch bend and key velocity control outputs. These can be connected to the corresponding HG-16 front panel jacks labeled **CV pitch** and **CV velocity**, correspondingly.
- HG-16 requires at least one external envelope generator module, preferably with several independent ADSR outputs; maximum of 4 different can be connected. For less complex setups, several envelope inputs and/or modulation inputs can be connected together to a single generator output, so suitable “multiples” rack modules are recommended. However, the delivery also includes two one-to-five 3.5mm modules that can alternatively be used if your rack does not have excess multiples.

Note: The gate input is only available through the Eurorack bus. Typically the gate signal is only used to trigger external envelope generators that in turn are used for various modulation functions through the 4 envelope input jacks and/or through the 2 modulation inputs. Modes 7, 9, and 12 require a direct gate trigger to be input, but they also have an alternative input for the trigger.

Connections and controls in detail

The group of four **envelope input** jacks need to be connected to envelope generator outputs. They tolerate positive voltage from 0 to a maximum of +8 V. In most operating modes the full scale is reached with a 6 V input, and higher voltages will saturate. Their input impedance is 10 000 Ohms, therefore a typical envelope generator can easily drive several inputs simultaneously.
Mainly for testing purposes, HG-16 has one constant voltage output labeled **D.C. out** that can be connected to the **envelope** or **modulation** inputs. It allows playing the unit even without any external envelope generators. However, this means that you have a constant envelope for all the sounds or minimum attack/release duration if the gate signal is switched on.
In most operating modes the minimum requirement is to connect one envelope generator output to input no. 1. but the full functionality of HG-16 calls for several independent generators. Virtual switchboard circuitry is provided. It routes the envelopes to control different features in the synthesizer: see the later section "Virtual switchboard" for instructions.

In the front panel there are also two additional inputs **modulation1/2** with the same input characteristics as the envelope inputs. They are used for various external modulation inputs, and typically also require external envelope outputs to be connected, see their specific use in detail in each of the operating mode instructions.

When connecting patch cables between the inputs and outputs, be careful to always plug the wire first to the inputs, and only after that to the outputs. This is to prevent accidentally shorting the hot plug end to the grounded metal surfaces. It is especially important when using the **D.C. out** -terminal which has a low output impedance of 130 ohms.

The remaining connectors and controls in the **front panel** are the following:

- **volume**, a typical output volume potentiometer.
- **adjust 1** and **adjust 2** are multiturn rotary knobs that have control functions specific to each operating mode.
- Three position toggle switch **gate on/off/envelope select**. The HG-16 assumes that the modular synthesizer gate signal is connected through the rack bus. In typical Eurorack systems the gate signal controls note activation and e.g. external envelope triggering. However, because the HG-16 relies on external envelopes and an external note determining VC input, it can be played both gate connected and disconnected from the bus. The unconnected state (toggle switch center, or "off" position) is mostly preferred, because then you can also utilize the envelope generator release functionality (i.e. when the gate signal would otherwise terminate the note). The operating modes 7, 9 and 12, however, require the gate to be connected, so the toggle switch has to be in the "on" position in these modes. The third position of the toggle switch, "envelope select" is described in the section on the virtual switchboard.
- 16 optical slider potentiometers. As expected, they are used for setting the amplitude of the 16 harmonics in the basic mode, however, in the various operating modes, they may have other uses, see the mode documentations.
- Finally, the **mode select** rotary selector is used for selecting one of the 14 possible operating modes of the synthesizer. Note that modes over 9 have the led "+10" illuminated.
  NOTE: rotate the selector rapidly from 9 to 0 or from 0 to 9, otherwise intermediate spurious switch states prevent moving to or from the states over or under 10. However, stay at least 1 second at 9 or 0 before the +10 border crossing, because the selector has a rather slow debounce filter.

In addition to the front panel controls, there are also three ground lift switches at the **backside**: They can be found at a larger opening in the steel grille, at the top of the enclosure. See their use in the section "Grounding and shielding". Additionally, there is a horizontally placed single switch that connects the Eurorack bus VC note and the front panel envelope 4 input. It allows the use of e.g. MIDI modules that do not have the note determining voltage accessible through the bus.

List of Operating modes

The HG-16 has 14 different operating modes, selectable by rotating the **mode select** knob. The mode selector only has 10 positions, therefore modes over 9 are indicated by the light emitting diode to the left of the selector. When rotating the selector CW from 9 to 0, the diode is lit, and you need to add 10 to the mode selector number. Correspondingly, when rotating the knob counter-clockwise (CCW) back from 0 to 9, the mode number goes back under 10. The available control knobs, sliders, and patch cable inputs may have very different functions in each mode, so studying each mode instruction separately is needed. Note that there is some delay in the "+10" led, so avoid the rapid turning of several positions when going over 9 and under 10, otherwise the +/- 10 mode switching may not react.

The modes are listed and shortly described here; see the later sections for a detailed description of each mode.

Mode 0: 16 harmonic frequencies with individual envelope control for each; common modulatable PM synthesis and phase noise for each harmonic.
Mode 1: 8 harmonics, with individual envelope control for each, and individual modulation depth control (noise and/or PM) for each using the remaining 8 sliders.
Mode 2: Harmonic filtered noise bands from a voltage controlled comb filter, with individual envelope control for each.
Mode 3: Base note with selectable interval notes, maximum 7, each 8 harmonics.
Mode 4: Pseudo-VCF with noise control.
Mode 5: True chorus with 10 chorus members, each 8 harmonics.
Mode 6: Single channel vocoder.
Mode 7: Frozen spectrum/transient capture single channel vocoder.
Mode 8: Two channel vocoder.
Mode 9: Frozen spectrum/transient capture two channel vocoder.
Mode 10: Same as mode 0, but only noise with two parameter controls.
Mode 11: Same as mode 1, but only noise with two parameter controls.
Mode 12: 15 harmonics with envelope control from a built-in sequencer.
Mode 13: Pseudo-VCF with PM control.
Mode 14: Frequency modulated harmonic generator.

In addition, location 15 of the mode selector is used for selecting the software assisted tuning and calibration facility.

Virtual switchboard

A key characteristic of the HG-16 is the employment of a virtual switchboard that allows any of the eight **envelope inputs** to be routed to any of the 16 slide potentiometers which then will scale the external envelope signals connected by typical patch cables from external envelope generators. A typical use would be to first determine a steady state spectrum using the sliders (one slider for each harmonic), then the dynamic behavior by the external envelope that was linked to each harmonic using the virtual switchboard. Depending on the operating mode, the sliders may have other control purposes, but in most cases the same modulating linkage can still be used for the alternative controls. The use of the virtual switchboard is described in the following.

At power-up, all 16 sliders will be connected to **envelope input no. 1** through the virtual switchboard. If the user has previously defined some other configuration, it can be reloaded from the nonvolatile memory by momentarily flipping the toggle switch to the position "envelope select", and then returning it back to one of the other positions. Note that the toggle switch has three positions: **gate on** (lever up), **gate off** (lever middle position), and **envelope select** (lever down).

A routing configuration can be programmed and stored for each of the 16 sliders by turning the toggle switch to the position "envelope select" and then moving the selected one of the 16 sliders. Each move either towards the maximum or towards the minimum over a threshold location will advance the selected envelope channel by one. Because the number wraps over from 4 back to 1, any of the four channels can be selected for the harmonic (or other function) corresponding to the slider. The currently selected envelope channel is lit, so the user can follow the selection easily. When you flip the toggle switch back up, the selected channel selection configuration is activated and also stored in the nonvolatile memory where it stays over power down.

When in the programming position of the toggle switch, any number of slider connections can be reprogrammed without the need to toggle the switch back and forth between single slider programming events. Note that only the last moved slider is activated, therefore if you accidentally move more than one slider at the same time, you need to check back that all channels are in the desired order. If you leave the moved slider at the very minimum or maximum after selecting channels, such accidents are not likely to occur.

Note that in order to easily check the actual stored envelope channel configuration, the first move of any slider in the toggle switch down position does not advance the envelope channel selection, but just shows the current stored channel. So when the toggle switch is pushed down, no channel indication leds are first lit. If you then move any slider over its threshold up or down, the previously stored channel is first illuminated for that slider, and if the toggle switch is returned back up without any additional slider movement, no change of the channel configuration has been done. Only if you continue moving the slider up or down after the first move, the channel selection for the slider will be changed and shown with the next illuminated led.

The stored configuration is specific to each of the synthesizer operating modes selected with the rotary switch. However, when you change the mode, the current configuration stays. Only if you flip the toggle switch again, the previously loaded configuration is active for the new mode. This property can also be utilized for the purpose of storing several alternative configurations for a preferred mode. Because there are 10+10 possible different positions to be selected with the rotary **mode select** switch (and only 14 active operating modes), there are also 20 memory locations for the different virtual switchboard configurations. Therefore it is possible to “steal” an alternative switchboard configuration for any mode by turning the rotary selector from the current mode to some other mode, then flipping the toggle switch down and up again, and returning back to the original mode by turning the rotary selector.

Also note, that when the 16 sliders are used for some other purpose than harmonic amplitude determination in some other operation mode, this switchboard channel programming may also apply to such situation. Please refer to the detailed documentation of the selected mode.

Slider calibration

The slider maxima and minima are factory calibrated to follow approximately the decibel scale under the sliders. If for some reason some of the sliders seem to be very different from the others, there is a way to recalibrate them. Keep the HG-16 running, and the toggle switch in the “off” position. For the minimum value calibration, push all sliders to the minimum position. Then locate a hole to the right of the **adjust 2** -knob. Under that hole there is a small push switch that can be activated by pushing with a narrow stick. There should be a clear but low mechanical click when pushing. Keep it pushed for a few seconds and release. Turn power off and on again. Then move all sliders to their maximum positions, and repeat the same action. Now the sliders should better follow the decibel scale. The scale is very coarse, so more accurate values can be found, e.g. if you have a digital sound card with an FFT spectrum display.

Note: Each recalibration requires prior power reset, i.e. you cannot immediately redo calibration without turning power down and up again, nor do successive zero and maximum calibration without power cycling in between! Of course, you first need to check the reason for the need to recalibrate. Because the optical potentiometer sliders do not have dust covers, prevent any debris from entering between them or inside the wedge-like opening in the sliders. If there is dirt in the openings, use gentle vacuuming to clean them.

Tuning

HG-16 uses a precision 12 bit A/D converter to convert the VC note signal from the bus to the note value. Therefore it stays well in tune, and normally tuning is not needed. However, if the input VC note value is inaccurate, HG-16 has means for compensating for the inaccuracies. The two common errors are:

1. An offset error: each C note should correspond to an exact integer VC voltage value. HG-16 scale starts at C1 = 32.70Hz which corresponds to an input voltage value of 0V. A constant difference from the integer values along the scale will make an offset error.
2. A scale error: the input should follow accurately the 1V/octave scale. If the scaling is incorrect, there is an increasing difference from the correct note when going the scale up.

HG-16 has a front panel opening below the text **tune**. Under it there is a trimmer for fine adjustment for compensating possible input scale error. This is typically sufficient for normal situations. If not, there is a software facility for adjusting both the offset and scaling errors. Errors can also occur in the pitch bend signal (**CV pitch** input), so they can also be compensated in software. This is described in the following.

Software tuning
For software tuning, turn the **mode select** -knob to position 15. The output signal amplitude is now controlled only by the **volume** knob. Only sliders 1 and 16 are functional in this mode. When **slider 16** is fully down, you can compensate for VC note errors. When it is fully up, you can compensate for errors in pitch bend and/or adjust the pitch bend range. Start by moving both **slider 1** and **slider 16** fully down and connecting the output to a precision tuner. Play C1# in the keyboard (not C1, because C1# allows you to observe both positive and negative offset errors). The offset error can now be zeroed by turning **adjust 1**. Only the internal A/D offset is compensated, so the adjustment range is small. For large offset error compensation, see the pitch bend offset compensation section later. Next, play a high note, e.g. C5, and use **adjust 2** for possible scale error compensation. If no pitch bend corrections are needed, you can now raise **slider 1** fully up. This move will burn the new calibration values to the nonvolatile memory.

Pitch bend tuning
If pitch bend adjustments are needed, keep **slider 1** down and move **slider 16** fully up. Connect your pitch bend signal source to **CV pitch.** Check that the pitch bend signal is inactive. Use **adjust 1** to compensate for possible pitch bend offset. Activate the pitch bend signal and use **adjust 2** for adjustment of the desired range. Again, when **slider 1** is moved up, the values are stored in the nonvolatile memory.

Warning: The sum of VC note and pitch bend offsets make the total note offsets. Therefore pitch bend offset can also be used for compensating such large offset errors in the VC note signal that are out of its own adjustment range. However, then you have to remember that when you disconnect the **CV pitch** source, the tuning is different. Also, if you have adjusted the pitch bend offset, you have to redo the VC note calibration process described above. Because during this tuning process it would be impossible otherwise to find out which offset is wrong, there are additional means for hearing the difference: With zero or disconnected **CV pitch** bend, the output signal has both the fundamental and second harmonic at equal amplitudes when the pitch bend signal is close to zero. Otherwise only the fundamental frequency can be heard. Therefore, you can find the ideal situation when there is zero voltage at the pitch bend input - and it thus has no effect on the note - by adjusting **adjust 1** when **slider 16** is up until you hear both. If you need to adjust the pitch bend offset so large that only the fundamental is heard, then you have to remember that disconnecting pitch bend will change the tuning.

Grounding, shielding and RF interference

Inputs

The Eurorack community is diverse, therefore the common grounding and signal shielding practices may not be obeyed in all environments. There should be only one common ground between the different modules in a rack, the bus ground strip. Multiple grounding points may generate ground loops that add noise to sensitive inputs. However, typically patch cables also carry a signal ground in their sleeve contact. This is usually not a problem for such low output impedance, large amplitude signals that are applied to e.g. the envelope inputs within the same rack. However, in a case when signals from a remote rack are connected to the HG-16 inputs, large ground loops may be created. The unwanted additional ground usually comes from a common power jack safety ground connection, so the obvious remedy is to use systems with fully isolated power supplies. The secondary remedy is to use the HG-16 **ground lift switches**. Note that there should still be at least one common signal ground, and hopefully close to the most sensitive input. Therefore it may not be a good idea to lift all the switches, because then the possibly remaining common ground may be close to some noisy power supply lines. The HG-16 ground lift switches correspond to three regions in the circuitry. If you keep the circuit board positioned so that the ground lift switches are up, and look from the backside of the system, the leftmost switch (no. 3) is connected to the **Modulation 1** input ground, the middle one to **Modulation 2, CV pitch**, and **CV velocity** grounds, and the third to all the **envelope input** grounds. The switch levers are very short, so use a small sharp tool for lifting the levers.

Outputs
The HG-16 outputs have their signal grounds permanently connected. Because the signal output has a balanced differential output, it does not necessarily need a separate signal ground, so if there are noise problems, use the receiving amplifier ground lift switches. Or, if there are none, you can even disconnect the signal cable shield from the ground (but only at the output side).

RF/EM Interference
The HG-16 has high speed digital circuitry, therefore special concern has been paid to prevent high frequency emission to sensitive analog circuitry inside the same rack or outside it. All the inputs and outputs have RF filters tuned to block high frequency signals. The module has grounded metal shielding on both sides: the front panel is aluminium, and the backside has an iron grille. The exposed front opening at the sliders has the slider support bars also grounded. When connecting the module in a (metal) rack, the attachment screws also connect the chassis to the rack metal parts that are typically grounded, but separately from the signal ground.

PM vs FM
The HG-16 has two different periodic modulation methods, phase modulation (PM) used in several modes, and frequency modulation (FM) used only in Mode 14. The difference of these methods in terms of resulting signal characteristics is described in the following.
As the name suggests, PM varies the instantaneous phase of the sinusoidal signal by the modulating waveform. The result is a controlled distortion that can be heard as increasing harmonics with increasing modulating signal amplitude. Because several of the operating modes have the possibility to add independent modulation to distinct harmonics, each harmonic will correspondingly be distorted, therefore creating very spectrally rich signals within separately controlled spectral areas. Because the modulation is limited inside the base frequency period, all the distortion components will remain harmonic (except for some modes that allow signal folding/aliasing with extreme modulation).
FM adds frequency shifting modulation over an arbitrary period that is not limited within the modulated signal period like in PM. Therefore the resulting spectrum is not limited to harmonics of the modulated signal. Most percussion instruments have nonharmonic spectra, therefore FM is useful for generating percussive sounding signals, e.g. when the modulation comes from an envelope generator signal with steep amplitude variations. In fact, many instruments that are periodic in their steady state, have complex nonperiodic spectra within their attack and/or decay periods. Such periods can easily be generated in the HG-16 because the modulation parameters can be controlled with external envelope generators, even independently for each generated harmonic.

Operating modes in detail

Mode 0: 16 harmonics, global noise and phase modulation

The first operating mode generates 16 harmonic frequencies with individual envelope control for each using the corresponding 16 sliders. In addition to the nominal amplitude set by the sliders, each harmonic can be dynamically modulated by an external envelope generator output when connected to one of the 4 envelope inputs. The process of determining which one of the 4 possible envelopes is being used for each particular harmonic has been described earlier in the section "Virtual switchboard". Additionally, there is the possibility to add phase noise for each of the harmonics. The noise can be dynamically controlled by connecting an envelope signal source to the **modulation 1** -input. The modulation strength of the envelope is adjusted using the multi-turn knob **adjust 1**. Turning the knob CW will increase modulation strength, and decrease when turned CCW. Reaching the maximum or minimum values will blink the overload led next to the **volume** control, then further turning has no effect.
In the same way, there is the possibility to add phase modulation (PM) synthesis modulation using input **modulation 2**, and adjust its strength using **adjust 2**. For this adjustment, only the minimum rotation is indicated by the overload led. This is because excessive modulation creates a very noisy and spectrally rich signal that could be used for special effects, if not for musical purposes.

The other inputs, **CV pitch** and **CV velocity** are used in the typical way, i.e. the former for pitch bending, and latter for key velocity based amplitude control. These signals are available from typical MIDI input modules (not included).

Note that adding simultaneously both noise and PM is also possible. However, in that case, also some constant band low frequency noise will be added, in addition to the band limited voice around the selected harmonic. Some of that noise is filtered away, but not all, otherwise also the lowest harmonics would be filtered out.

Tip: because the settings are quite complex, you can ease testing by connecting the **D.C. out** to **modulation 1** and **modulation 2** and do the adjustments with a constant note VC in. If you have access to a spectrum analysis SW (included in most DAWs), then you can also observe how the spectrum looks while listening. Remember always first to connect the patch cable to the inputs, and only after that to **D.C. out** to avoid shorting the output.

As a summary, here is a list of the patch cable connections required:

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired harmonic sliders. (see the section "Virtual switchboard"). Remember that on power up, all are connected to input 1, and only if you toggle the switch to "envelope select" position and back again, the previously stored connections are activated.
2. If you want to use the noise and/or PM modulation features, connect also envelope generator outputs to **modulation 1** and/or **modulation 2**.
3. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

Mode 1: 8 harmonics, individual noise and phase modulation

Mode 1 is similar to Mode 0, with the following exceptions:

- only 8 harmonics are generated
- the noise and/or PM modulation strength for each of the 8 harmonics can individually be adjusted using the corresponding sliders 9 - 16, i.e. they can be used for weighing the input fed to **modulation1 / modulation 2**.

The individual adjustment allows creation of complex spectra that have sections of noisy or clean areas when the noise adjustments are used. The individual phase modulation, on the other hand, adds variable spectra that are harmonics of the selected overtones, instead of the fundamental only.

Mode 2: Comb filtered pseudoperiodic signal

Mode 2 uses a white noise source as an input to a comb filter, thereby creating a pseudoperiodic harmonic signal. The speed of the filter is adjusted using the input note control voltage, therefore the apparent pitch of the pseudoperiodic signal follows the same calibrated note pitch as the other modes.

The comb filtered signal is fed to a 15 channel bandpass filter and its output levels can be adjusted with the first 15 sliders, and modulated by any of the 4 **envelope input** sources, selectable as described before. Furthermore, the comb filter feedback can be adjusted making the output signal more or less noisy: i.e. when there is no feedback, the white noise is fed raw to the 15 BP filters. The filters are also tuned to harmonic center frequencies, so the same apparent pitch still remains, although very noisy. When the comb filter feedback approaches unity, the noisiness decreases and the signal goes closer to a periodic harmonic signal; if only one slider is up and others in the fully attenuated position, the signal is closest to a sinusoid, but obviously still quite noisy.

The comb filter feedback coefficient can also be adjusted and modulated, allowing very dynamic sounds to be made, varying between noisy and more sinusoidal signal with the envelope amplitude. The modulating envelope has to be connected to the **modulation 1** input. The sensitivity for the modulation can be adjusted using the **adjust 2** knob, and the modulation feedback offset value corresponding to zero modulation amplitude using the **adjust 1** knob. The modulation direction can be either from noisy to less noisy, or the opposite direction. This is controlled by **Slider 16** that works in on/off mode: when **Slider 16** is fully down or off, increasing modulation amplitude increases the noise level, and when **Slider 16** is up, increasing modulation amplitude decreases the noise level.

Note that the release rate from high noise to low noise is slow, because the comb filter stabilization takes some time. When moving to the opposite direction the rate is fast.

modulation 2 has no effect in this mode. Furthermore, the Eurorack gate signal is not used in this mode, even if the toggle switch is in the "**on**" -position. Therefore the output level is all the time controlled by the external **envelope inputs** linked to the corresponding sliders.

Notice that because the signal is fully created from noise, the noise floor is also much higher than in the other modes. This is not a flaw but inherent for this synthesis method, because this mode uses subtractive synthesis, and the applied bandpass filters do not have infinite stopband attenuation. You can also note that the noise floor is clearly lower if you move slider 1 off and instead use the other sliders for the corresponding amplitude control of the pseudoharmonics.
This mode has been found especially useful for noisy flute -type of voices. This is demonstrated by several sound samples in the download section, including exaggerated samples. You can also hear a clear difference from the signal of e.g. Mode 0 even if you try to adjust the harmonics noise bands equal. All noise is not created equal!

As a summary, here is a list of the patch cable connections required:

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired harmonic sliders. Only sliders 1 to 15 are usable in this mode. Remember that on power up, all are connected to **envelope input 1**, and only if you toggle the switch to "envelope select" position and back again, the previously stored connections are activated.
2. If you want to have the noise modulation function, connect also envelope generator output to **modulation 1**. If you want to keep the noise at constant level, you can alternatively connect **modulation 1** to **D.C. out a**nd adjust the noise to desired level using **adjust 1** and **adjust 2.** Use **slider 16** to determine the modulation direction.
3. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

Mode 3: Base note with selectable intervals

This mode can generate the whole set of musical intervals within one octave, i.e. maximum of 7 notes. Each note then can have a maximum of 8 harmonics. The first 8 sliders are used in the same way as in the basic form for controlling the corresponding harmonic amplitudes. These are identical for all the notes, but obviously the spectral envelopes will differ, due to the interval distances. The sliders 9 – 15 are used for setting the interval amplitudes above the base note. The numbers below the upper ones clarify the intervals, i.e. 1 (below 9) for the base note, 2 (below 10) the second, 3 for third, etc, up to the seventh. The total maximum of spectral peaks then is 7 x 8 = 56, although seldom all the intervals are used simultaneously, if not for special effects.

The key selection is made by pulling the spring loaded “key select” switch up, simultaneously when pushing down the desired major key in the keyboard. The dual keys naturally follow this setting, i.e A minor also is selected when the C key is pressed, Bb minor also results when the C# key is pressed, etc. The default key at power up is C major/A minor.

Note that because both the harmonics and the chord elements can be modulated by one of the 4 selectable **envelopes** as described before, you can make e.g. quite complex glissando-type effects, when using envelope controls with different durations for the different chord notes, or even make a transition from one chord to another during the activation of a single base note. In the case of rapidly varying envelopes, such activation would be quite tricky to achieve "manually" with a normal polyphonic keyboard.

The slider 16 is not used in this mode, neither knobs **adjust 1**, nor **adjust 2**.

As a summary, here is a list of controls and the patch cable connections required:

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have selected for the desired sliders using the virtual switchboard. Note that both the harmonics (sliders 1 to 8) and the 7 interval amplitude responses (sliders 9 to 15) can be modulated. Remember that on power up, all are connected to input 1, and only if you toggle the switch to "envelope select" position and back again, previously stored connections are activated, or you can program new connections.
2. Pull up the “key select” switch to select the key to be played.
3. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

Mode 4: Pseudo-VCF with noise control

Mode 13: Pseudo-VCF with PM control

Although this mode uses additive synthesis, it is still possible to do signal processing that closely resembles voltage controlled filter (VCF) operation. In vintage analog synthesizers VCF was one of the main attractions. In this mode, the harmonic generator makes 8 harmonics that are fed to a spectrum shaper. The harmonic levels from the generator are controlled using sliders 1-8, while sliders 9 - 16 are used to determine the pseudofilter spectral response in the same way as a typical graphic equalizer would do. The 12 center frequencies of the sliders 9- 16 are listed in the following at the reset condition. (because the sliders starting from 16 also have additional numbers starting from 1, they are also used here in parentheses for convenience):

9 (1): 121Hz
10 (2): 269Hz
11 (3): 550Hz
12(4): 915Hz
13 (5): 1450Hz,
14 (6): 2150Hz,
15 (7): 2930Hz,
16 (8): 3750Hz

The last 4 pseudofilter passbands are controlled by the same sliders as the first 4 ones, **but only if there is some signal at envelope input 3**. If envelope 3 has zero voltage input, then only the first 8 pseudofilter bands are active.

9 (1): 4750Hz
10 (2): 6100Hz
11 (3): 7750Hz
12 (4): 9500Hz

The -6dB points of the "filters" are adjusted to coincide with the neighboring filter -6dB corners. Because additive synthesis allows perfect attenuation, the pseudofilters have zero responses outside their designed width. These zero response corners are set to match the center frequencies (=maxima) of the adjacent pseudofilters.
The VCF function can be activated by feeding a frequency control envelope to **modulation 2**. At reset, or when coming from some other mode, the modulation sensitivity for that input is zero. Only when you rotate the **adjust 2** knob, the sensitivity for the external modulation increases. When rotating CW, an increasing modulating voltage at **modulation 2** will shift the pseudofilter center frequencies higher. Correspondingly, rotation of **adjust 2** CCW from the reset state will shift the center frequencies lower with increasing modulation voltage. As usual, you can alternatively connect the **D.C. out** voltage to **modulation 2** and use **adjust 2** to shift the pseudofilter bands to fixed lower or higher frequencies. The magnitude of the frequency shift in response to the modulation is equal to all the pseudofilter bands.

Note that when compared to real filters, the time domain responses of the pseudofilters do not have relation to the filter steepness. Therefore a very rapid frequency modulation will make slight clicking or arpeggio-type voices that may also be useful for some special effects. Of course, if the adjacent slider differences are small, the spectral changes will appear smoother.

In addition to the frequency shifting -type of voltage control, this mode also allows amplitude modulation of each of the pseudofilter responses. Modulation can be activated for the bands corresponding to sliders 9-16 from any of the 4 **envelope inputs** using the virtual switchboard in the same way as for the harmonic frequencies. Again, note that the reset condition defaults to **envelope input 1** for all the sliders, and the stored configuration or new configuration can be loaded by using the toggle switch as described earlier.

The difference between modes 4 and 13 is their use of additional modulation. Mode 4 has similar noise control available as found in Mode 0. A noise modulating envelope can be connected to **modulation 1**, and the sensitivity adjusted using **adjust 1**. Correspondingly, in Mode 13, they are used for phase modulation and sensitivity.

An interesting feature in Mode 13 is that because the filter is pseudo-VCF and not a real, the harmonics created by the PM are not filtered out based on their own frequency, but instead based on the frequency of the fundamental that was modulated. As an example, if you have a harmonic at a frequency of 550Hz, then its amplitude is determined by Slider 11 (provided no modulation is controlling VCF at **modulation 2** or **adjust 2** is at its reset position). When PM strength is increased by **modulation 1** & **adjust 1**, it creates harmonics at multiples of the 550Hz frequency, but still all these harmonics are also controlled by Slider 11. This way you can get very dramatic spectral effects just by one modulating input at **modulation 2**.

As a summary, here is a list of the patch cable connections and adjustments:

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired harmonic sliders 1 – 8, and for the filter amplitude sliders 9 – 16. Remember that on power up, all are connected to input 1.
2. Connect envelope generator output to **modulation 2** to do the center frequency modulation of the pseudofilters; use **adjust 2** to control its sensitivity.
3. If you want to use the noise features (Mode 4), or PM (Mode 13), connect also envelope generator output to **modulation 1**and use **adjust 1**to set the modulation level.
4. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

- Use sliders 1 -8 to set the harmonic spectra.
- Use sliders 9– 16 to set the pseudofilter spectral response.
- If you want to use the 4 highest bands, connect a signal source also to envelope input 3.

Mode 5: True chorus with 10 chorus members, each having a maximum of 8 harmonics. Also the possibility to generate non-harmonic overtones.

Mode 5 generates 10 different fundamental frequencies, each with a maximum 8 harmonics. The fundamental frequencies are generated close to each other, therefore making a true chorus of 10 members ("true" used here to emphasize the difference from typical chorus effects that loop the same voice to create a similar effect). The difference of the frequency of the individual voices can be adjusted using **adjust 1**, and also modulated using an envelope connected to **modulation 1**. The maximum deviation of the chorus voices from the nominal note frequency is a little more than +- one full note. All the 10 voices will be inside this range, with approximately equal distances. At reset all voices are at the nominal note, and you need to turn **adjust 1** CW to increase the deviation, with some envelope or **D.C. out** connected to **modulation 1**.

The spectra of the chorus voices can be adjusted and modulated in the same way as in Mode 0, i.e. using sliders 1 - 8 and their linked **envelope inputs**. All the voices have identical spectra (save the fact that all the harmonic frequencies follow the same relative frequency shift as their fundamentals).
The last 8 sliders from 9 to 16 are used to control the relative amplitudes of the chorus members, in the logical order from the lowest to the highest. Note that these can also be modulated in the same way as harmonics, i.e. by linking the sliders 9 -16 to some of the **envelope inputs** 1 – 4 through the virtual switchboard. Figure 3 shows an example with 10 chorus members, each having just 3 harmonics.

This mode also has the possibility to adjust the harmonics outside of harmony by turning **adjust 2** CW. The whole group of 10 overtones will be shifted equally. The direction of shift is always towards higher frequencies from the nominal harmonic frequency, up to 1.8 times the nominal frequency. This adjustment can be used to generate voices that resemble natural non-harmonic voices, like bells or drums with proper envelope shaping of amplitudes. The adjustment using **adjust 2** advances stepwise, generating audible clicks, therefore not useful during live performance.
A drawback from the true chorus operation is that such close frequencies inevitably will interfere, i.e. make beating effects with their periods varying inversely proportionally to the frequency difference. If the amplitudes are identical, there will be periodically almost complete cancellation and full double amplification of the beating harmonic (note that they occur at different fundamental frequencies for different harmonics). For avoiding the full cancellation, it may be useful to use different amplitudes for the chorus members by adjusting sliders 9 - 16, and when both the deviation and the individual amplitudes are modulated dynamically, the beating will not be periodic and not so disturbing. If acceptable, also slight shifting of the overtones out of harmony can be used for adjustment.

As a summary, here is a list of the patch cable connections and adjustments:

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired harmonic sliders 1 - 8. Remember that on power up, all are connected to input 1.
2. Correspondingly, Connect envelope generator outputs to to all those of the **envelope inputs** you have activated for the desired harmonic sliders 9 - 16 to control the chorus member amplitudes. If you want to keep the member amplitudes constant, connect the selected **envelope inputs D.C. Out**.
3. Connect an envelope output to **modulation 1** if you want to modulate the frequency deviation of the chorus. Otherwise connect **D.C. Out** to **modulation 1** to keep the deviation constant. With no signal at **modulation 1** there is no chorus.
4. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

- Set the chorus member spectra by sliders 1-8 (all members have identical spectra) and relative chorus member amplitudes using sliders 9 -16. Note that slider 9 controls three chorus members, while the rest 10 – 16 each control a single member.
- Use **adjust 1** to control the sensitivity to the signal at **modulation 1**.
- Use **adjust 2** to shift the overtones out of harmony, if desired.

Mode 10: 16 harmonics with two parameter noise control

This mode is similar to Mode 0, with the following exceptions:

- No phase modulation.
- **modulation 2** is used for modulating the setting of **adjust 2** (see its description below; **modulation 1** still controls the noise amplitude).
- **adjust 1** and **adjust 2** are both used for noise adjustment. Their operation is described in the following.

adjust 1 works in the same way as in Mode 0, i.e. when turned fully CCW (until the overload led blinks) the noise is minimized, and turning it CW noise gradually increases. **adjust 2** works by varying the duration of the noise injected to the periodic waveform. Because the noise acts as an additive phase component for sinusoidal harmonics, increasing the duration beyond the sinusoid period will cause frequency shift. Turning it CCW until the led blinks minimizes the duration, and then the noise is similar as in Mode 0. When **adjust 2** is turned CW, the noise injection period slowly increases. This has the effect of randomly varying the frequency of all the harmonics. **adjust 1** continues to adjust the amplitude of the noise (multiplied by the envelope connected to **modulation 1**), so there will be varying effects when these two are adjusted.
When the noise injection period with **adjust 2** is kept moderate, and the amplitude of noise is varied with **adjust 1**, there will be clicking-type noise with varying strength (modulated by **modulation 1** input). When **adjust 2** is turned more CW, the effect will sound more like random frequency hopping of the harmonics, rather than noise. The hopping period is increased with increased CW turning of **adjust 2**, so the signal spends more time at the random frequencies. Notice that the range of **adjust 2** is several tens of full rotations, so you can hear the frequency hopping slowly slow down when you turn it CW. Again, it is convenient to connect **modulation 1** to the **D.C. out** -terminal for testing or adjustment trials.
The envelope connected to **modulation 2** has the effect of multiplying the setting of **adjust 2.** If it is 0V or left open, the noise injection period remains at zero, therefore **adjust 2** has no effect; the same applies to **adjust 1**/ **modulation 1**. Again, for testing, or if you just want to keep the noise parameters constant, connect **D.C. out** to **modulation 1&2**.

Tip: In other modes that have the phase noise adjustment, the bandlimited noise slowly increases. In this mode the audible effect is effectively multiplied by **adjust 2**. Therefore, if you first do a large rotation of **adjust 1**, and only then turn **adjust 2** CW, the result is very chaotic frequencies within the full audible spectrum. Therefore it is recommended that you work by turning successively a little both **adjust 1** and **2**, and listen how the combined effect varies the sound.

Mode 11: 8 harmonics with individual two parameter noise control

This mode is similar to Mode 10, with the exceptions:

- only 8 harmonics
- the noise modulation amplitude for the 8 harmonics is individually controlled by sliders 9 - 16. Thus selected harmonics can be kept noise free by keeping their sliders at minimum, while others have adjustable noise strengths determined by their slider settings as described in Mode 10.

Mode 12: Harmonic generator with built-in sequenced envelope generator

This mode can generate a signal with 15 harmonic frequencies. Instead of the external envelope generator inputs, an internal one is used, and its output is fed to a virtual delay line so that a maximum of 4 delayed envelope waveforms can be used for modulating the harmonics. No external envelopes can be used in this mode, save the one that may optionally be used for controlling the PM synthesis feature for all the harmonics through the **modulation 2** input.

The internal envelope generator only has control on the attack period of the envelope which is the most important to the instrument timbral characteristics. The release period is fixed to a relatively rapid decay rate.
The knobs **adjust 1** and **adjust 2** are used for the envelope shape determination. **Adjust 1** controls weighing between two alternative attack waveform shapes, the other a monotonously rising waveform, and the other a sinusoidally varying, overshooting waveform. Figure 12.1 shows the resulting envelope shapes with the two extreme adjustment cases when turning **adjust 1**, and one middle adjusted case.

The toggle switch has to be in the **gate on** position to trigger the envelopes. If your Eurorack does not have the gate signal connected to the bus, you can alternatively connect the gate or other trigger source to the **modulation 1** input.

adjust 2 is used to vary the speed of the envelope attack edge together with **Slider 16**. When turning **adjust 2** CW, the speed is increased, i.e the attack period becomes shorter for both the monotonous and overshoot waveforms. It also correspondingly compresses the periods between the 4 sequenced outputs. CCW rotation correspondingly increases the durations of the envelopes. **Slider 16** is used as an adjustment for the trigger delays between the internal envelope generator outputs. Raising the slider will increase the delay between the 4 sequenced outputs. The total duration of the delay also depends on **adjust 2** that controls the speed to run through the delay.

All the sequenced waveforms will have identical shape, only their delay from the gate trigger is different. For all the 15 harmonics corresponding to the sliders, any of the 4 differently delayed, identical waveforms can be selected. The selection is done exactly in the same way as selecting one of the external **envelope inputs** in the basic mode, i.e. by turning the toggle switch down to the position "envelope select", and then moving the selected slider up and down until one of the lit leds next to the 4 **envelope inputs** shows the desired delay. The delay length corresponds to the order number of the input, so if the led next to the input 1 is lit for the moved slider, the corresponding harmonic has zero delay; if it is lit next to input 2, it has 1/4th part of the total delay, and if next to input 4, it has the maximum delay. Returning the toggle switch back up, will activate and store the corresponding delay setup in the nonvolatile memory.
Similarly to the basic mode, when you power the system up, all delays default to 0, corresponding to input 1, and only when you momentarily flip the toggle switch down and again up, the stored delay setup is recalled.
Because no external envelopes are used, you need to keep the toggle switch turned up, in the **gate on** -position for playing in order to get the note start trigger signal, the gate signal, for the envelope generator. Note that the gate signal is only available from the Eurorack bus.

An optional input can be used for **modulation 2**. It controls the phase modulation of all the harmonics equally. With zero voltage or no input, there is no modulation, and all the harmonics are pure sinusoids. There is no other control of the modulation strength like in the basic mode, so for this input, you probably will like to use an external envelope generator with good amplitude control, because the maximum voltage level of about 6 volts will make an extremely spectrally rich modulation result.

Tip: One example of acoustic instrument -type sounds is generated when you have a**djust 1** fully CCW, **adjust 2** turned to make a pretty rapid transient, and a fast attack-decay envelope connected to **modulation 2**. The resulting sound is close to trumpet, because the fast tremolo from the overshooting amplitude envelope resembles the player lip action, and the short PM modulation during the attack period makes an effect that is somewhat similar to the trumpet spectrum variation.

As a summary, here is a list of controls and the patch cable connections required for Mode 12:

1. The external **envelope** generator inputs 1 - 4 need not to be connected (if you have cables already connected to them, no harm, you can keep them connected, but they have no effect).
2. If you want to use the PM modulation feature, connect also an envelope generator output to **modulation 2**. Maybe it is best to first try without any connection. **adjust 1** or **adjust 2** do not have control on the PM modulation sensitivity, therefore the envelope amplitude directly makes the PM.
3. Use **slider 16** and **adjust 2** to control the speed and inter-envelope delays of the 4 envelopes, and **adjust 1** to control the envelope shapes.
4. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.
5. Keep the toggle switch in the **gate on** -position. If your MIDI interface module does not have the gate signal connected to the Eurorack bus, use **modulation 1** as the trigger input.
6. Check the harmonic slider virtual link connections to the **envelope inputs** 1 - 4 because they determine the internal envelope generator delays.

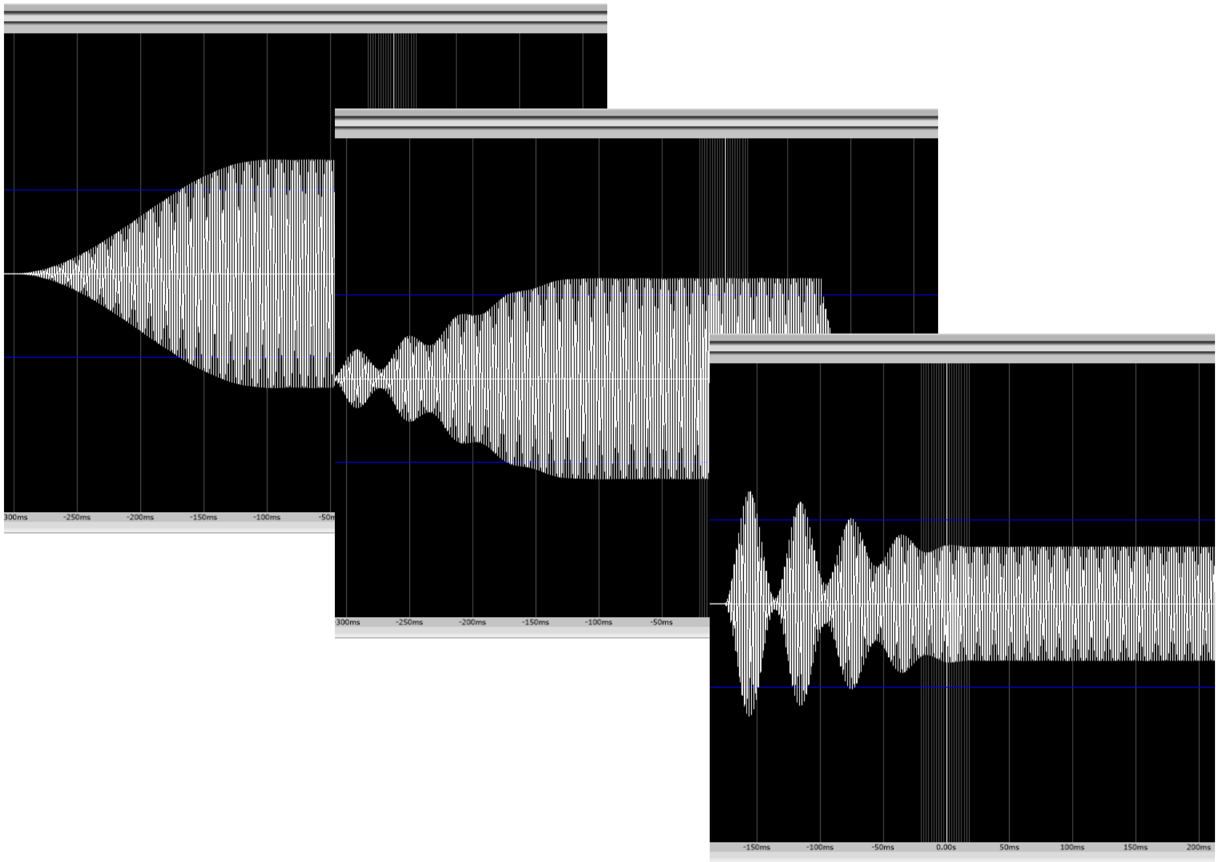

Figure 12.1: Mode 12 is the only mode that uses an internal envelope generator/sequencer. The three envelope plots show the effect of ***adjust 1*** *at the two extreme cases (left & right), and in the middle adjusted case for the attack period of the note. The duration can be adjusted using* ***adjust 2,*** *and each of the 16 harmonics will have a similar copy of the envelope, but with the selection of 4 different possible delay periods from the gate trigger.*

Mode 14: Frequency modulated harmonic generator

As was explained in the section **”PM vs FM”,** frequency modulation of the harmonic generator signal will create also nonharmonic components that are especially useful for generating percussion-type sounds. In this mode, the maximum number of base note harmonics is 6. The harmonic amplitudes can be controlled in the typical way using sliders 1 – 6 and their linked envelope inputs.

The modulating signal will be created using a second harmonic generator. Its harmonic amplitudes can be adjusted using sliders 7 and 8 and their linked envelopes. The modulating signal can have a maximum of 2 harmonics. A global modulation amplitude control can be connected to **modulation 2.** Furthermore, the effect of the modulation can individually be adjusted for each of the 6 note harmonics using sliders 9 – 14, in the corresponding order.

To get any FM, you need to have at least one pair of the corresponding sliders 1 – 6 and 9 – 14 raised, at least one of 7 – 8 raised, and live envelopes or **D.C. Out** connected to **modulation 1 & 2** and to all the **envelope inputs** that you have linked to the activated sliders.

In this mode, the modulating signal is not added raw to the note determining voltage, in the same way as the typical pitch bending signal of a keyboard would do. Instead, each of the 6 harmonics are separately frequency modulated using the same modulator defined by sliders 7 – 8. In practice, this means that the effect of the modulation does not scale with the harmonic number. The principle can be seen in the spectrum plot of Figure 14.1, where only 3 harmonics are activated. The highest and lowest ones are modulated using the same amplitude, therefore having widely spread frequencies around them. The harmonic between them has its modulation strength slider at minimum, thereby yielding no FM. (This picture is from the corresponding mode in HG-30 that has more harmonics available).

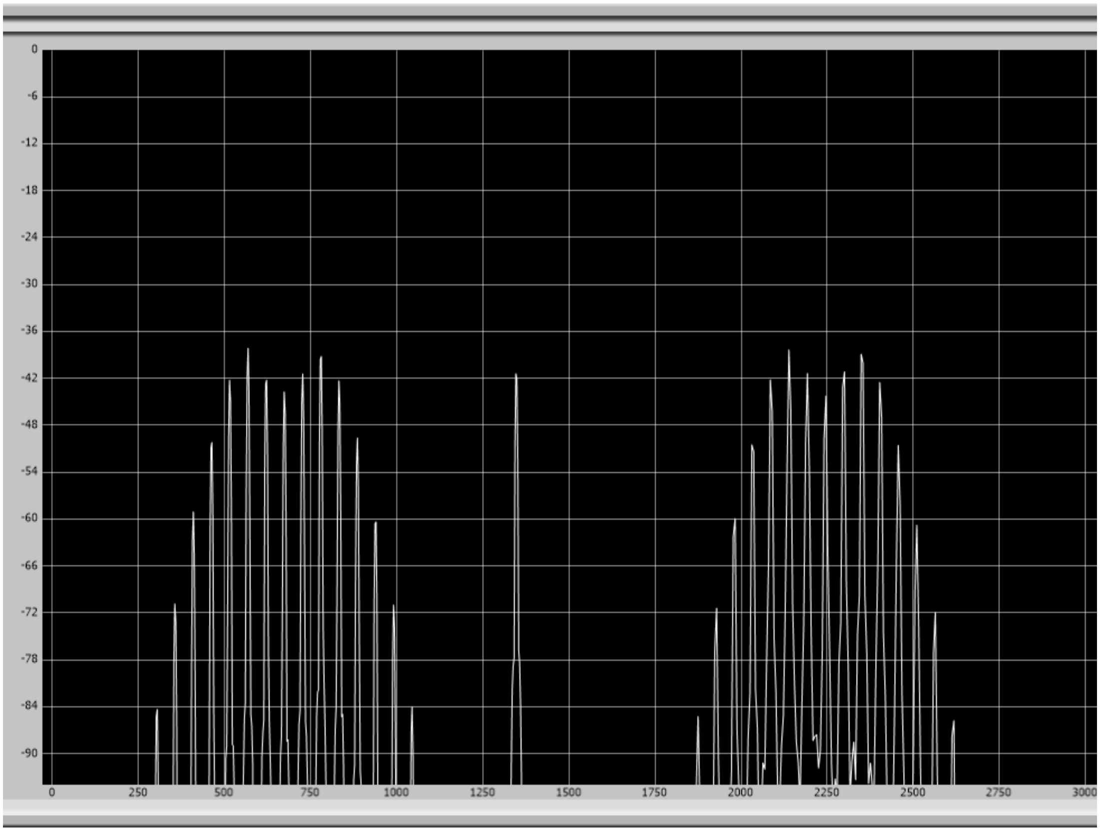

Figure 14.1: A spectrum showings the effect of selective FM: the 3rd harmonic (left) and 10th harmonic (right) are frequency modulated with about equal strength, so they have very similar sidelobes around them, while the 6th harmonic in the middle has no modulation.

The range of modulating signal frequencies is very wide, therefore this mode can be used all the way from slow vibrato/pitch bend -type of effects up to high frequency modulation that makes sidelobes far from each of the modulated base frequency spectral components. Both the modulating signal amplitude and frequency can be varied.

The modulating signal frequency can be adjusted using **adjust 1** and **adjust 2** and modulated using an envelope output connected to **modulation 1**. Specifically, **adjust 1** defines the minimum (offset) frequency for the modulation, while **adjust 2** defines the modulation gain, i.e. how much the frequency is shifted by the envelope signal connected to **modulation 1.** The direction of the modulation can be defined using **slider 16**. When it is off, or fully down, zero amplitude at **modulation 1** will keep the modulating frequency at the maximum that equals to the base note frequency, so it becomes phase modulation (PM). Rising amplitude at **modulation 1** will then adjust the modulating frequency down to the offset limit set using **adjust 1.** The sensitivity for the downshifting can be increased by turning **adjust 2** CW**.** At reset the offset is at minimum and gain at maximum, so if slider 16 is off, you will hear the note pitch swept to a very low frequency even with a very low amplitude at **modulation 1**. Now turning adjust 1 CW will shift the offset higher and you can hear the modulation range to get narrower. Alternatively, you can turn **adjust 2** CCW which decreases the gain, therefore the signal at **modulation 1** shifts the frequency less and less down.

When **slider 16** is up, the modulation direction is opposite, i.e. with zero amplitude at **modulation 1,** the frequency of the modulator is low and will be shifting up with increasing amplitude. Again, **adjust 1** defines the offset frequency at zero modulation, and a**djust 2** the gain upwards.

The modulating signal harmonic amplitudes can be adjusted using sliders 7 – 8. These sliders and their linked envelopes define the modulating signal amplitude, together with **modulation 2**. Through the virtual switchboard, each slider 7 - 8 can be assigned their own amplitude, which is then multiplied by the global modulating amplitude connected to **modulation 2.** In addition, each harmonic has its own modulation adjustment so always the pairs from sliders 1 – 6 and 9 – 14 should be adjusted together to set the corresponding harmonic amplitude and its modulation amplitude.

The chain of possible modulation amplitude controls may appear overly complex, but it allows dynamic control of all the signal aspects:

1. Global modulation amplitude through **modulation 2,**
2. Individual modulating signal harmonic amplitude modulation through the virtual switchboard inputs linked to sliders 7 – 8.
3. Individual base note harmonic modulation strength through the virtual switchboard inputs linked to sliders 9 – 14. (so effectively multiplying the results from 1. and 2. above).
4. And finally the individual base note harmonic amplitude (including its modulation spectra) modulation through the virtual switchboard inputs linked to sliders 1 – 6.

To make it a little simpler, typically it is sufficient to just link all sliders 7 – 8 and 9 – 14 to one envelope input and plug **D.C. Out** to it. Then you just have one global dynamic amplitude control through **modulation 2**, and the modulating spectra and its effect can be statically be adjusted using sliders 7 – 8, and 9 – 14, correspondingly.

Tip 1: When using high frequency modulation, it is typically sufficient to use only slider 7, i.e. pure sinusoidal modulation, because the FM process already generates an ample number of spectral peaks. Also, for the signal to be modulated, it is useful to first start with only the note fundamental (sliders1 and 9 raised). Try first with slider 7 low to hear the modulation increase when it is lifted. Remember to link the slider 7 to a separate **envelope input** so that you can have a dedicated modulation strength controlling envelope for it, or just **modulation 2** globally for both sliders 7 and 8.

Be careful when doing the adjustments, because both the frequency modulating signal amplitude and frequency will modify the signal. Basically, the modulation amplitude adjusts the width of the created group of frequencies, and the modulator frequency the number of peaks inside this spread.
In order to make all the complex variations possible, the anti-aliasing may be partly bypassed using the **slider 15**. When it is off, the antialiasing logic also filters out such excessive high frequencies that result from strong modulation (while the excessive base harmonics are always filtered out). Higher settings of slider 15 filter less. Therefore, when slider 15 is up, use of high harmonics, and/or high pitch notes, and/or high frequency modulation, and/or high modulation harmonics may cause folded frequencies that may or may not be desired for special effects. Again, for easy control setting, it is useful first to connect **D.C.out** to the active inputs, so you can hear the maximum effect that any live envelope will have when playing.

Tip 2: Straightforward FM can be created by feeding external modulation to the pitch bend -input **CV pitch**, because it also has a wide bandwidth, but then the spectral spreading of each harmonic is proportional to the harmonic number.

As a summary, here is a list of the patch cable connections required for mode 14:

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired base note harmonic sliders. Only 6 harmonics are used in this mode, i.e. sliders 1 to 6 are usable for the base signal spectrum setting. Remember that on power up, all are connected to **envelope input 1**, and only if you toggle the switch to **envelope select** position and back again, the previously stored connections are activated.
2. For modulating signal harmonic amplitude control, connect envelope generator outputs to those **envelope inputs** that you have linked for the sliders 7 - 8. Adjust sliders 7 – 8 to set the relative amplitudes for each of the modulator harmonics.
3. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the individual harmonic frequency modulating signal amplitude defining sliders 9 - 14. Check the correspondence of sliders 1 – 6 and 9 – 14.
4. For modulating signal frequency control, connect an envelope generator output to **modulation 1** and adjust the modulation offset frequency to a desired level using **adjust 1,** and gain using **adjust 2.**
5. Define downward or upward modulation by setting slider 16 down or up, correspondingly.
6. Connect global modulation amplitude control envelope to **modulation 2**.
7. Use slider 15 to remove (slider off) or allow (slider up) possible aliased, folded frequencies.
8. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

Mode 6: Single channel vocoder

Quick start guide for the connections:

Input signals:

mic: Microphone input, balanced 3.5mm TRS plug
modulation 1: No function
modulation 2: No function

envelope inputs:

envelope 1: base note spectral shift
envelope 2: No function
envelope 3: No function
envelope 4: No function (optionally CV note input, if the PCB switch is closed)

Outputs:

Output is normally from the **out** TRS jack.

Adjustments:

adjust 1: spectral shift for base note (big speaker/small speaker, envelope 1 has to have voltage)

Slider 1: On/off, automatic/manual noise control. If off, or fully down, the noise characteristics can be adjusted using sliders 2 - 5, otherwise automatic noisiness decision made in the vocoder.

Slider 2: If Slider 1 is off, adjusts the noise level at the frequency band corresponding to the fundamental, otherwise adjusts the sensitivity for automatic noise generation for the band.

Slider 3: Corresponding noise adjustment for harmonics 2&3.

Slider 4: Corresponding noise adjustment for harmonics 4&5.

Slider 5: Corresponding noise adjustment for all the remaining harmonics.

Slider 6: Noise gate: clips the bottom of the analyzed speech spectrum which typically is noisy; increased clipping also removes lower spectral contents for sound coloring.

Slider 7: Spectral shaping; exaggerates the differences of spectral peaks and valleys when shifted upwards from the default off-position.

Slider 8: Control of the interpolation between the analyzed spectral bands. This is an on-off control and its change only comes into effect after system reset or mode change. When slider 8 is off at the minimum position, the interpolation is maximally smooth. This results in decreased spectral resolution, but minimizes possibly unwanted effects with rapid note changes that vary the spectrum. Pulling slider 8 up will improve spectral resolution. The audible differences are quite small, though, in most cases.

Sliders 15 and 16: Microphone digitiser noise gate. In addition to the spectral noise gate adjustment with Slider 6, also the input microphone A/D has noise gating. The default gating from system reset can be varied using these sliders. If you want to increase the gating threshold, move Slider 15 to the maximum position and move 16 up and down between the minimum and maximum positions. Each sweep up or down increases the threshold by 3dB. Correspondingly, when Slider 15 is at the minimum position, each sweep decreases it by -3dB. Overload led blinking shows reaching of the minimum or maximum threshold.

Detailed description

Mode 6 implements a classical channel vocoder that is based on a filter bank spectrum analysis. Modern digital signal processing technology and hardware allows many improvements over the analog designs. Typical analog bandpass filters have uneven group delay resulting from bandwidth narrowing. This makes poor time response, i.e. typical ringing at the center frequency, and/or slow response to rapid spectral changes. In contrast, with DSP we can implement narrow, perfectly linear phase filters without sacrificing time domain characteristics. Therefore all frequencies are passed through the filters with the same delays. This can be heard as a small latency of about 18 milliseconds, but without additional coloring. In our case we have 21 filters, each of order 255, up to the frequency of 5200Hz, covering the practical speech spectrum in high resolution.
Ironically, many of the suboptimal features of classical analog vocoders have created the desired sound effects for music creators. Therefore our implementation has quite many adjustments for finding the desired sound quality. Anyway, the main function of the vocoder is to separate the note pitch from the spectral shape of the analyzed input voice. Then, during the synthesis, these parameters can be arbitrarily modified, e.g. varying timbres, creating a "big singer" by shifting the spectral peaks downwards, or "small singer" by shifting them upwards while the note remains the one played from the keyboard. Other effects include exaggerating the spectral peaks or valleys, flattening the spectrum, and varying the noisiness from fully periodic to completely noisy, whispering sound. Most of these effects can also be modulated by external envelopes.

Our implementation has the specialty that the synthesis is not made using another filter bank, but instead using the harmonic generator, where each harmonic amplitude is weighted by the analysis filter bank output that is closest to the frequency of the harmonic. This is quite an economic method, therefore in the two channel version, the synthesis part could be made with two separate outputs whose characteristics can be separately controlled. The tradeoff, however, is that while the single channel vocoder generates up to Nmax=101 harmonics, the two channel version only generates 53 + 50 harmonics, therefore limiting the maximum frequency correspondingly (e.g. for a fundamental corresponding to C2 (= 65.41Hz), harmonics extend to 3467Hz for the two channel version, when the single channel version range is up to 6606Hz for C2). For both versions, the maximum harmonic frequency is limited below 10kHz, therefore notes whose fundamental is more than 10kHz/Nmax, will have less harmonics than indicated above. Also note that only if the **adjust** knobs are rotated towards the small singer, i.e. shifting the spectrum upwards, the output spectrum contents exceeds the analysis filterbank upper limit of 5200Hz.

Practical operation tips

A short list of required connections can be found in the beginning of the mode instructions. Note that the envelope input 1 needs to be connected to get the corresponding function working. At first it is perhaps best to connect it to **D.C. Out** so that you can try the controls. Testing may also be easier when keeping the gate off. Also make sure that your microphone has a 3.5mm TRS-type (“stereo”) plug with the hot connections in T(ip) and R(ing), otherwise you may get strong hum. A dynamic microphone is recommended. The mic input has autogain so higher output electret microphones may work, but there is a risk of overloading.

Because the vocoder transforms any noise to a periodic sound (especially when the noise controls are off), all background makes easily the output signal to have a “snaring” sound quality. Therefore a good choice for the microphone is a close talking, background suppressing one, e.g. Shure 503BG or similar. For suppressing the snaring background, there are also other means. Pushing Slider 6 upwards will cut the bottoms of the spectral envelopes and thus work as a spectral domain noise gate. Try different settings for adapting to your working environment. Excessive background cutting may also be used for some special effects. Sliders 15 and 16 can be used as a normal noise gate that works by silencing the microphone completely. This adjustment may be used e.g. in live performance to prevent the other instruments to interfere (but then obviously the singer has to use louder voice).

Other tips for more natural voice is to add some noise to the output using Sliders 2,3,4, and 5. Because natural voices typically have more noise at the higher frequencies, try to use 5 highest, and then successively lower settings for the rest of the sliders. If you add higher amount of low frequency noise by Sliders 2 and 3, the effect sounds like a wind blowing to the microphone. For manual noise settings, keep Slider 1 off. When it is on, the “automatic” mode tries to analyze the noisiness of the microphone input and adjust the output correspondingly, with the sensitivities of the bands controlled by Sliders 2 – 5. However, our implementation for voiced/unvoiced is quite simple, and may not work well for all singers. An interesting setting to get a very whispering-type sound is when you have a strong high frequency noise added in the manual mode.

The setting of the spectral interpolation determining slider 8 does not make much difference when the note determining voltage changes slowly. When Slider 8 is on, the spectrum follows the original one a little better, but because the analysis filterbank has good resolution, the fidelity is quite good also when it is off. Smoother spectral interpolation helps to attenuate any possible clicks when the pitch changes rapidly (this is because there is no synthesis filterbank that would smooth the time response, and the harmonic generator has very rapid response).

This vocoder has not been intended to be used as a high quality autotune. This is because the output pitch follows faithfully the note played, rather than just trying to correct off-key singing, and the voice quality changes more than in a dedicated autotune processor. Its use is mainly in special effects due to the versatile means to modify the spectral parameters, and the means for live modulation using external envelopes. Definitely the most dramatic effect is the big singer/small singer (timbre) adjustment and/or modulation, in the extreme cases when a small singer can use very low pitch or a big singer a very high pitch. Also, the transient speedup possibility may be appreciated by rap performers.

Mode 7: Single channel freeze spectrum/capture transient vocoder

The vocoder makes it possible to build a spectrum with even more than 30 harmonic frequencies using a proper input signal to the microphone (i.e. also other than human voices). Mode 7 has been added in order to be able to fix a desired spectrum using the frequency analysis filter bank of the vocoder. It allows three different operations:

1. Freeze spectrum: When turning the mode selector either from Mode 6 or Mode 8 to Mode 7, the instantaneous spectrum is frozen and can then be played and modified using mostly the same controls as in Mode 6.
2. Transient capture: a duration of 1.1 seconds of the input signal can be captured, vocoded, and then played back using the live vocoder controls, and additionally with variable playback speed.
3. Some typical vocal and special effect transients are stored in the non-volatile memory and can be played readily.

In this mode, the controls and **envelope input** functions are equal to the ones in Mode 6, with three exceptions:

1. Because we now have stored data, its replay speed can be varied. The replay speed is controlled using sliders 9 and 10. When **Slider 9** is off (= fully down), raising **Slider 10** will slow down the replay, and when **Slider 9** is not off, raising **Slider 10** will speed up the replay. As you can hear, because in the vocoder the sound parameters are isolated, the replay speed does not change either the pitch or the spectral envelope (as would be the case if the raw original sound sample would be played back).
2. **Slider 1** is used for controlling the capture and storage operations. Thereby the automatic/manual noise control stays only manual in this mode.
3. **Envelope input 3** is used for determining if a stored transient is played or a new transient is captured.

The following instructions explain the three use cases:

1. **Freeze spectrum**: If you want to freeze a spectrum coming from Mode 6 or 8, then keep Slider 1 at the middle position. Sing a desired vowel continuously while simultaneously switching the mode selector from 6 or 8 to 7. Now the spectrum at the switching moment will be frozen and you can continue playing in the same way as in Mode 6 (except that no microphone is needed any more). Leave e**nvelope input 3** open or grounded.

2. **Transient capture**: Keep the three position toggle switch in the middle position (**gate off**) and leave e**nvelope input 3** open or grounded. Then move Slider 1 to its lowest position. Now the vocoder works in the real time mode, as explained in Mode 6, and you can listen to the signal from the output jack. The transient capture starts when raising Slider 1 upwards. The distance from the "off" position now determines the capture sensitivity. Sensitivity increases with increased distance, i.e. when the slider is close to the bottom position, a large amplitude is needed for triggering the capture, and when close to the middle position, even a very small signal in the microphone will trigger. So try to synchronize your speech or other signal with the slider movement, or just try with a suitable sensitivity setting and loud enough voice to automatically start the recording.

   A duration of 1.1 seconds is always recorded. If you want the transient to die away when played, then stay silent at the end of this duration, otherwise the last remaining spectrum is sustained after the transient when played longer than 1.1seconds. Playing the transient needs to have the toggle switch in the **gate on** position, so that each keyboard key activation starts the transient. Please check that your Midi interface has the gate signal connected to the rack bus. The transient is volatile, so it will not be stored over power down.

3. **Stored vocoder transient playback**: The vocoder has 26 fixed transients stored in the nonvolatile memory, containing both human voice and musical instrument spectra. These are 1.1s each. Reading the transients is done in the following way:

   Keep the toggle switch in the **gate on** position. Connect e**nvelope input 3** to **D.C. Out**. Now rotating **adjust 1** CW causes the transients toggle one item forward each time the gate signal rises, i.e. when any keyboard key is pressed. Correspondingly rotating it CCW toggles them backward. The

overload led blinks when either the maximum or minimum is reached. The rotated angle is irrelevant, always the next transient in list is found for one key activation. When you have found a suitable one, keep **slider 1** up, disconnect e**nvelope input 3** and play the sample. Now **adjust 1** comes back to its default function, i.e. big singer/small singer adjustment. Use sliders **9** and **10** to control the playback speed as described above.
If your Eurorack does not have the gate trigger signal connected to the bus, you can alternatively use a patch cable to connect the gate output of an external MIDI module or other trigger source to **modulation 1,** and keep the toggle switch in the **gate off** position. Now you can toggle the transients forward and backward as described above using **adjust 1** and your keyboard.

The list of stored transient is shown in the following table. As you can hear when experimenting with the transients, by default, the vocoder works best with human voices, but still some instruments also come pretty realistic, thanks to the large number of analysis filters and their good time domain performance. The obvious limitations in instrument replay are due to the fact that non-harmonic nature of the signal is not retained which means that most percussion instruments are not reproduced well. Also, the overall spectrum envelope may vary drastically for musical instruments with the key played, and the vocoder only stores the instantaneous spectrum for the specific key used in the recording. This is in contrast to a human singer whose vocal tract for a vowel can remain constant regardless of the pitch of the vocal cords.

Transient number	description
1	“la” with low amplitude at end
2	“la” with high amplitude at end
3	“lalala”
4	“jee” with high amplitude at end
5	“jee” with low amplitude at end
6	“yeah”
7	“gimmegimmegimme”
8	“wau”
9	“ayayay”
10	“rock”
11	“ha”
12	“hello”
13	“beibe”
14	“mymy”
15	“dopido”
16	“cmon”
17	“hey” with high amplitude at end
18	“hey” with low amplitude at end
19	trumpet
20	Mouth harp 1
21	Mouth harp 2
22	trombone
23	harpsicord
24	piano
25	timpani
26	“saynomore”

Mode 8: Two channel vocoder

Mode 8 controls are similar to Mode 6 controls, but because it adds a second voice to the synthesis part, it has a second set of controls for several functions. In addition, it has settings for determining the interval between the base note and the second note. And because the intervals have to follow standard keys, the key selection has been added by pulling up the **key select** switch. Not surprisingly, the default key is C major/A minor.

Quick start guide

Inputs signals:

mic: Microphone input, balanced 3.5mm TRS plug
modulation 1: No function
modulation 2: No function

envelope inputs:

envelope 1: base note spectral shift
envelope 2: 2nd note spectral shift
envelope 3: No function
envelope 4: No function (optionally CV note input, if the PCB switch is closed)

Notes: Vocoder is typically used without external envelope modulation; in that case connect all the **envelope inputs** 1 - 2 to the **D.C. out** output for enabling the corresponding functions.

Outputs:

Output is normally from the **out** TRS jack.

Adjustments:

adjust 1: spectral shift for base note (big speaker/small speaker)
adjust 2: spectral shift for second note (big speaker/small speaker)

Slider 1: On/off, automatic/manual noise control. If off, or fully down, the noise characteristics can be adjusted using sliders 2 - 5, otherwise automatic noisiness decision made in the vocoder.

Slider 2: If Slider 1 is off, adjusts the noise level at the frequency band corresponding to the fundamental, otherwise adjusts the sensitivity for automatic noise generation for the band.

Slider 3: Corresponding noise adjustment for harmonics 2&3.

Slider 4: Corresponding noise adjustment for harmonics 4&5.

Slider 5: Corresponding noise adjustment for all the remaining harmonics.

Slider 6: Noise gate: clips the bottom of the analyzed speech spectrum which typically is noisy; increased clipping also removes lower spectral contents for sound coloring.

Slider 7: Spectral shaping; exaggerates the differences of spectral peaks and valleys when shifted upwards from the default off-position.

Slider 8: Control of the interpolation between the analyzed spectral bands. This is an on-off control and its change only comes into effect after system reset or mode change. When slider 8 is off at the minimum position, the interpolation is maximally smooth. This results in decreased spectral resolution, but minimizes possibly unwanted effects with rapid note changes that vary the spectrum. Pulling slider 8 up will improve spectral resolution. The audible differences are quite small, though, in most cases.

Sliders 11 to 15 define the pitch difference between the two vocoder output channels or notes. The numbers below the upper ones clarify the intervals, i.e. 3 (below 11) for the third, etc, up to the seventh.

key select The key selection is made by pulling the spring loaded “key select” switch up, simultaneously when pushing down the desired major key in the keyboard. The dual keys naturally follow this setting, i.e A minor also is selected when the C key is pressed, Bb minor also results when the C# key is pressed, etc. The default key at power up is C major/A minor.

Sliders 15 and 16: Microphone digitiser noise gate. In addition to the spectral noise gate adjustment with Slider 6, also the input microphone A/D has noise gating. The default gating from system reset can be varied using these sliders. If you want to increase the gating threshold, move Slider 15 to the maximum position and move 16 up and down between the minimum and maximum positions. Each sweep up or down increases the threshold by 3dB. Correspondingly, when Slider 15 is at the minimum position, each sweep decreases it by -3dB. Overload led blinking shows reaching of the minimum or maximum threshold. Note the dual function of slider 15.

Mode 9: Two channel freeze spectrum/capture transient vocoder

Mode 9 is similar to Mode 7 for the dual channel vocoder case, so please follow its instructions for use of **Slider 1, Slider 9, Slider 10, envelope input 3**, and the toggle switch, and otherwise follow Mode 8 instructions.

Quick mode setup

Mode connections and adjustments are listed shortly for each mode in the following. For more detailed instructions, please read the longer mode sections. Also note that instead of the proposed external envelope generator outputs below, the **D.C.out** voltage from the front panel plug can be connected to all the envelope inputs, if you just want to keep the corresponding control voltage constant. If left unconnected, several modes will not output any signal. Always first connect a patch cable to an input, and only after that to an output, in order to prevent shorts. Remember the virtual switchboard instructions: on power up, all sliders are connected to envelope input 1, and only if you toggle the switch to the "envelope select" position and back again, the previously stored connections are activated. Study the section "Virtual switchboard" for storing non-default linkage between the inputs and the sliders.

Mode 0, 16 **harmonics**

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired harmonic sliders.
2. If you want to use the noise and/or PM modulation features, connect also envelope generator outputs to **modulation 1** and/or **modulation 2**.
3. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

- Noise modulation strength using **adjust 1**.
- Phase modulation strength using **adjust 2**.

Mode 1, 8 harmonics, individual modulation control

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired harmonic sliders. Only 8 harmonic sliders used in this mode.
2. If you want to use the noise and/or PM modulation features, connect also envelope generator outputs to **modulation 1** and/or **modulation 2**.
3. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

- Noise modulation strength using **adjust 1**.
- Phase modulation strength using **adjust 2**.
- Use sliders 9 – 16 to individually adjust the noise/PM modulation for the corresponding harmonics 1 – 8.

Mode 2, comb filtered noise

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired harmonic sliders. Only sliders 1 to 15 are usable in this mode.
2. If you want to have the noise modulation function, connect also envelope generator output to **modulation 1**.
3. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

- Adjust noise modulation strength using **adjust 2.**
- Adjust comb filter feedback using **adjust 1.** This effectively determines the noise level.
- When Slider 16 is fully down, increasing modulation amplitude increases the noise level, and when Slider 16 is up, increasing modulation amplitude decreases the noise level.

Mode 3, 7 intervals x 8 harmonics

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have selected for the desired sliders using the virtual switchboard. Note that both the 8 harmonics (sliders 1 to 8) and the 7 interval amplitude responses (sliders 9 to 15) can be modulated.
2. Pull up the **key select** switch to select the key to be played.
3. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

– No function for **adjust 1, adjust 2, modulation 1,** nor **modulation 2**.

Mode 4, Pseudo-VCF with noise control, Mode 13: Pseudo-VCF with PM control

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired harmonic sliders 1 – 8, and for the filter amplitude sliders 9 – 16.
2. Connect envelope generator output to **modulation 2** to do the center frequency modulation of the pseudofilters.
3. If you want to use the noise features (Mode 4), or PM (Mode 13), connect also envelope generator output to **modulation 1.**
4. Optionally, connect the **CV pitch** and **CV velocity**inputs to the corresponding MIDI input module outputs.

- Use **adjust 2** to control of the pseudofilter center frequency modulation sensitivity.
- Use **adjust 1** set the modulation level for noise (Mode 4) or PM (Mode 13).
- Use Sliders 1 -8 to set the harmonic spectra.
- Use sliders 9– 16 to set the pseudofilter spectral response.
- If you want to use the 4 highest bands, connect a signal source also to envelope input 3.

Mode 5, True chorus with 10 chorus members, each having maximum of 8 harmonics. Also possibility to generate non-harmonic overtones.

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired harmonic sliders 1 – 8.
2. Correspondingly, Connect envelope generator outputs to to all those of the **envelope inputs** you have activated for sliders 9 - 16 to control the chorus member amplitudes.
3. Connect an envelope output to **modulation 1** if you want to activate/modulate the frequency deviation between the chorus members. Modulation 2 has no function
4. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

- Set the chorus member spectra by sliders 1-8 (all members have identical spectra) and relative chorus member amplitudes using sliders 9-16.
- Use **adjust 1** to control the sensitivity to the signal at **modulation 1**.
- Use **adjust 2** to shift the overtones out of harmony, if desired.

The vocoder modes 6-9

The vocoder modes are quite different from all the other modes, therefore please study carefully the settings, starting perhaps from Mode 6. Both Mode 6 and Mode 8 have quick setup instructions in the beginning of their corresponding detailed descriptions, so they are not repeated here.

Mode 10, 16 harmonics with two parameter noise control , Mode 11: 8 harmonics with individual two parameter noise control.

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired harmonic sliders.
2. Connect envelope generator outputs to **modulation 1** and **modulation 2**.
3. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

- **modulation 1** is used for modulating the setting of **adjust 1**.
- **modulation 2** is used for modulating the setting of **adjust 2** .
- **adjust 1** and **adjust 2** are both used for noise adjustment. **adjust 1** controls the noise amplitude, and **adjust 2** the noise injection period.
- Mode 10 has identical control for all the 16 harmonics, while Mode 11 has individually adjustable strength for the 8 harmonics using sliders 9 – 16.

Mode 12, Harmonic generator with built-in sequenced envelope generator

1. Connect external envelope generator output to **modulation 2** (but it has no adjustment, its amplitude directly makes the PM modulation, so use a well adjustable external generator or leave open for no PM).
2. Use the virtual switchboard to link the 15 harmonics to one of the **envelope inputs 1 – 4** to determine their internal envelope trigger delay.
3. Turn **gate on**, now the Eurorack note gate signal triggers the internal envelope generators. If it is not connected to the bus, trigger through **modulation 1**.
4. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.

- use **adjust 1** to control the envelope shape (see Figure 4)
- use **adjust 2** and **slider 16** to control the envelope duration and relative delays between the 4 possible trigger events.

Mode 14, Frequency modulated harmonic generator.

1. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the desired base note harmonic sliders. Only 6 harmonics are used in this mode, i.e. sliders 1 to 6 are usable for the base signal spectrum setting. Remember that on power up, all are connected to **envelope input 1**, and only if you toggle the switch to **envelope select** position and back again, the previously stored connections are activated.
2. For modulating signal harmonic amplitude control, connect envelope generator outputs to those **envelope inputs** that you have linked for the sliders 7 and 8. Adjust sliders 7 – 8 to set the relative amplitudes for the modulator harmonics.
3. Connect envelope generator outputs to all those of the **envelope inputs** that you have activated for the individual harmonic frequency modulating signal amplitude defining sliders 9 - 14. Check the correspondence of sliders 1 – 6 and 9 – 14.
4. For modulating signal frequency control, connect an envelope generator output to **modulation 1** and adjust the modulation offset frequency to a desired level using **adjust 1,** and gain using **adjust 2.**
5. Define downward or upward modulation by setting slider 16 down or up, correspondingly.
6. Use slider 15 to adjust antialiasing lowpass filter cutoff.
7. Connect global modulation amplitude control envelope to **modulation 2**.
8. Optionally, connect the **CV pitch** and **CV velocity** inputs to the corresponding MIDI input module outputs.