19 BV512 Vocoder

Introduction

The BV512 is an advanced vocoder device with a variable number of filter bands. It also has a unique 1024-point FFT vocoding mode (equivalent of 512-band vocoding) for very precise and high quality vocoded speech. By connecting the BV512 to two instrument devices, you can produce anything from vocoded speech, singing or drums to weird special effects.

Even if you have worked with a vocoder before, please read the following section. Knowing the basic terms and processes will make it much easier to get started with the BV512!

How does a vocoder work?

Carrier and modulator

A vocoder accepts two different input signals, a “carrier” and a “modulator”. It analyzes the modulator signal, applies its frequency characteristics to the carrier signal and outputs the resulting “modulated” carrier signal.

In the most typical case, the carrier signal is a string or pad sound and the modulator signal is speech or vocals - the result will be a talking or singing synth sound. The modulator could also be drums or percussion (for rhythmically modulated sounds and effects) or any sound with changing frequency content.

Filter bands

Technically, a vocoder works in the following way: The modulator signal is divided into a number of frequency bands by means of bandpass filters (called the “modulator filters” or “analyzing filters”). The signal in each of these bands is sent to a separate envelope follower (which continuously analyzes the level of the signal). The carrier signal is sent through the same number of bandpass filters (the “carrier filters”), with the same frequency ranges as the filters for the modulator signal. The gain of each bandpass filter is controlled by the level from the corresponding envelope follower, and the filtered signals are combined and sent to the vocoder’s output.

In this way, the carrier is filtered to have roughly the same frequency characteristics as the modulator. If the modulator signal has a lot of energy in one of the frequency bands, the gain of the corresponding filter band for the carrier signal will be high as well, emphasizing those frequencies in the output signal. If there is no signal at all within a frequency band in the modulator signal, the corresponding band in the output signal will be silent (as the gain will be zero for that filter).

There are several factors determining the quality of the vocoder sound, but the most important is the number of filter bands. The larger the number of filter bands, the closer will the output signal follow the modulator’s frequency characteristics. The BV512 offers 4, 8, 16 or 32-band vocoding.

✪ **Even if a high number of bands will make the sound more precise and intelligible, this isn’t always what’s desired! Vocoding with a lower number of bands can give results that sound different, fit better in a musical context, etc.**

FFT vocoding

The BV512 has an additional FFT mode, in which the vocoding process isn’t based on bandpass filters as described above. Instead, FFT (Fast Fourier Transform) analysis and processing is used. This equals 512 “conventional” frequency bands and results in a very precise and detailed vocoder sound. Note:

- The FFT mode is best suited for vocoding speech or vocals, giving crystal clear and highly intelligible results. It is not so well suited for vocoding drums and percussion, since the FFT process is inherently “slower” than the regular filtering and doesn’t respond as quickly to transients, and also there will be a slight delay added to the signal (in the region of 20ms). A workaround solution to this would be to move the modulator signal slightly ahead to compensate for the delay.
- Where the conventional filter bands are distributed logarithmically (i.e. the same number of filter bands per octave), the 512 bands in the FFT mode are distributed linearly. This means a lot of the bands will be in the high frequency range - this is one of the reasons for the clear sound but it is also something to keep in mind when making settings for the vocoder in FFT mode.

Setting up for basic vocoding

This tutorial describes how to connect and use a typical vocoder setup. We assume here that you have a MIDI keyboard connected. For details on the parameters, see page 214.

1. **Make sure there's a MIxer device in the rack (with at least one free channel).**

2. **Create the instrument device you want to use for the carrier signal.**
   This could typically be a synth or a sampler. In this example we choose a Subtractor synthesizer.

3. **Set up the carrier device for a sustaining, bright sound.**
   It's important to have high frequencies in the carrier. On the Subtractor, a simple but effective carrier sound would be based on a sawtooth wave, with the filter fairly open. For more about choosing carrier sounds, see page 216.

4. **Select the carrier device and create a BV512 Vocoder.**
   If you flip the rack around you will see that the Vocoder is automatically routed as an insert effect for the carrier device (using the Carrier Input jacks).

5. **Press [Shift] and create the instrument device you want to use for the modulator signal.**
   Pressing [Shift] will add the device without auto-routing it to a mixer - this makes sense since we want to route it to the Vocoder in this case.
   For a modulator device you would typically either want a sampler (with vocals or speech samples), a drum machine or a Dr.Rex device (with vocal or rhythmic loops). For simplicity we use a Dr.Rex device in this example.

Carrier

Vocoder

Modulator

6. **Flip the rack around and route the output of the Dr.Rex to the Modulator Input jack on the BV512.**

7. **On the BV512 Vocoder, turn the Dry/Wet knob fully to the left ("dry").**

   This will let you hear the unprocessed sound of the modulator device only - useful for the next step:

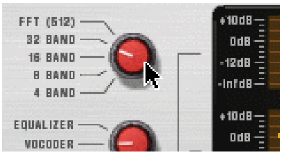
8. **Load a loop into the Dr.Rex device and click the Preview button to start playback.**
   For example, you could simply choose one of the Dr.Rex Drum Loops in the Factory Sound Bank.

9. **Turn the Dry/Wet knob on the vocoder fully to the right ("wet").**
   Now you won't hear anything - since there is no carrier signal.

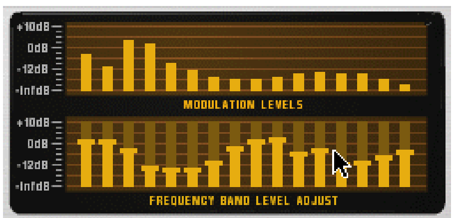
10. **Route MIDI to the carrier device by clicking in the MIDI symbol column for its track in the sequencer.**

11. **Play a chord or a note on your MIDI keyboard.**
    What you hear now is the vocoded sound, e.g. the carrier sound processed to have the same tonal characteristics as the modulator.

12. **Try the different filter band options and note the difference in sound.**



13. **You can also adjust the vocoder sound by clicking and dragging the bars in the lower display.**
    Each bar corresponds to a frequency band, with low frequencies to the left and high frequencies to the right. You adjust the level of a band by dragging its bar up or down. Clicking and dragging across the bars allow you to change the levels of several bars, much like drawing an eq curve.



The upper display shows the spectrum of the modulator signal, for display only.

→ **To reset a band to ±0 dB, press [Command] (Mac) or [Ctrl] (Win) and click on it.**
You can also reset all bands to zero by bringing up the context menu for the Vocoder device and selecting "Reset Band Levels".

14. **If the vocoder sound is "muddy" or indistinct, try raising the "HF Emph" knob on the Vocoder.**
    This parameter (High Frequency Emphasis) boosts the high frequencies in the carrier signal.

15. **Try out the other parameters if you like.**
    See page 214 for details.

That's it - a basic vocoder setup!

Vocoded vocals

The most common usage for a vocoder is probably the typical "singing" or "talking synth" sound, using vocals or speech as modulator. Since Reason doesn't support live audio input you cannot sing and play in real time - instead you need to use sampled speech or vocals (with e.g. an NN-19 or NN-XT as the modulator device). The procedure for this is roughly the same as in the tutorial above, but this time you need to record or enter some notes in the sequencer for the modulator device (since the samplers don't have pattern or Preview playback). Here's a quick guideline:

1. **Create the carrier device.**

2. **Select the carrier device and create a BV512 vocoder.**

3. **Select the BV512 and create the modulator device (typically an NN-19 or NN-XT sampler device).**

4. **Load the vocals or speech samples into the sampler device and assign them to keyzones as desired.**
   For details about using sampler devices, see the respective device chapter.

5. **Record or enter some notes on the sequencer track for the sampler device, so that the vocal samples are played back where you want them in the song.**
   To hear the unprocessed sound of the sampler device, set the Dry/Wet control on the BV512 to "Dry", as above. When you're done, turn the control back to "Wet" to get the vocoded sound.

6. **Route MIDI to the carrier device.**

7. **Start sequencer playback and play notes or chords on your MIDI keyboard.**
   The result will be the classic vocoded vocal sound.

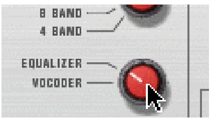
8. **At this point you may want to record the notes or chords you play for the carrier device.**
   As MIDI is already routed to the carrier device track, all you need to do is start recording and play along.

Using the BV512 as an equalizer

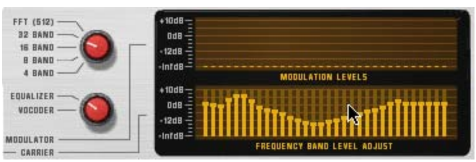
The BV512 has a unique equalizer mode, in which the device works purely as an insert effect (the modulator input isn't used). This allows you to use the processing filters of the vocoder as a kind of graphic equalizer.

Setting up

1. **Select the device that you want to process through the BV512.**
2. **Create a BV512 device.**
   It is automatically connected as an insert effect, using the Carrier Input jacks.
3. **Set the switch to the left of the displays to "Equalizer".**



In use



In equalizer mode, you cut or boost frequencies by clicking and dragging in the lower display - just as with a regular graphic equalizer. The usage and results differ depending on which mode is selected:

4 - 32 band mode

As in vocoder mode, the number of bars in the display conforms to the number of bands selected (4, 8, 16 or 32). With a higher number of bands you get a more detailed control over the frequency response. However:

➜ **In these modes, the equalizer will "color" the sound even if all bands are set to ±0 dB!**
This is due to phase interaction and overlap between the bandpass filters.

Therefore you probably want to use the 4 - 32 band mode for coloring and mutating sounds - not for subtle, "clean" equalizing.

FFT (512) mode

In FFT (512) mode you still get 32 bars in the display, but the each bar may control several frequency bands (remember that there are 512 bands in FFT mode). Since the frequency bands are distributed linearly in FFT mode, bars to the left in the display control few frequency bands while bars to the right control many frequency bands.

➜ **In FFT (512) mode, setting all bands to ±0 dB is the same as bypassing the equalizer - the sound will not be affected.**
This makes FFT mode suitable for "clean" equalizing, where you want to boost or cut some frequencies without changing the basic sound character.

➜ **However, FFT mode equalizing is not suited for very drastic frequency cuts or boosts, as this may give audio artefacts due to the workings of FFT processing.**
Still: as always, there are no hard and fast rules. Let your ears judge!

➜ **Keep in mind that FFT mode also introduces a slight delay to the signal.**

BV512 parameters

On the front panel of the BV512 Vocoder, you will find the following parameters and displays:

Parameter	Description
Bypass/On/Off switch	In Bypass mode, the carrier signal passes through the device unaffected and the modulator signal is disregarded. In On mode, the device outputs the vocoded or equalized signal. Off mode cuts the output, silencing the device.
Level meters	Show the signal level of the carrier and modulator signals, respectively.
Band switch	Selects the number of filter bands (4, 8, 16 or 32) or FFT (512) mode.
Equalizer/Vocoder switch	Determines whether the BV512 should work as a vocoder or an equalizer. In Equalizer mode, the Modulator input is disregarded (see page 213).
Modulation level display	The upper display shows the spectrum of the modulator signal.
Frequency band level adjust	The lower display allows you to adjust the level of each filter frequency band, by clicking and dragging the corresponding bar. In vocoder mode this affects the vocoded sound. In equalizer mode, this is where you cut or boost frequencies. To reset a band to ±0 dB, press [Command] (Mac) or [Ctrl] (Win) and click on its bar in the display. To reset all bands, select "Reset Band Levels" from the device context menu. Note: when FFT (512) mode is selected, each of the 32 bars in the display corresponds to several frequency bands, with bars to the right in the display controlling progressively more bands (due to the FFT bands being linearly distributed over the frequency range).
Hold button	Clicking this button "freezes" the current filter settings. While the button is lit, the modulator signal doesn't affect the sound - the carrier signal is filtered with the settings as they were the moment you activated Hold. Click the button again to turn off Hold. Hold is also automatically reset (turned off) when you stop sequencer playback - just like the pitch bend and modulation wheels on synth devices. This function can be controlled via CV or MIDI, for sample and hold-like effects. The Hold button is not available in Equalizer mode.

Parameter	Description
Attack	This is a global attack time control, affecting all envelope followers (see page 210). Normally you probably want this set to zero, to make the vocoder react as quick as possible. Raising the Attack time can be useful for "smearing" sounds, creating pads, etc. Not available in Equalizer mode.
Decay	Similarly, this controls the decay time for all envelope followers, i.e. how quick the filter band levels drop. Adjust this according to taste and context. Not available in Equalizer mode.
Shift	Shifts the carrier filters up or down in frequency, drastically changing the character of the vocoded (or equalized) sound. This parameter can be controlled via CV, for phaser-like sweeps and special effects.
HF Emph (High Frequency Emphasis)	Boosts the high frequencies in the carrier signal. This is sometimes desired to get a clearer vocoded sound. The reason is that a carrier signal should theoretically contain roughly equal energies in all frequency ranges for best results - in a typical synth sound the high frequencies are often weaker than the low frequencies. Raising the HF Emph control will rectify this. Not available in Equalizer mode.
Dry/Wet	Determines the balance between modulator sound (dry) and vocoded sound (wet). To get the pure vocoder sound, set this to wet (turned fully right). Not available in Equalizer mode.

Connections

The back panel of the BV512 offers the following connections:

Individual band levels

These are CV outputs and inputs.

- The upper row outputs CV signals generated by the envelope followers for each frequency band.
- The lower row are CV level inputs to the individual bandpass filters through which the signal is processed (the “vocoder filters”).
  Connecting a CV signal to one of the inputs breaks the internal signal path from the corresponding envelope follower (in other words, that frequency band is now controlled by the CV signal you’ve connected - not by the corresponding frequency band in the modulator signal).
- If 16 band mode is selected, each output/input pair corresponds to a separate frequency band. In 8 band or 4 band mode, only the 8 first or 4 first output/input pairs are used. In 32 band mode, each output is a mix of two adjacent frequency bands and each input controls two bands. Finally, in FFT (512) mode each output/input pair corresponds to several frequency bands.

There are several interesting uses for the Individual band levels connectors: you can cross-patch frequency bands so that e.g. low frequencies in the modulator signal controls high frequency bands in the vocoder, you can extract CV signals for controlling synth parameters in other devices, you can base the vocoding on CV signals from other devices rather than on a modulator signal, etc. See page 218 for details.

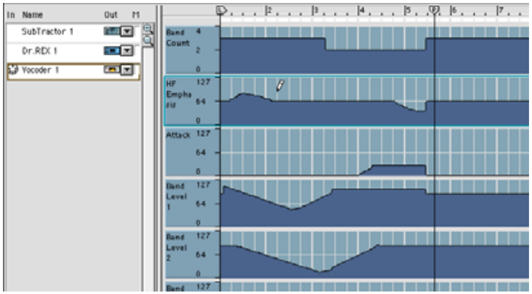
Other CV connections

Connection	Description
Shift (CV in)	This allows you to control the Shift parameter from an external CV source. A sensitivity knob determines how much the Shift setting is affected by the CV signal.
Hold (Gate in)	When a gate signal is sent to this input, the Hold function is activated (see page 214). Hold remains on until the gate signal “goes low” (falls to zero). By connecting e.g. a Matrix to this input, you can create “stepped” vocoder sounds, sample and hold-like effects, etc.

Audio connections

Connection	Description
Carrier input	This is where you connect the instrument device that provides the carrier signal (or the device to be processed in Equalizer mode) - typically a synth or sampler device. The vocoder can handle mono or stereo carrier signals.
Modulator input	This is where you connect the instrument device that provides the modulator signal, in mono. This connection is not used in Equalizer mode.
Output	In Vocoder mode, the outputs carry a mix between the vocoded signal and the modulator signal (as set with the Dry/Wet control on the front panel). In Equalizer mode the output is the carrier signal, processed through the equalizer filter. Note that the output will be in mono if the Carrier input is in mono, and vice versa - the BV512 does not process mono into stereo.

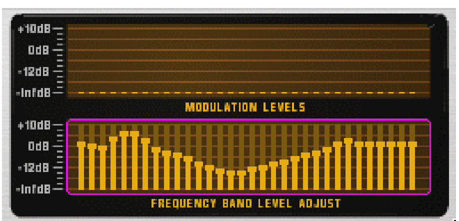
Automation



All parameters on the front panel can be automated in the standard manner. The individual band levels (the bars in the lower display) will be edited on separate lanes in the sequencer. Note:

→ **As with the other effect devices, you have to manually create a sequencer track for the BV512.**

→ **Although the band level adjustments can be edited individually, they are treated as one automatable parameter on the device panel.**
This means that if any single band level control is automated, there will be a frame around the whole lower display on the device panel. [Ctrl]-clicking (Mac) or right-clicking (Win) in the lower display and selecting "Clear Automation" will remove the automation for all bands. Similarly, selecting "Edit Automation" will open the sequencer with lanes for all band levels shown.



The frame indicates that one or more band level controls are automated.

Tips and tricks

Choosing a carrier sound

As always, which carrier sound to choose is a matter of taste and musical context. However, here are a few guidelines to help you get a good result:

- The carrier sound should preferably have a lot of harmonic content (brightness) - dark or muffled sounds will not "give the vocoder much to work with".
- Often, you want the carrier sound to sustain at an even level (i.e. it shouldn't "die out" when you hold a chord). Similarly, you most often want a reasonably fast attack (although not with a distinct, sharp click or edge).
- You may want a sound that is rather static over time, without drastic envelope control of filter cutoff for example.
- If you want to play vocoded chords, the carrier sound must of course be polyphonic.

Here are some hands-on suggestions for carrier sounds:

→ **A simple Subtractor pad based on a sawtooth waveform.**
You could simply start with the initial patch (as set up when you create a new Subtractor device). Open the filter, turn off envelope modulation of the cutoff frequency and raise the Amp Envelope Sustain.
If you want a classic, rich chorus-like sound, use two detuned oscillators - or better still, add a UN-16 Unison device as an insert effect between the Subtractor and the vocoder!

A simple but effective carrier sound setup.

→ **A similar fat carrier sound can be obtained using a Malström device with a patch based on the "Sawtooth*16" graintable.**
With the Malström you can get a stereo carrier signal with no extra devices: simply select the "Sawtooth*16" graintable for both oscillators, detune the oscillators slightly with the Cent controls and raise the Spread parameter to the desired stereo width. No filter routings are necessary.

→ **For a more distinct and precise sound, try using a narrow pulse waveform.**
You get this by selecting e.g. a sawtooth wave on the Subtractor, setting the Phase Mode selector to "–" and turning the Phase knob to the left until you get the desired sound. This type of carrier sound lends itself well to monophonic vocoder lines in the lower registers.

→ **Use noise as a carrier.**
Try using pure noise (possibly filtered down a bit) for robotic voices, whispering and special effects. It's also very useful to add a bit of noise to a sawtooth or pulse sound - this makes vocoded speech clearer and more intelligible.

→ **Use sampled strings or choir sounds.**
A rich drawbar organ sample can also be a cool carrier sound.

→ **For unusual vocoder sounds, try using the Malström as carrier device, e.g. with a glassy, digital pad sound selected.**
Try turning up the Attack and Decay controls on the BV512, for smeared, rhythmic or pseudo-random modulation of a pad.

Choosing a modulator sound

The modulator sound should typically have varying level and harmonic content. As we've already mentioned, the most typical modulator sounds are vocals or speech and drums or percussion.

→ **The quickest way to get a modulator sound is to use a rhythmic loop in the Dr.Rex device (as in the tutorial on page 211).**
This way you don't have to program a rhythm pattern. On the other hand, using a Redrum as modulator allows you to create exactly the rhythm you want and fine-tune the sounds and the groove.

→ **To use your "own" vocals as modulator sounds, you need to record them as WAV or AIFF files (using any audio recording utility on your computer) and load the files as samples into an NN-19 or NN-XT device.**

→ **Instead of using a sampler device as modulator for speech or vocals, you could slice the vocal samples in Propellerheads' ReCycle application, and play them back with a Dr.Rex device.**
This would make it easier to work with vocoded vocals, especially if you are experimenting with different tempo settings or grooves. Tip: You can copy the MIDI notes played by Dr.Rex to the carrier track so that the original rhythm of the speech/vocal is preserved.

Using the modulator as carrier

You can get cool special effects by using the same device both as carrier and as modulator. For example, try processing a Redrum device in the following way:

1. **Create a Redrum device and set up the desired patch and pattern.**
2. **Create a Spider Audio Merger & Splitter device.**
3. **Create a BV512 Vocoder.**
4. **Flip the rack around and connect the devices in the following way:**

The output of the Redrum goes into the splitter section of the Spider, and is split into two signals. One signal goes into the carrier input of the vocoder, the other goes into the modulator input.

This is essentially the required connections, but for best results it's a good idea to add some distortion and/or compression to the carrier signal - this increases the amount of high frequencies in the carrier signal:

5. **Press [Shift] and create a Scream 4 distortion device.**
6. **Connect the distortion device as an insert effect between the Spider and the carrier input of the vocoder.**
Now, the carrier signal will be processed in the distortion device, but not the modulator signal.
7. **Play back the pattern and experiment with the settings on the vocoder and distortion device.**

→ **This technique can also be used to process vocals and speech.**

→ **Try adjusting the Shift parameter for new effects and sounds.**
Remember that you can route CV to the Shift parameter on the back of the BV512 - use e.g. a Matrix or an LFO output on a synth device!

Controlling the Hold function

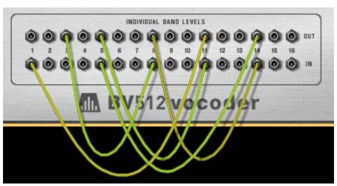
As described see page 214, pressing the Hold button on the front panel "freezes" the current filter spectrum until you deactivate it again. This can be used for creating sample & hold-like effects, stuttering or garbled vocoder sounds:

- Connect e.g. the Gate output on a Matrix device to the Hold input on the back of the BV512. By playing back a gate pattern on the Matrix, the Hold function will repeatedly be turned on and off according to the programmed rhythm in the pattern. Hold will be active for the length of each gate signal.
- Automate the Hold function with the main sequencer, either by recording it or by drawing in its controller lane.
- If you route MIDI to the BV512 you can control the Hold function in two ways by default: By pressing a damper pedal connected to your MIDI controller or by playing the note C4. In both cases, the Hold function will be momentary - Hold is on until you release the pedal or key.

Using the individual band level connections

As described on page 215, the individual band level connectors on the back are CV output and input jacks. The upper row sends out the CV signals from the envelope followers for the different frequency bands, while the lower jacks are CV inputs for controlling the individual bandpass filters (breaking the internal connection from the envelope followers). There are several interesting things you can do with these connections:

Crosspatching frequency bands



By connecting outputs to inputs in alternative configurations, you can drastically change the result of the vocoding. For example, you could have low frequencies in the modulator signal give high frequencies in the vocoded sound and vice versa. Note:

- In 4 band and 8 band mode, only the 4/8 first output/input pairs are used.
- In 32 band mode and FFT (512) mode, each connection corresponds to two or several frequency bands.
  This means that connecting an output to the input with the same number is *not* the same as using the internal signal path (no CV cable connected). You can hear this quite clearly in FFT (512) mode: connect all outputs to the corresponding inputs and gradually remove the CV cables while listening to the vocoder sound - the sound will progressively get more detailed.

Extracting CV from the vocoder

You can connect an individual band level output to any CV input on any device. This means you can use the vocoder as an envelope follower, having elements in the modulator sound control a parameter in another device, e.g. an effect. Note:

- The Attack and Decay settings on the BV512 panel affect the envelope followers, and thus the rise and fall times of the CV signals from the individual band level outputs.
- If you are using the vocoder in a mode with many bands, but want a broader frequency range to generate the CV signal, you can merge several band outputs into one CV signal - use a Spider CV Merger & Splitter device.

Controlling vocoder bands from an external source

Connecting a CV source to an individual band input breaks the internal connection from the corresponding envelope follower. This way you can "manually" control the vocoder filters. Some applications:

➜ **Connect the CV outputs for one or more envelopes in the carrier device to individual band inputs.**
When you play the carrier instrument, one or more of the bandpass filters in the vocoder will automatically open, adding an extra attack to the sound. Useful if you really want to "play" the carrier, rather than just hold a chord.

➜ **Connect the gate outputs on a Redrum device to individual band level inputs.**

With this connection (and no device connected to the Modulator input), the Redrum will serve as a pattern sequencer, opening and closing different filter bands. To adjust the gate times, set the drum sounds to Gate mode and use the Length parameter. The result is totally different from using the audio signal of the Redrum as modulator.

The vocoder bands are now solely controlled by the gate signals from the drum channels - the modulator input isn't used.
Note that you can use a Spider CV Merger & Splitter device to split a gate signal, sending it to several bands. Also, note that the velocity of the programmed drum notes govern the level of the corresponding filter bands.

"Playing" the vocoder from a MIDI keyboard

If you have routed MIDI to the BV512, playing notes from C1 and up will control individual filter bands. For example, in 16 band mode, C1 controls band 1, C#1 band 2 and so on up to D#2 (which controls band 16).

- The level of the bands is proportional to key velocity (how hard you play).
- A band will be "open" until you release the corresponding key.
- Bands to which you have connected a CV signal (using the individual band level inputs on the back panel) will not respond to MIDI keys.

Note that with this function, you "play the modulator". You still need a carrier signal to get any sound. Typically, you would first record the notes or chords for the carrier device in the sequencer, then route MIDI to the vocoder and "play" it from your MIDI keyboard while playing back the recorded carrier notes.

✪ **An interesting application of this is to patch the vocoder as an insert effect for the whole mix (the output of the main mixer connected to the carrier input, with no modulator device connected), and "play the vocoder". Only the frequency bands for which you press keys will be let through. Use the FFT (512) mode for best results.**

Using the BV512 as a reverb

This is a very special trick which can be quite cool. Proceed as follows:

1. **Create a Redrum device.**
   The "vocoder-reverb" is best suited for drums, even though nothing stops you from using it on other sounds.

2. **Create a Subtractor and a vocoder.**
   The Subtractor will automatically be routed to the carrier input. We don't need a dedicated modulator device in this setup.

3. **Flip the rack around and connect Aux send 1 on the Mixer to the modulator input on the vocoder.**

4. **While you're there, re-route the vocoder output to Aux return 1.**
   This way, our vocoder-reverb will be connected as a regular send effect.

5. **Set the vocoder to FFT (512) mode, turn the Decay knob to between 6 and 7 and turn the Dry/Wet control to "wet" (fully right)**

6. **On the Subtractor, set up a noise sound as follows:**
   Turn the Oscillator Mix knob fully to the right.
   Turn on the Noise section (but make sure Osc 2 is off).
   In the Noise section, turn Color to around twelve o'clock.
   Open the filter fully and make sure resonance is set to 0.
   Make sure Filter Envelope Amt is 0 (and turn off velocity modulation.
   Raise the Sustain to full in the Amp Envelope section.

Now we want the Subtractor to play a continuous noise. You could just route MIDI to it, play a note and keep it pressed, but that will probably wear you out in the long run. Better to use a Matrix:

7. **Create a Matrix and route it to the Subtractor.**
   We really only need the Gate connection - the note number isn't important with the noise patch.

8. **Set up a one step pattern with a tied gate (press [Shift] and draw the gate) and start playback on the Matrix.**
   Now the vocoder gets a continuous noise signal as carrier.

9. **Create a suitable drum pattern on the Redrum and start pattern playback.**

10. **Gradually turn up send 1 for the Redrum channel in the mixer.**
    This now serves as a balance control between the dry drum sound and the reverb, generated by the voccoded noise! Set it to a suitable reverb level.

11. **Use the Decay control on the vocoder to adjust the reverb decay time.**

12. **Use the Noise Color control on the Subtractor to make the reverb darker or brighter.**
    You could use the filter cutoff for this as well.

That's it - a pretty good reverb sound with a lot of control. Although the settings above give the most natural sound, you can vary the sound and create special-FX reverb in the following ways for example:

- Switch the vocoder to a lower band mode.
- Lower the cutoff and add some resonance in the Subtractor filter.
- Modulate the Subtractor filter with a fast LFO.
- Set the Subtractor filter to HighPass mode to remove the bottom end from the reverb.
- Turn off the Matrix controlling the Subtractor and "play" the noise patch yourself (or from the sequencer). This way you can create gated reverb effects, etc.

Creating a stereo reverb

What you've got above is a mono reverb. Here's how to make it stereo:

1. **Select the Subtractor and create a Spider Audio Merger & Splitter device.**
2. **Create a DDL-1 delay.**
3. **Connect the devices in the following way:**
   The Subtractor output should be routed to a Splitter input on the Spider. One split output should be routed to one of the carrier inputs on the vocoder, the other split output should be routed to the delay. The delay output (mono) should be routed to the other carrier input on the vocoder.

The vocoder will now get a "fake-stereo" noise carrier signal.

4. **Make sure the output from the vocoder is connected in stereo to the Aux return on the Mixer.**
5. **Finally turn down the Feedback on the delay, set it to all "wet" and set the decay time to a second or so.**

When you now start playback on the Redrum, the reverb will be in stereo!

RV7000 Advanced Reverb

The RV7000 is a high quality reverb processor. It features nine different reverb and echo algorithms, ranging from rooms and halls to special effects. Since the RV7000 comes with a number of useful reverb presets, you could simply select one and tweak the most important parameters on the main panel - or you could use the Remote Programmer panel to fine-tune the reverb in great detail.

The RV7000 also contains an equalizer and a gate section. Both of these are for processing the actual reverb sound, making it possible to get virtually any kind of reverb character, including gated reverb.

About the Patch format

Like the Scream 4 device, the RV7000 features programmable effect presets. In the Factory Sound Bank will find a number of preset Patches which can be used as they are or provide you with a good starting point for further tweaking.

Patches use the Windows file extension "*.RV7". Loading and saving Patches is done in the same way as for instrument devices.

Connections

Typically you connect the RV7000 as a send effect, as this allows you to use it for processing several different mixer channels. However, it's also possible to use it as an insert effect - use the Dry/Wet control on the main panel to adjust the balance between the dry, unprocessed sound and the reverb. Note:

➜ **The RV7000 is a true stereo reverb, which means that it will use the stereo input information when processing both channels (without summing the input channels).**
It's also possible to use it as a mono in - stereo out effect. Which type of connection to use (mono or stereo in) depends on the material. If the audio sources are in mono (or in stereo but with no important difference between the left and right channel) using a mono input is sufficient.

➜ **If you want to use RV7000's Reverse reverb effect, you should consider connecting it as an insert effect or using Send 4 on the Mixer, with Pre-fader mode selected (and the channel fader lowered).**
This is because you typically don't want to hear the dry sound when using the Reverse effect. See page 237.

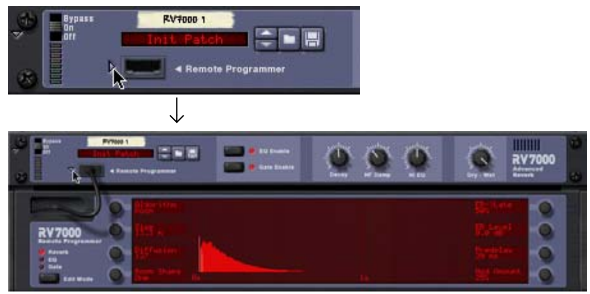
The main panel

The RV7000 main panel.

When you create an RV7000, only the main panel will be shown. This contains a section for handling patches, on/off buttons for the EQ and Gate sections, the most important reverb parameters and a dry/wet mix control. To select a reverb patch and make coarse adjustments, this is all you need.

The remote programmer

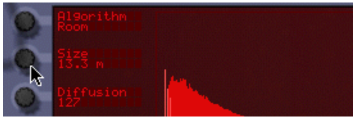
Clicking the arrow button next to the "cable slot" on the main panel brings up the remote programmer panel.



This is where you make detailed settings for the reverb. Note:

- The Edit Mode button to the left determines which section to make settings for, Reverb, EQ or Gate.

- Settings are made with the eight dials around the graphic display. The functions of the dials differ depending on the selected Edit Mode and the selected reverb algorithm. Next to each dial, the display shows the name and value of the corresponding parameter.



- Not all modes and algorithms use all eight dials. If a dial isn't used in the selected mode, nothing will be shown next to it in the display.
- You cannot make settings in the graphic display itself - this is for showing a graphic representation of the selected reverb.

Reverb algorithms and parameters

About the main panel parameters

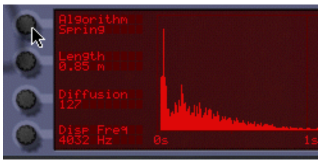
On the main panel you find three reverb parameters that are available for all algorithms:

Parameter	Description
Decay	This governs the length of the reverb or the feedback if an echo algorithm is selected.
HF Damp	Controls how quickly the high frequencies should decay in the reverb. Raise it to gradually remove high frequencies, making the reverb sound warmer and less bright.
HI EQ	This is a high-shelving EQ that works much like a typical treble control on a mixer or amplifier. Lower the setting for a softer reverb sound or raise it to get more high frequencies.

Selecting an algorithm

You select a reverb algorithm in the remote programmer panel:

1. **Click the remote programmer arrow button on the main panel to display the remote programmer panel.**
2. **Make sure the Edit Mode button is set to Reverb.**
3. **Use the top left dial to select a reverb algorithm.**
   The selected algorithm is shown in the display next to the dial.



Here's a quick overview of the nine algorithms - for details and parameter descriptions, see below.

Algorithm	Description
Small Space	Emulates a small enclosed space (a small room or a resonant body).
Room	Emulates a room with adjustable shape and wall character.
Hall	Emulates a hall.
Arena	Emulates a large arena, with separate pre-delay for the left, right and center reverbs.
Plate	Emulates a classic plate reverb.
Spring	Emulates a spring reverb, as used in e.g. guitar amplifiers.
Echo	An echo effect with gradually diffusing echo repeats. Can be synced to Reason's tempo.
Multi Tap	A multi-tap delay with four different delay lines and tempo sync.
Reverse	A reverse reverb effect that "pushes" the dry sound to appear after the reverb. The result is a backwards reverb leading up to the direct sound.

Small Space

This algorithm places the sound in a small enclosed space, ranging from a tiny resonant body to a room. The parameters are:

Parameter	Description
Size	The size of the emulated space.
Mod Rate	The reverb can be randomly modulated for a more even sound (or for special effects). This parameter sets the rate of modulation (the amount is set with Mod Amount).
Room Shape	Select from four different room shapes, affecting the character of the reverb.
LF Damp	Controls how quickly the low frequencies should decay in the reverb. Raise it to gradually remove low frequencies, making the reverb sound "thinner" and less boomy.
Wall Irreg	Adjusts the positioning of the emulated walls in the small space. The lowest setting emulates two directly opposed walls while higher settings emulate more walls and angles, for a more complex resonance.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the reverb.
Mod Amount	Sets how much the reverb will be modulated. Use fairly low settings when emulating real rooms and resonant bodies, and higher settings for special effects.

Room

Emulates a medium-sized room, with the following parameters:

Parameter	Description
Size	The size of the emulated room.
Diffusion	At low Diffusion settings, you will hear the individual reverb “bounces” more clearly, while higher settings produce a more “smeared”, dense and even reverb.
Room Shape	Select from four different room shapes, affecting the character of the reverb.
ER->Late	The first “answers” in the reverb are called early reflections (ER) and are typically more pronounced than the actual reverb tail. This parameter sets the time between the early reflections and the reverb tail. This is set as a percentage - the actual delay time depends on the Size setting.
ER Level	Adjusts the level of the early reflections. “0” is normal level.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the early reflections and reverb.
Mod Amount	Sets how much the reverb will be modulated. Moderate modulation gives a natural, less static sound.

Hall

Emulates a hall.The parameters are the same as for the Room algorithm above (but the Hall algorithm offers larger Size settings).

Arena

Emulates the ambience in an arena or concert hall, with long pre-delay times (separate for left, right and center):

Parameter	Description
Size	The size of the emulated arena or hall.
Diffusion	At low Diffusion settings, you will hear the individual reverb “bounces” more clearly, while higher settings produce a more “smeared”, dense and even reverb.
Left Delay	The predelay time for the left side of the reverb.
Right Delay	The predelay time for the right side of the reverb.
Stereo Level	Adjusts the level of the left and right sides of the reverb. “0” is normal level.
Mono Delay	The predelay time for the mono (center) reverb signal.
Mono Level	Adjusts the level of the mono (center) reverb signal. “0” is normal level.

Plate

A classic plate reverb, excellent for vocals for example. The parameters are:

Parameter	Description
LF Damp	Controls how quickly the low frequencies should decay in the reverb. Raise it to gradually remove low frequencies, making the reverb sound “thinner” and less boomy.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the reverb.

Spring

An emulation of a spring reverb as can be found in guitar amplifiers, organs, etc. The spring reverb has the following parameters:

Parameter	Description
Length	Sets the length of the simulated spring.
Diffusion	At low Diffusion settings, you will hear the individual reverb "bounces" more clearly, while higher settings produce a more "smeared", dense and even reverb.
Disp Freq	When sending a signal to a real-life spring reverb, the initial transient will produce a quick, characteristic sweeping tonal noise. This is because different frequencies in the sound are delayed by different amounts (a phenomenon called dispersion). This parameter controls the frequency of that sound.
LF Damp	Controls how quickly the low frequencies should decay in the reverb. Raise it to gradually remove low frequencies, making the reverb sound "thinner" and less boomy.
Stereo (on/off)	Determines whether the output of the spring reverb should be in mono or stereo.
Predelay	Sets the predelay time, i.e. the delay between the source signal and the start of the early reflections and reverb.
Disp Amount	Sets the amount of dispersion effect (see Disp Freq above).

Echo

This is an advanced echo effect, with diffusion controls and tempo sync. When Echo is selected, the Decay control on the main panel controls the echo feedback (the number of echo repeats). The parameters are:

Parameter	Description
Echo Time	Sets the time between each echo. When Tempo Sync (see below) is off, the echo time is set in milliseconds (10 - 2000 ms); when Tempo Sync is on you set the echo time as a number of 1/16 notes or 1/8 triplet notes, in relation to the current song tempo.
Diffusion	When this is set to 0, the echo will sound as a standard delay with clear, precise repeats. Raising the Diffusion setting will introduce additional echoes very close to the "main" echo repeats, causing a "smeared" echo sound. This will also expand the echo stereo image.
Tempo Sync	Determines whether the echo time should be freely set ("off") or synchronized to Reason's tempo ("on").
LF Damp	Controls how quickly the low frequencies should decay in the echoes. Raise it to gradually remove low frequencies.
Spread	Adjusts the spacing of the additional echoes added by the Diffusion parameter. For a very smeared echo (sound more like a reverb), set both Diffusion and Spread to their maximum values.
Predelay	Sets an additional delay time before the first echo repeat.

Multi Tap

The Multi Tap delay produces up to four different delays with separate delay times, panning and level. The whole set of four delay taps can then be repeated at a given rate. Again, the Decay control on the main panel controls the feedback (the number of repeats for the whole multi tap set). All delay times can be tempo synced.

Note: this algorithm is handled a bit differently since you make separate settings for each delay tap:

- The parameters to the left of the display are common for all taps.
- You use the Edit Select parameter in the top right corner to select which tap to make settings for - the three parameters below affect the currently selected tap.

Tap 2 selected for editing.

- You can also set Edit Select to "Repeat Tap" - this is where you specify the repeat time for the whole multi tap "package".
  With short Repeat times, the first tap may be repeated before the last tap has sounded. This can be used to create very complex multiple delay effects.

The common parameters (to the left) are:

Parameter	Description
Tempo Sync	Determines whether the delay times and repeat times should be freely set ("off") or synchronized to Reason's tempo ("on").
Diffusion	Raising the Diffusion setting will introduce additional echoes very close to the "main" repeats, causing a "smeared" delay sound.
LF Damp	Controls how quickly the low frequencies should decay in the echoes. Raise it to gradually remove low frequencies.

When Tap 1 - 4 is selected with the Edit Select parameter, you can make the following settings for the selected delay tap:

Parameter	Description
Tap delay	Sets the delay - the time from the source signal to the tap. When Tempo Sync is off, the delay time is set in milliseconds (10 - 2000 ms); when Tempo Sync is on you set the delay as a number of 1/16 notes or 1/8 triplet notes, in relation to the current song tempo.
Tap level	Adjusts the level of the selected tap.
Tap pan	Adjusts the pan of the selected tap.

When Repeat Tap is selected with the Edit Select parameter, there is only one parameter to the right in the display:

Parameter	Description
Repeat Time	Sets the time between each repeat of the whole multi tap set. The number of repeats is set with the Decay control on the main panel. When Tempo Sync is off, the repeat time is set in milliseconds (10 - 2000 ms); when Tempo Sync is on you set the repeat time as a number of 1/16 notes or 1/8 triplet notes, in relation to the current song tempo.

Reverse

The Reverse reverb algorithm in RV7000 is special in that it actually "moves" the source audio as well. Sounds fed into the Reverse reverb are "sampled", a reverse reverb is created and played back and finally the "sampled" original sound is played back. For example, if you feed a snare drum hit into the Reverse reverb, you will hear a rising "backwards" reverb, followed by the snare drum hit.

Therefore, you probably don't want to hear the first, original (dry) sound. There are two ways to set this up:

➜ **Connect the RV7000 as an insert effect and make sure the Dry/Wet control on the main panel is set fully to "Wet".**

➜ **Connect the RV7000 as a send effect using send 4 on the Mixer, activate the Prefader (P) switch for the send and lower the mixer fader completely for the source signal.**
That way, the signal will be sent to the reverb but the dry sound from the Mixer channel isn't heard. Again, the Dry/Wet control on the reverb should be set to "Wet".

Note:

With this algorithm, raising the Decay setting on the main panel will make the reverse reverb start earlier and build up under a longer time. Similarly, the HF Damp parameter affects how fast the high frequencies are built up in the reverse reverb. In the remote panel, the Reverse algorithm has the following parameters:

Parameter	Description
Length	This sets the time from when the source signal is fed into the reverb until it is played back again. It is during this time you will hear the reverse reverb, as shown in the display. The time can be set in milliseconds or as note values, depending on whether Tempo Sync is off or on. Note: As stated above, the Decay setting determines the length of the actual reverse reverb - in essence how soon it starts after the source signal. But of course, the reverse reverb cannot start *before* the original source signal! If you set Decay to a longer time than the Length setting, the reverse reverb will start abruptly, immediately when the source signal is fed into the reverb. If this sounds complicated, just take a look at the RV7000 display and try the settings - you will soon see how it works. Note also that very high Length settings demand a lot of processor power. This can be reduced by adjusting the Density parameter, see below.
Density	Density governs the "thickness" of the Reverse effect. If this parameter is turned down to zero, the effect produces individual delays rather than a dense "wash", which can be used as a special effect. Worth noting is that if Density is set to around 50%, this can considerably reduce the CPU load without altering the sound of the effect too much. Exactly how much the Density parameter can be reduced without altering the sound depends on the source material.
Rev Dry/Wet	Sets the balance between the "moved" source signal ("dry", low values) and the reverse reverb ("wet", high values).
Tempo Sync	Determines whether the Length setting should be freely set ("off") or synchronized to Reason's tempo ("on").

The EQ section

The equalizer in RV7000 affects the wet reverb sound only and is used for shaping the character of the reverb. There are two EQ bands, one for low frequencies (shelving) and one full-range parametric EQ.

→ **To activate the EQ, click the EQ Enable button on the main panel so that the indicator lights up.**

→ **To make EQ settings, select "EQ" with the Edit Mode button to the left in the remote programmer panel.**

→ **In this mode, the remote programmer display shows a frequency curve, indicating the settings you make with the EQ parameters.**

The parameters are:

Parameter	Description
Low Gain	The amount of cut or boost of the low-shelving filter.
Low Freq	The frequency below which the Low Gain cut or boost is applied.
Param Gain	The amount of cut or boost for the parametric EQ.
Param Freq	The center frequency of the parametric EQ, e.g. at which frequency the level should be decreased or increased.
Param Q	This governs the width of the affected area around the set center frequency. The higher the value, the narrower the affected frequency range.

→ **Remember that you have a third EQ band at your disposal - the HI EQ parameter on the main panel.**
The reason why this is on the main panel and not in the EQ section is simply that it's a setting you may want to adjust often, without having to open the remote programmer panel.

The Gate section

The Gate section allows you to create gated reverb effects with a lot of options and possibilities. You can either trigger the gate from the source audio signal or via MIDI or CV.

When triggering the gate from the source audio signal, it works like this:

- The gate "listens" to the source (dry) signal and opens whenever the signal reaches a certain threshold level.
- The reverb sound is sent through the gate - when the gate is closed you won't hear the reverb.
- When the source signal level drops below the threshold level, the gate closes after a time that depends on the Hold parameter and the level of the source signal (see the parameter table).

→ **If you need the gate to be open for an exact duration (time), you should trigger it via MIDI or CV.**
In audio trigger mode, the actual gate time will vary depending on the source signal.

When triggering the gate via MIDI or CV, it works like this:

- The reverb sound is sent through the gate - when the gate is closed you won't hear the reverb.
- Whenever the gate receives any MIDI note (sent to the RV7000) or a gate signal (connected to the Gate Trig CV input on the back of the RV7000), the gate opens for the duration of the note or gate signal.

Note:

→ **To activate the Gate, click the Gate Enable button on the main panel so that the indicator lights up.**

→ **To make Gate settings, select "Gate" with the Edit Mode button to the left in the remote programmer panel.**

→ **In this mode, the remote programmer display shows two meters - one showing the signal level (with an indication of the threshold level) and one showing the status of the gate.**
This is useful for checking what happens, how the gate triggers, etc.

The parameters for the Gate section are:

Parameter	Description
Threshold	When Trig Source is set to “Audio”, this determines the audio signal level at which the gate opens. If you raise this setting, only very loud sounds will open the gate.
Decay Mod	This modulates the reverb Decay parameter so that the decay time is lowered when the gate closes. When this is set to zero, no decay modulation happens - this means that if the gate is closed and then opened again, you may hear “previous” reverb tails that are still ringing. If you raise the Decay Mod setting, the decay will automatically be lowered when the gate is closed, eliminating this effect.
Trig Source	Determines whether the gate should be triggered by audio or MIDI/CV, as described above.
High Pass	A high-pass filter that affects the audio that triggers the gate (only active when Trig Source is set to “Audio”). If you raise this setting, sounds with low frequencies only will not open the gate. Note that this setting doesn’t affect the sound of the reverb, only the triggering mechanism.
Attack	Determines how long it takes for the gate to open after a triggering signal has been received.
Hold	This parameter is only active when Trig Source is set to “Audio”. Hold affects how quickly the gate closes, in the following way: Internally, the gate is controlled by an envelope follower that analyzes the source signal level and generates a “level CV signal” accordingly. This signal is compared to the Threshold level to determine whether the gate should be opened or closed. The Hold parameter affects how quickly the envelope follower responds when the source signal level drops - you could say that this is the decay control for the envelope follower. The higher the Hold setting, the longer it will take for the envelope follower signal to drop below the threshold level and close the gate. But the resulting time also depends on the source signal level - with a loud signal, it will take longer time for the envelope follower to drop to the threshold level. Therefore, the actual gate time depends both on the Hold setting and on the character of the source audio.
Release	Determines how long it takes for the gate to close after the Hold time.

CV Inputs

On the back of the RV7000 you find three CV inputs. These are:

Parameter	Description
Decay	Controls the reverb decay or echo/delay feedback via CV.
HF Damp	Controls the HF Damp parameter on the main panel.
Gate Trig	Used for triggering the Gate section with a gate signal. The length of the gate signal determines the length of the gated reverb.