# ANTIDOTE


User's Manual

SYNAPSE
AUDIO SOFTWARE

# Contents

# 1 Introduction and Overview

Thank you for choosing Synapse Audio Antidote!

Antidote RE is a virtual-analog synthesizer for Propellerhead Reason. Antidote combines high audio quality, flexibility and a fast work flow in an easy-to-use plugin. It was specifically developed to complement Reason, and to yield the best possible sound, integration and user experience.

Antidote RE comes feature packed with two stereo oscillator banks, each emitting a stack of up to 24 high quality virtual-analog oscillator waveforms.

The oscillators pass through the filter stage, which employs the latest zero-delay feedback designs. This recent technology is quickly gaining a lot of popularity, as it is able to mimic the behavior of analog filters much better than previous designs.

Envelopes and LFOs further shape the sound, and the modulation matrix — which can modulate almost any sound parameter — adds the necessary depth to realize complex sounds. The output of the synthesis stage finally passes through a massive effect chain comprising EQ, Distortion, Phaser, Chorus, Delay, Reverb and a Compressor effect. All effects can be enabled and used simultaneously.

The back panel allows to connect Antidote RE to other devices, to add extra modulation via CV Inputs, to feed the effect chain with the outputs from other instruments and to connect Reason's step sequencers.

In contrast to conventional plugin standards, there is little difference between using Antidote RE as an instrument and as an effect. Both is possible and intended, and the vast modulation options apply equally well in both cases.

# 2 Basic Operation

## 2.1 Patch Controls

The patch operation in Antidote RE is the same as in any other Reason device. To select a patch, either click on the patch name, the folder icon or the arrow buttons. To save a patch, click on the disk icon.

## 2.2 Polyphony

Each key that you press triggers a voice, with a pitch determined by the key number. As each voice requires CPU time, the total number of available voices is limited. The maximum number of available voices can be adjusted by changing the POLY parameter in the top left. Note that two special modes exist, **Mono** and **Mono-Legato**. In both modes, only a single voice is audible at a time. When multiple keys are pressed in succession and sustained, the one hit last is audible. Release it to snap back to the previous key (make sure this one is still pressed!). This permits a unique style of playing, which is particularly useful in combination with the Glide knob. You can smoothly glide from one note to another in this way. The difference between Mono and Mono-Legato is that the Mono mode retriggers all envelopes whenever you press a key, while the Legato mode will not change the envelope states during a slide from one note to another.

## 2.3 Pitch Wheel

The pitch bend wheel changes the pitch of the current sound. The minimum and maximum settings can be changed on the backside of the device, in semitones. There is two separate controls, one for up and one for downwards motion. By default, the pitch bend ranges from -2 (down) to +2 (up) semitones.

## 2.4 Modulation Wheel

The modulation wheel changes one or more sound parameters in real time, and adds expressiveness to your performance. The parameter to change can be specified in the modulation matrix, by selecting Mod Wheel in the source column, then choosing an arbitrary sound parameter to modify from the destination list. Move the corresponding AMT knob to the right, to specify how strong the modulation should be at most.

A common application of the modulation wheel is to open the filter with it, or to add vibrato/tremolo type effects. To do this, use LFO1*MW or LFO2*MW in the modulation matrix source column, which multiplies the current LFO state with the modulation wheel value (see chapter 4 for a more detailed description of the modulation matrix).

## 2.5 Aftertouch

Aftertouch is another common way to add expressiveness to a sound. Aftertouch measures the pressure applied to all keyboard keys as a whole. When holding down a chord, for instance, then increasing pressure, you can add a vibrato effect to the sound. The aftertouch programming is identical to how you program the modulation wheel. Using the previous example, choose LFO1*AT or LFO2*AT as a source, then any arbitrary sound parameter from the destination list, to get a vibrato type effect.

Note that aftertouch is available in many, but not all MIDI keyboards. Consult your MIDI keyboard manual to find out if your keyboard supports aftertouch.

## 2.6 Using Antidote as an effect device

Antidote RE can be used as a killer effect unit and was designed with this application in mind. Use the Init Patch from the root folder as a starting point for effect work. This patch sets all parameters to their default values, including all effect parameters. Connect your audio input signal to Antidote by using the EXT IN jacks on the back side. If you want to adjust or automate the input level, use the red EXT IN knob on the front. This avoids having to flip between the front- and back side of the device just to do level corrections.

# 3 Sound Parameters

This section describes how an Antidote patch is constructed, as well as the operation of all front panel knobs and switches, except the arpeggiator and modulation matrix (which are covered in chapters 4+5).

## 3.1 Patch Structure

The structure of an Antidote patch is shown in fig. 3.1. The block diagram shows the basic working principle of the entire synthesizer.

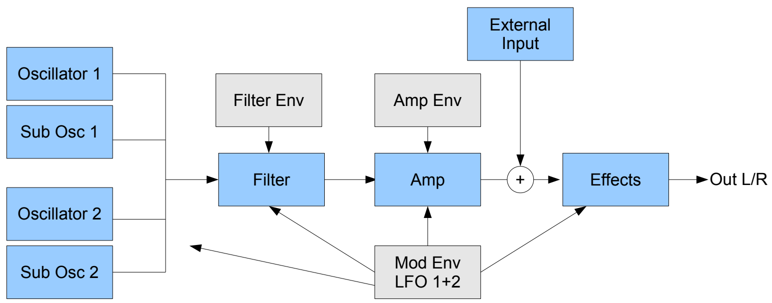



Figure 3.1: Patch Structure.

Whenever a MIDI note is played, a voice is triggered to synthesize that note. Each voice comprises three major building blocks, the oscillators, the filter, and the amplifier. The blocks emulate the three basic properties of a sound: Pitch, Timbre and Volume. The oscillator block controls the pitch and basic timbre of a sound by generating one or more periodic waveforms. The resulting signal is typically very bright. To further refine the timbre, the signal is processed by the filter block, which attenuates frequencies specified by the user; usually, high frequencies are removed. Hence, this type of synthesis is commonly called "subtractive". The third block controls the volume of the sound.

On their own, the three basic building blocks synthesize a completely static sound. This is in contrast to acoustic sounds, where pitch, timbre and volume change over time. In order to obtain this possibility in a synthesizer, envelopes and low-frequency oscillators are used to add dynamic variation to a sound. The most important envelope is the amplitude envelope ("Amp Env"), which is essential to fade in and fade out notes and thus to make a synthesizer playable like a real instrument in the first place. Also important is the filter envelope ("Filter Env"), which dynamically controls the brightness and thus the timbre of a sound over time. The modulation envelope ("Mod Env") can be freely assigned to any sound parameter.

## 3.2 Oscillators

An oscillator generates a periodic waveform and forms the basic building block of the majority of synthesizers (the most common waveforms are illustrated in fig. 3.2). Antidote offers two banks of oscillators, plus two sub oscillators. Both oscillator banks allow you to instantiate between 1 and 24 oscillators with the same waveform shape, but with a different tuning each. The sub oscillators are sawtooth and pulse waveforms, and play one octave below oscillator bank 1 and 2, respectively.

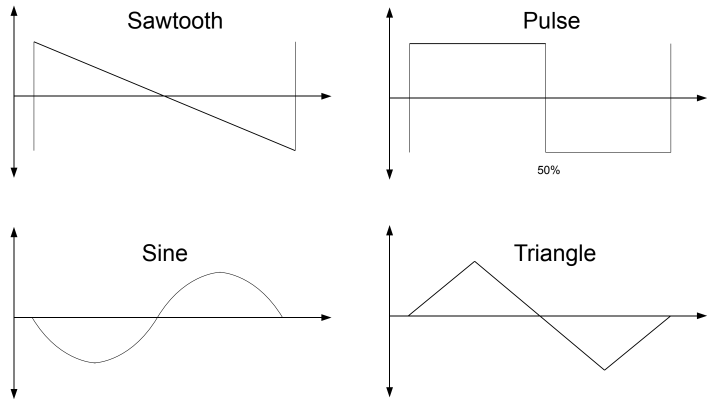



Figure 3.2: Basic oscillator waveforms.

### 3.2.1 Oscillator 1+2

#### WAVEFORM and MODIFIER

Each oscillator bank allows you to choose between six different oscillator types. Each oscillator type has a unique property which is controlled by the MODIFIER knob. All other parameters work in the same way, regardless of which type is chosen (Exception: Noise).

- Sawtooth: A high-quality analog-style sawtooth oscillator. Each sawtooth in the bank has a virtual master oscillator, which allows to hard sync the sawtooth to the frequency of the master oscillator. When set to zero, the master oscillator has no effect, resulting in a regular sawtooth waveform output. As the MODIFIER knob is increased, however, the master frequency is progressively lowered relative to the specified pitch, producing the well-known oscillator sync sound.

- Digital Saw: A basic sawtooth waveform, followed by a highpass filter. The digital sawtooth oscillator generates a lot of aliasing noise, which is particularly audible at high frequencies. This is useful as a creative effect, for instance to synthesize noisy high string notes, as popularized by older digital gear. The MODIFIER knob adjusts the frequency of the highpass tracking filter, progressively increasing its relative frequency. As a result, the first overtones of the sawtooth get attenuated.

- Pulse: A pulse waveform with adjustable pulse width (see fig. 3.3). The MODIFIER knob controls the pulse width, the center position of 50% corresponds to a square wave.

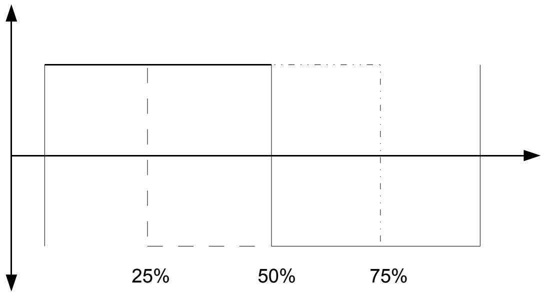



Figure 3.3: Pulse Width.

- Sine-Triangle: This oscillator blends seamlessly between sine and triangle waveform shapes. The MODIFIER knob controls the mix ratio. Turned fully to the left, a pure sine wave is created. Turned to the right, only the triangle is audible.

- Saw-Triangle: Same as above, except that this oscillator mixes a sawtooth and a triangle waveform.

- Noise: Generates random white noise. Use MODIFIER to filter the noise, in order to obtain different timbres. Note that the tuning parameters have no effect on this oscillator type, since white noise has no pitch.

- Ringwave: Synthesizes bell-type waveforms using ring modulation. Use MODIFIER to change the timbre of the sound.

- WT (Wavetables): A selection of wavetables, each containing a number of distinct waveforms, with a smooth blend between them. Use the MODIFIER knob to set the position in the wavetable. Modulate the modifier knob with an envelope or LFO to obtain typical wavetable sounds.

## COUNT

This parameter specifies the number of oscillators to use per bank. Set to OFF, the entire oscillator bank is turned off and will not consume any CPU resources.

## DYAD

The DYAD parameter allows to double the entire bank with all its oscillators and their settings, and transpose it up by a selectable number of semitones. For example, if COUNT is set to 5 and DYAD is set to +24, Antidote would play back 10 oscillators in total. The first 5 oscillators would run at their regular pitch, the extra 5 oscillators two octaves higher. This feature is useful to build chord stabs. It can also come in handy if one oscillator bank is set to noise and additional tuned oscillators are needed.

## SEMI

The SEMI control adjusts the primary tuning of the oscillator bank in semitones. The range spans +/- 24 semitones. A larger range can be obtained by using the modulation matrix, if required.

## FINE

This parameter adjusts the fine tuning of the oscillator bank in cents. A value of +/- 100 corresponds to a semitone.

## DETUNE

This parameter adjusts the detune of all oscillators in the bank. Higher values corresponds to stronger detuning. Note that more than one oscillator needs to be chosen in the COUNT field for this parameter to be audible. Detune works on all oscillator types except noise.

## SPREAD

This parameter spreads the oscillators in the stereo field, from monophonic to full stereo. Note that more than one oscillator needs to be chosen in the COUNT field for this parameter to be audible.

## PHASE

Whenever a note is triggered, all oscillators need to start at a certain position within the waveform cycle. PHASE sets this initial starting phase, from 0 to 359 degrees. Turned fully clockwise, the oscillators are in free run mode, which means they start at random phases. This is the default behavior, and strongly recommended when using more than one or two oscillators in the bank. Otherwise, strong beating will occur when all oscillators in a bank start from the same waveform position.

## MODIFIER

See WAVEFORM and MODIFIER above.

## KEYTRACK

When triggering a note, usually its MIDI key number will tune the oscillators to the proper frequency for this key. With semi and fine tune at their default positions, pressing MIDI note A4 would set an oscillator base frequency of 440 Hz, for example. In some cases, however, it can be useful to change this key tracking. When creating percussive sounds, it is often preferable to turn off key tracking completely. This is accomplished by setting key track to zero, which means the key number has no effect on the pitch of the oscillators.

#### PAN

Adjusts the panorama position of the entire oscillator bank, from left to right. The default is center.

### 3.2.2 Sub Oscillators and Mixer

The levels of both oscillator banks as well as their respective sub oscillators can be adjusted by using the faders in the mixer section. The first sub oscillator, labeled SUB 1, emits a square wave. Its pitch is tuned precisely one octave below oscillator bank 1. The second sub oscillator, labeled SUB 2, emits a sawtooth wave, pitched one octave below oscillator bank 2. Sub oscillators are often used to add more body to a sound.

When a fader is set to zero, that oscillator is inaudible. The DRIFT parameter adjusts the tuning drift of all oscillators over time, a property known of vintage analog synthesizers. To have perfect and stable oscillator tuning, the fader should be at zero.

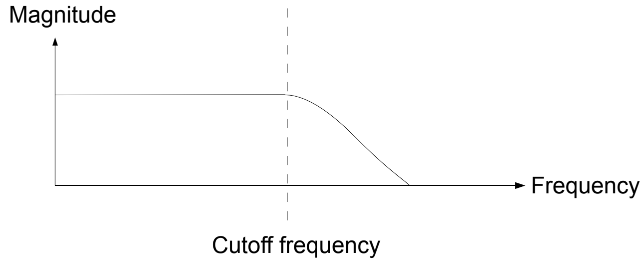
## 3.3 Filter

The raw oscillator sound is typically too bright to be useful. Furthermore, the periodic nature of the oscilators results in a dull timbre. Many natural instruments like a flute or piano feature a short, bright transient behavior, and then decay to a more steady, darker timbre. This behavior can be modeled by using a time-varying filter. The filter section is located below the first LFO.

#### CUTOFF

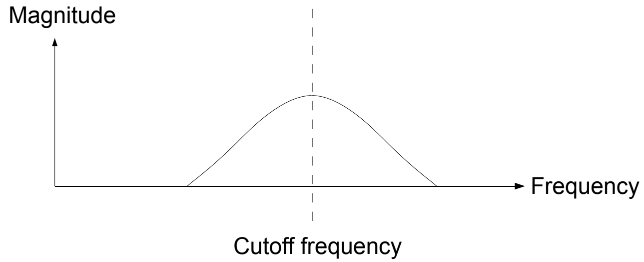
Perhaps the most important filter parameter is the CUTOFF knob. It sets the corner frequency where the filter operates. Its meaning depends on the filter type chosen:

- For the low-pass filter types, frequencies above the cutoff frequency are damped:



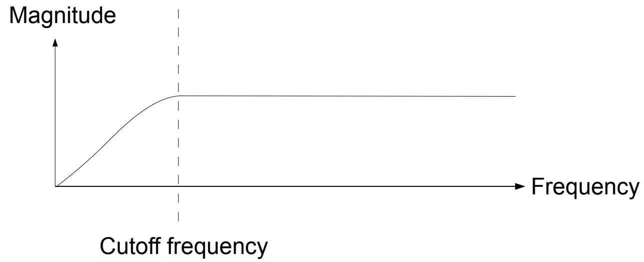
Antidote features four lowpass filter types, which differ in how strong the damping is per octave. The one-pole filter will attenuate frequencies above the cutoff by 6 dB, the two-pole by 12 dB, the three-pole by 18 dB, and finally the four-pole by 24 dB per octave.

- The band-pass filter damps frequencies around the cutoff frequency. As a result, bass and treble get attenuated.

  The Bandpass filter attenuates the frequencies around cutoff by 12 dB per octave.

- The high-pass filter attenuates all frequencies below the cutoff frequency and passes the higher frequencies unchanged.

  The Highpass filter attenuates the frequencies below cutoff by 12 dB per octave.

- Diode Ladder

  This filter is a special kind of low-pass filter, which models analog filter circuits based on diodes (or transistors hooked up as diodes). The response and the resonant tuning of such filters differs from the standard low-pass filters, and is useful to recreate some vintage analog sounds.

Of the above filter types, the low-pass filter types are the most common, as they fully preserve the bass frequencies and allows the natural progression from a bright to a dark timbre when being modulated.

To modulate the cutoff frequency and produce a dynamically changing timbre, the LFOs and filter envelope can be used. Both options will be discussed later in this chapter.

## RESO

If the output of a filter is fed back to its input, resonance occurs, which is a sinusodial oscillation near the cutoff frequency (see fig. 3.4). The RESO knob controls the depth of this effect. At lower settings, resonance can be used to add presence to a sound. Using higher settings, the sinusodial oscillation gets strong enough to use the filter in a similar fashion as an oscillator (try setting the low-pass filter to maximum resonance, with Key Track set to 100% and Cutoff to 0%). This property is furthermore useful to create special effect sounds such as laser guns, electronic bass drums etc.

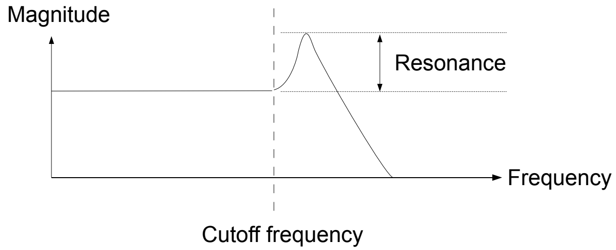



Figure 3.4: Response of a resonant low-pass filter.

## ENV

This knob controls how much the filter envelope (described later in this chapter) affects the cutoff frequency. Set to zero, the filter envelope has no effect on the cutoff frequency. At 100%, the envelope spans the entire cutoff range from the minimum to the maximum value.

Most sounds will use a low-pass filter with an envelope amount setting in between the two extremes and the envelope attack and sustain set to their minimum values. This creates the most common timbre which is a bright start followed by a darker sustain stage, a property shared by many acoustic instruments.

In rare cases, you may also want to set the envelope amount to a negative value. This can be helpful to create sounds which become bright when releasing a key. A negative envelope amount can be set using the modulation matrix, with the envelope amount knob set to zero.

## KEY TRACK

The key track parameter determines how much the cutoff frequency is affected by the MIDI key note. Set to zero, all notes share the very same cutoff frequency as specified by the **CUTOFF** parameter. Nonzero values move the cutoff according to the key pressed, with higher keys corresponding to higher cutoff frequencies.

## 3.4 Modulation Envelope

An envelope is used to model the progression of timbre, volume or pitch of a sound, from start to finish. An envelope is triggered whenever a key is hit. The modulation envelope can be assigned to almost any sound parameter via the modulation matrix. All envelopes in Antidote can be described by four stages called Attack, Decay, Sustain and Release (ADSR), see fig. 3.5.

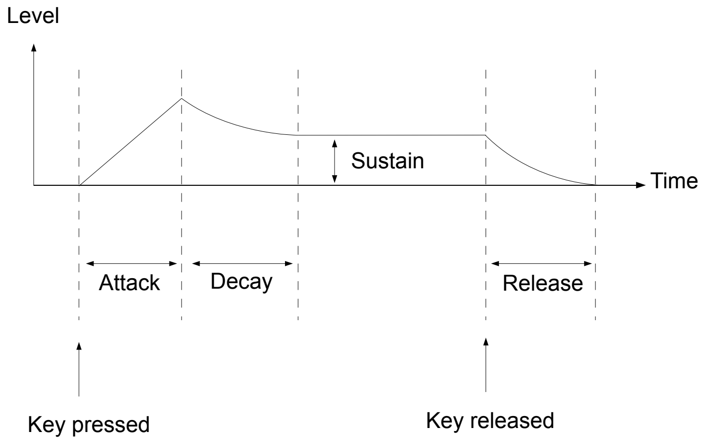



Figure 3.5: The modulation envelope.

### ATTACK

The ATTACK parameter specifies the duration it takes for the envelope to reach its maximum value. When set to zero, the envelope immediately starts at the peak value. The slope of the attack stage is linear.

### DECAY

After reaching the peak, the decay stage commences. During the decay stage, the envelope falls back to a lower level, the sustain level. The DECAY control specifies the duration of the decay stage, i.e. how long it takes to fall back to the sustain level. The slope of the decay stage is logarithmic.

### SUSTAIN

This parameter specifies the sustain level that is reached after the decay stage ends. The sustain stage lasts as long as a key is pressed.

### RELEASE

The final release stage is triggered whenever a key is released. The RELEASE parameter specifies the duration it takes the envelope to hit zero. The slope of the release stage is logarithmic like the decay stage.

## 3.5 Filter Envelope

The filter envelope modulates the filter cutoff frequency and thus the timbre of the sound. Many sounds start with a bright timbre and then decay to a darker tone. This behavior can be modeled with the filter envelope. The depth of the effect is controlled with the ENV knob in the filter section. The filter envelope has the same shape as the modulation envelope (see fig. 3.6).

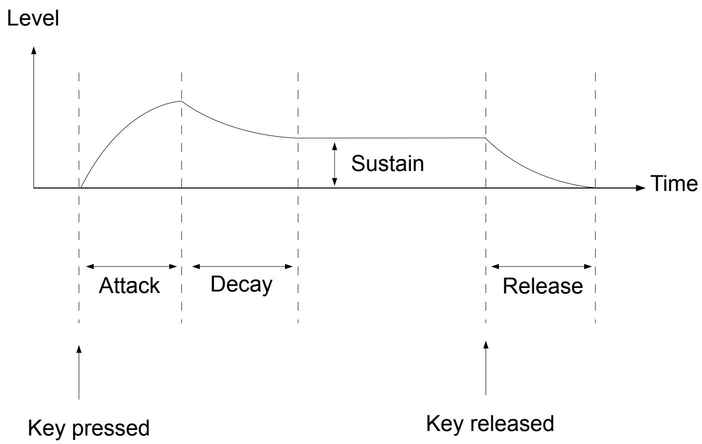



Figure 3.6: The filter envelope.

### ATTACK

The ATTACK parameter specifies the duration it takes for the envelope to reach its maximum value. Most sounds use a setting near the minimum in order to start bright.

### DECAY

After reaching the peak, the decay stage commences. During the decay stage, the envelope falls back to a lower level, the sustain level. The DECAY control specifies the duration of the decay stage, i.e. how long it takes to fall back to the sustain level.

### SUSTAIN

This parameter specifies the sustain level that is reached after the decay stage ends. The sustain stage lasts as long as a key is pressed.

### RELEASE

The final release stage is triggered whenever a key is released. The RELEASE parameter specifies the duration it takes the envelope to hit zero. Note that when SUSTAIN is set to zero, the RELEASE parameter may have no effect if the envelope has previously reached zero already.

## 3.6 Amplitude Envelope

Located next to the Filter envelope, the amplitude envelope controls the progression of the volume of a sound (see fig. 3.7). It works in the same manner as the filter and modulation envelopes.

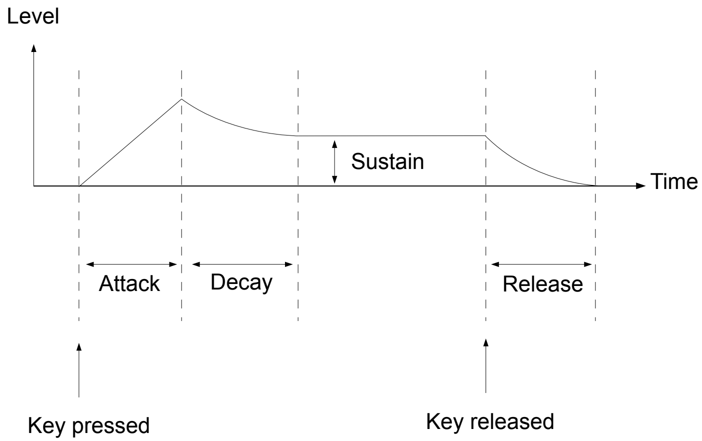



Figure 3.7: The amplitude envelope.

### ATTACK

The ATTACK parameter specifies the duration it takes for the amplitude envelope to go from zero to its maximum level, with a linear slope.

### DECAY

The DECAY parameter specifies the duration of the decay stage, i.e. how long it takes the amplitude to fall back to the sustain level.

### SUSTAIN

This parameter specifies the sustain level that is reached after the decay stage ends. The sustain stage lasts as long as a key is pressed.

### RELEASE

The final release stage is triggered whenever a key is released. The RELEASE parameter specifies the duration it takes the envelope to hit zero. Note that when SUSTAIN is set to zero, the RELEASE parameter may have no effect if the envelope has previously reached zero already.

## 3.7 LFOs

Using oscillators, the filter unit and envelopes, it is possible to control the basic properties of a sound, such as timbre, volume and pitch. For many bass and percussive sounds this is enough to get good results, but for pad or lead type sounds, the sustain stage may still sound dull. This is because the pitch, filter cutoff and volume are steady in this stage and do not change.

This is where LFOs (low frequency oscillators) come into play. LFOs work just like ordinary oscillators, generating a periodic signal using similar waveforms (see fig. 3.8). They are inaudible, however, and their only purpose is to continually change one or more aspects of the sound. The most typical applications are modulating the volume, cutoff or pitch, resulting in a vibrato or tremolo effect. Antidote's two LFOs are much more capable than that, however, as almost any parameter discussed so far can be used as a modulation destination. Additionally, LFOs can modulate each other in volume or frequency to obtain yet more interesting variations.

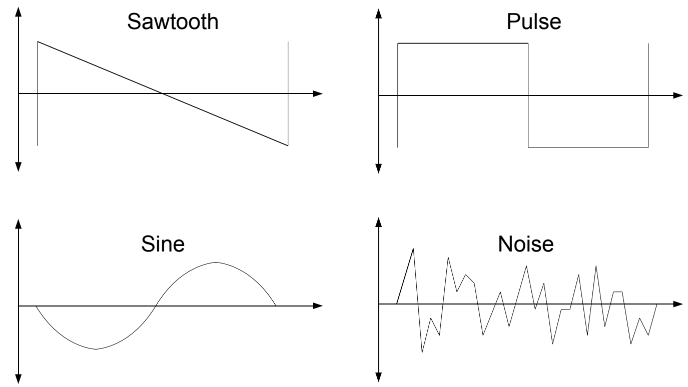



Figure 3.8: Basic LFO waveforms.

The two LFOs are controlled by the sections labelled LFO 1 and LFO 2 on the left and right side of the user interface. The LFO can target an arbitrary sound parameter, which can be chosen by clicking on the drop-down list box. Sometimes it is desirable to control more than just a single target; in this case, the modulation matrix can be used, which is covered in a later chapter of this manual.

## Shape

Use the drop-down list to select one of the available waveform shapes (see fig. 3.8). Sawtooth, Ramp, Triangle and Sine are periodic waveforms. S+H Noise (Sample-and-Hold Noise) is a random signal. It can be used for special effects or to simulate the behavior of old analog hardware.

## RATE

By default, LFOs run at a constant rate specified in Hz, independent of the MIDI note played. Typical settings are between 3-6 Hz for vibrato or tremolo effects. When the **SYNC** switch is enabled, the rate is specified in units of the current song tempo, such as quarters, eights or sixteenths notes, with either their standard durations, or in triplet (T) or dotted (*) form. Examples:

- 1/4 specifies the duration of a quarter note.

- 1/8+ sets the modulation rate to a dotted eight note.
- 1/16T sets the modulation rate to a sixteenth triplet.
- 1/1 sets the modulation rate to span one bar.
- 2/1 sets the modulation rate to span two bars.

### FREE-RUN

When free-run is enabled, the corresponding LFO runs continuously, independent of whether any keys are pressed. The LFO is thus global and shared by the entire machine. When free-run is disabled, the LFO is reset each time a key is pressed. In this mode, each voice has its own LFO, and its initial starting phase can be set via the modulation matrix, if desired.

## 3.8 Effects

Antidote offers eight effects units to further enhance the sound coming from the synthesis engine. All of them may be used simultaneously. It is important to note that the effects are global, that is all voices are first summed and then processed by the effect section. A selection of effect sound parameters can be modulated via the modulation matrix.

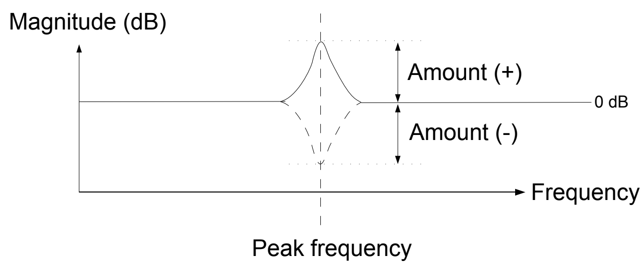
The effects are processed from left to right, in the order they appear. The equalizers (EQs) are applied first, Compressor is processed last.

### 3.8.1 EQ

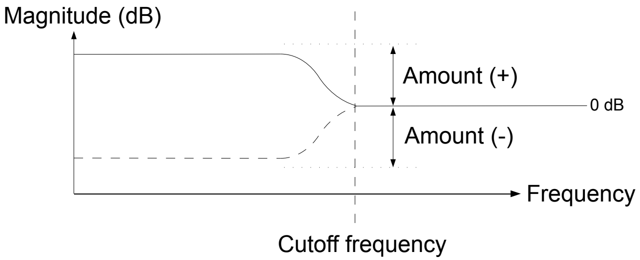
An equalizer (EQ) is used to boost or attenuate a certain frequency range. There is three basic types:

### Type

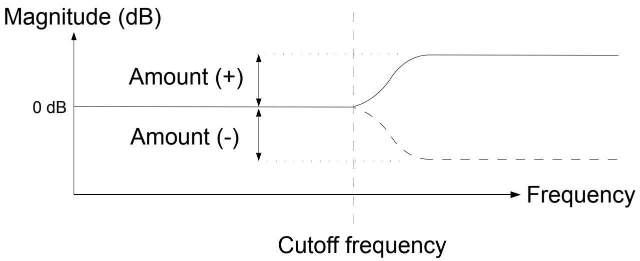
- **Peaking** amplifies or attenuates the region around the chosen frequency.




- **Lo Shelf** amplifies or attenuates frequencies below the chosen frequency.




- **Hi Shelf** amplifies or attenuates frequencies above the chosen frequency.




## LO GAIN

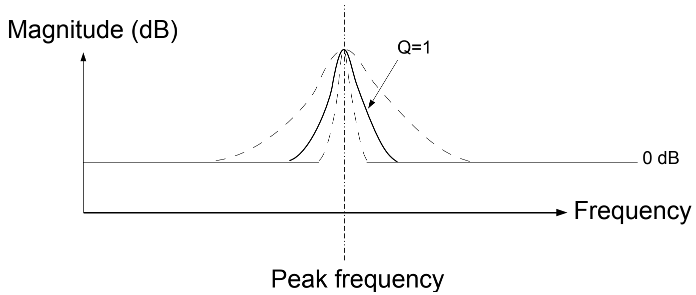
The first EQ in Antidote is a low shelf filter, centered around 80 Hz. The knob controls the cut or boost amount in decibels. At center position, there is no effect.

## MID FREQ

The second EQ in Antidote is a peaking filter for the middle frequencies. This knob sets the operation frequency of the EQ in Hz.

## MID Q

Adjusts the steepness of the Mid EQ. Q settings below 1 create broad peaks, while higher settings create narrow peaks.




## MID GAIN

Specifies how much to attenuate or boost the chosen mid frequency. At center position (0 dB) the signal is not affected.

### HI GAIN

This EQ is a high shelf filter, centered around 12 kHz. The knob controls the cut or boost amount in decibels. At center position, there is no effect.

### 3.8.2 Bass

The bass effect is part of the EQ section. It models a unique circuit found in some vintage analog synthesizers, which yields a particular frequency response in the low and low-mid regions. Using the bass effect will help emulate the sound character of such synthesizers.

### BS FREQ

Sets the center frequency of operation in the circuit. The default setting yields the classic vintage sound, but feel free to experiment with other settings. Note that towards higher frequencies, the effect will gradually vanish and become less pronounced.

### BS AMT

Adjusts the magnitude of the bass effect. When set to zero, it is completely bypassed.

### DRY/WET

Blends between the dry and processed signal. Note this includes both the EQ and Bass section.

### 3.8.3 Distortion

A distortion effect changes the signal in a nonlinear fashion, thereby introducing additional overtones. This results in a rather harsh sound, especially at extreme settings using high amplification factors.

### TYPE

Three different distortion modes are available in Antidote, Overdrive, Grunge, and Rate Crush. The first two modes are emulations of classic guitar stomp boxes. The third mode, Rate Crush, reduces the sampling rate of the signal by employing a sample-and-hold circuit.

### DRIVE

Adjusts the gain applied to the signal when entering the nonlinear distortion stage. Higher settings cause more distortion.

### SYMMETRY

Offsets the signal before the distortion stage. This will introduce even order harmonics and change the timbre of the sound. Note that changing symmetry can result in silence if the signal level is too low. Increase the drive knob in such cases.

### TONE

The tone control processes the signal after it has been distorted. It changes the mid frequencies of the distorted sound and thus the overall timbre of the effect.

### LOW CUT

Use the low cut filter to roll off low frequencies, to remove any mud in the bass region.

### HIGH CUT

Use the high cut filter to roll off high frequencies, often useful to reduce the harshness of the distortion effect.

### DRY/WET

Blends between the dry and processed signal. Combined with the Drive knob, the distortion can be tamed where needed.

### 3.8.4 Phaser

A phaser modifies a signal with a series of filters and then mixes it with the dry signal. The cutoff frequency of the filters is continuously varied.

### STAGES

Sets the number of phaser stages (2, 4 or 6). More stages result in a more distinct phasing sound, while less stages sound more subtle.

### FREQ

Adjusts the bottom frequency for the phaser. This is the lowest frequency the phaser will sweep to.

### SPREAD

Use this parameter to spread the filter poles, which will change the overall timbre of the phasing effect.

### FEEDBK

The output of the phaser can be fed back into the input, making the overall phasing effect a lot more pronounced. Both positive and negative feedback is allowed.

#### RATE

The rate of the filter modulation, relative to the current tempo (see the LFO section for a description of the available modes).

#### MOD

Sets the amount of filter modulation. When set to zero, the phaser acts as a static filter with the frequency determined by the FREQ knob.

#### LR OFFSET

Offsets the modulation between left and right channels. This makes the phaser sound stereo, even when its input signal is monophonic. Use this parameter to create wide stereo effects.

#### DRY/WET

Blends between the dry and processed signal.

### 3.8.5 Chorus/Flanger

Summing a signal with a time-delayed copy of itself creates a chorus effect, if the time delay is continuously varied as well.

When choosing small delay times and applying feedback, the classic flanger effect is obtained.

#### DELAY

Sets the base delay time in milliseconds.

#### RATE

Sets the modulation rate in Hz.

#### DEPTH

Sets the modulation depth from 0% to 100%.

#### FEEDBACK

Feeds the delay output back to the effect input, creating resonances. Both positive and negative feedback is possible in Antidote's chorus.

#### LR OFFSET

Offsets the modulation phase between the left and right channels. This makes the chorus sound stereo, even when the input signal is monophonic. Use this parameter to create spatial effects.

#### DRY/WET

Blends between the dry and processed signal.

### 3.8.6 Delay

A delay effect produces a series of echoes. The duration of the echoes is always locked to the host tempo in order to guarantee a musically useful result. Two different delay types are available.

#### TYPE

- **Simple** creates a series of echoes centered in the stereo field.
- **Ping-Pong** creates echoes alternating between the left and right channels.

#### RATE L/R

The delay time can be specified independently for the left and right channels. It is always locked to the host tempo and is thus specified in quarters, eights, sixteenths etc., optionally in triplet (T) or dotted (*) form. Examples:

- 1/4 specifies an echo duration of a quarter note.
- 1/8+ sets the duration to a dotted eighth note.
- 1/16T sets the duration to a sixteenth triplet.
- 1/1 sets the duration to span an entire bar.

#### COLOR

The echoes can be processed by a 6 dB/oct lowpass or highpass filter, making each subsequent echo darker or brighter than the previous one. Negative values correspond to darker echoes, positive values to brighter echoes, at zero the echo timbre remains identical.

#### FEEDBACK

The feedback parameter allows you to adjust how often the echoes are repeated. The percentage specifies the level change from one echo to the next, so 100% creates an infinite series of echoes, 50% cuts the level of each subsequent echo in half etc.

#### MOD-RATE

The L/R delay times can be modulated, to obtain a full lush stereo sound. This knob sets the rate of modulation in Hz.

#### MOD-AMT

The amount of delay line modulation. Set this knob to zero to turn off modulation entirely.

#### DRY/WET

Blends between the dry and processed signal.

### 3.8.7 Reverb

A reverb effect is used to create the illusion of a sound being played back in a spatial environment such as a living room, hall or cathedral. The reverb effect in Antidote is designed to give best results for synthetic sources, which are often more difficult to process than natural sounds.

#### PREDELAY

Adjusts the onset of the reverberated signal. When set to zero, the reverberated signal commences almost immediately. Higher settings delay the signal, which can be useful to change the perception of the room size.

#### TIME

Sets the reverb time in seconds.

#### HF DAMP

Using the high frequency damp parameter, the simulated room's wall materials can be adjusted. Higher settings correspond to reflective walls, lower settings to very absorbent ones. Lower settings will cause the reverb trail to become dark more quickly.

#### LOW CUT

The low-cut filter in the reverb effect can be used to remove unwanted low frequencies from the processed signal. This is useful for sounds containing strong bass frequencies, such as bass drums etc. Note that the dry signal is not affected by this, only the reverberated signal.

#### HIGH CUT

Use the high-cut filter to remove unwanted high frequencies from the processed signal. The dry signal is not affected by this, only the reverberated signal.

#### MOD

In a real environment, the sound of reverb is always alive. Synthetic reverbs, on the other hand, sound dull if they process the signal in a strictly linear fashion. Using synthetic sounds as input, the situation is even worse, as synthetic sounds may sound dull to begin with. Modulation is the cure, and with the MOD knob you can specify how much modulation to apply. With enough modulation, even a plain sawtooth will sound great when reverberated in Antidote.

## DRY/WET

Blends between the dry and processed signals.

### 3.8.8 Compressor

A compressor changes the dynamics of a signal. It is usually applied last in a signal chain, to make the signal louder or more punchy. When Antidote's delay and reverb units are enabled, it is particularly easy to hear the difference with and without compression applied.

## RATIO and THRESHOLD

The compressor monitors the peak of the incoming signal, then reduces the level above the specified threshold in the ratio you specify. For instance, if the threshold is -10 dB and the compression ratio is 2:1, every decibel above the threshold is cut in half, so a -8 dB signal becomes -9 dB, etc. When set to 100:1, the compressor acts like a limiter, not allowing levels above the threshold.

## ATTACK

The attack time specifies how quickly the compressor will react to peaks that exceed the threshold. The attack can be set down to the microsecond range, which essentially gives instant response. For special effects or to shape transients, higher values can be used.

## RELEASE

Use the release time knob to set the time it takes the compressor to go from compression to idle state. Typical values range from 50-200 ms. Often, you'll want to set the fastest attack and release times you can get away with. As soon as you hear disturbing artifacts, however, increase the attack or release time, or alternatively lower the threshold or ratio for a more gentle compression.

## DRY/WET

Blends between the dry and processed signals. Often a compressor is used 100% wet, but other values permit the so-called parallel compression, which gives yet more freedom in shaping the sound.

# 4 Modulation Matrix

One of the biggest strength of subtractive synthesizers is their ease of use. The pitch, timbre and volume of a sound and its progression over time can be controlled in a simple and straightforward way. The simplicity is achieved by employing a fixed structure with a limited set of parameters, however.

In order to create more complex patches, modern synthesizers offer a modulation matrix, where you can choose from a set of sources and link them to almost any sound parameter (see fig. 4)). In Reason, the concept of a modulation matrix becomes yet more powerful, as you can connect CV cables from other Reason devices and modulate Antidote's sound parameters with them.

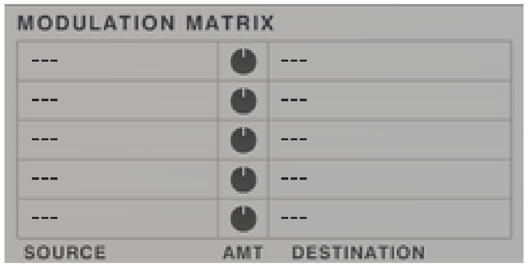



Figure 4.1: Modulation matrix.

The modulation matrix in Antidote is located in the center of the interface. It comprises 5 rows with source, amount and destination controls. The modulation envelope and both LFOs have one additional destination available.

After choosing a source and a destination, the AMT knob specifies how much to modulate the destination parameter. The modulation is bipolar, both positive and negative values are permitted. This is often useful, for example you can create the classic inverted filter envelope this way.

Destinations can be modulated more than once. For instance, the filter cutoff frequency can be modulated with a LFO and an envelope simultaneously.

## 4.1 List of Sources

### Velocity

The velocity of the note, ie how hard you press the key.

### Arp Accent and Gate

The accent and gate signal coming from the arpeggiator. See chapter 5 for a more detailed description of the arpeggiator's accent and gate signals.

## LFO 1

The output of the first LFO.

## LFO 1*MW

The output of the first LFO, multiplied by the modulation wheel. Use this parameter to make the LFO amount dependent on the modulation wheel, a frequently used performance tool.

## LFO 1*AT

The output of the first LFO, multiplied by aftertouch. Use this parameter to make the LFO amount dependent on aftertouch.

## LFO 2

The output of the second LFO.

## LFO 2*MW

The output of the second LFO, multiplied by the modulation wheel. Use this parameter to make the LFO amount dependent on the modulation wheel, a frequently used performance tool.

## LFO 2*AT

The output of the second LFO, multiplied by aftertouch. Use this parameter to make the LFO amount dependent on aftertouch.

## MWheel

The value of the modulation wheel.

## PWheel

The value of the pitch wheel. Note that this is a bipolar source, which can have positive or negative values. Also note that this source reflects the state of the pitch wheel directly, independent on the up / down settings on the back panel.

## ATouch

The amount of aftertouch applied.

## Constant

Constant means the source is a constant 100% and does not vary. Use this to change the operation range of sound parameters. For example, you can program a negative filter envelope this way (choose Constant as source, set a negative amount, then Filter Envelope as a destination).

### Random

A random value, which changes with every Note On. You can use this for random modulation related to the onset of a note. For instance you could have random volume each time you press a key.

### Mod Env

The current state of the modulation envelope.

### Filter Env

The current state of the filter envelope.

### Amp Env

The current state of the amplitude envelope.

### CV 1,2,3,4

The CV Input signals coming from the back panel.

### Key Follow

Yields a value proportional to the frequency of the current MIDI key pressed. This source can be used to shorten the envelopes for high notes, in order to mimic the properties of some acoustic instruments (e.g. guitar).

## 4.2 List of Destinations

### Pitch Coarse

The coarse pitch, ranging from -48 to +48 semitones. Frequently used amounts for this parameter thus include +/- 25 (one octave) and +/- 50 (two octaves).

### Pitch Fine

Use this destination to change the fine tuning in cents (cents are the fraction of a semitone).

### Filter Cutoff

The filter cutoff frequency, perhaps the most important destination. Modulating the filter cutoff frequency will largely shape the timbre of a sound.

### Filter Resonance

The filter resonance.

### Filter Resonance

The filter envelope amount.

### Filter Keytrack

The filter keytrack value. Modulate this parameter with a constant source to set negative values.

### Osc 1 Semi

Same as Pitch Coarse, but only affecting the first oscillator bank and its sub oscillator.

### Osc 1 Fine

Same as Pitch Fine, but only affecting the first oscillator bank and its sub oscillator.

### Osc 1 Detune

Changes the value of the Detune knob.

### Osc 1 Spread

Changes the value of the Spread parameter.

### Osc 1 Modifier

Changes the value of the Modifier knob.

### Osc 1 Pan

Changes the panorama of the oscillator bank. Note that this is a bipolar destination, to sweep from left to right and back, be sure to center the knob.

### Osc 1 Volume

Changes the volume of the first oscillator bank.

### Osc 2 Semi

Same as Pitch Coarse, but only affecting the second oscillator bank and its sub oscillator.

### Osc 2 Fine

Same as Pitch Fine, but only affecting the second oscillator bank and its sub oscillator.

### Osc 2 Detune

Changes the value of the Detune knob.

### Osc 2 Spread

Changes the value of the Spread parameter.

## Osc 2 Modifier

Changes the value of the Modifier knob.

## Osc 2 Pan

Changes the panorama of the oscillator. Note that this is a bipolar destination, to sweep from left to right and back, be sure to center the knob.

## Osc 2 Volume

Changes the volume of the oscillator.

## Sub 1 Volume

Changes the volume of the first sub oscillator.

## Sub 2 Volume

Changes the volume of the second sub oscillator.

## Mix Volume

Mix Volume is a special destination parameter not related to any of the front panel controls. It adjusts the global volume of the synth prior to the effect chain. This is used for global amplitude modulation, either via one of the two LFOs, or via the Gate. Note that the Mix Volume destination is generally not useful for other purposes. For instance, it should not be used to fade in or fade out a sound. Use Osc 1 or Osc 2 Volume for such purposes, which work on a voice level.

## LFO 1 Amount

The amount of LFO 1 modulation.

## LFO 1 Rate

Changes the LFO 1 rate.

## LFO 1 Phase

Sets the inital phase of LFO 1, ranging from 0 to 359 degrees. If the LFO free-run mode is turned off, the LFO will reset whenever you press a key. The default phase is zero, but by using this destination you can specify other values. This is useful whenever the modulation is strong enough to impact the transients of your sounds.

## LFO 2 Amount

The amount of LFO 2 modulation.

## LFO 2 Rate

Changes the LFO 2 rate.

## LFO 2 Phase

Sets the inital phase of LFO 2, ranging from 0 to 359 degrees. If the LFO free-run mode is turned off, the LFO will reset whenever you press a key. The default phase is zero, but by using this destination you can specify other values. This is useful whenever the modulation is strong enough to impact the transients of your sounds.

## Filter Env Atk

Changes the value of the attack time of the filter envelope.

## Filter Env Dec

Changes the value of the decay time of the filter envelope.

## Filter Env Sus

Changes the value of the sustain level of the filter envelope. This parameter can be used for filter cutoff modulation, commencing with the onset of the sustain stage.

## Filter Env Rel

Changes the value of the release time of the filter envelope.

## Amp Env Atk

Changes the value of the attack time of the amplitude envelope. A useful destination to make the attack time dependent on velocity, for instance, with lower velocities generating a slower (and thus softer) attack. Another option is to slightly randomize the attack time, to make repeated notes sound more interesting.

## Amp Env Dec

Changes the value of the decay time of the amplitude envelope.

## Amp Env Sus

Changes the value of the sustain level of the amplitude envelope. This parameter can be used for amplitude modulation, commencing with the onset of the sustain stage.

## Amp Env Rel

Changes the amplitude envelope release time.

### Mod Env Curve

A special destination not related to any front panel control. The mod envelope curve parameter tweaks the shape of the modulation envelope from a linear attack to logarithmic/exponential shapes. This is useful when programming kick sounds, where the modulation envelope is targeting pitch. The shape of the modulation envelope is critical to the sound in such situations, so being able to change the curve shape allows for a wider range of sounds.

### Arp Gate Length

Changes the value of the Arpeggiator's Gate knob.

### Ext In Amount

If you connect an external input signal, its level can be modulated with this destination.

### EQ Mid Freq

Changes the value of the EQ's mid frequency control. This can be useful to create phaser-like sounds, by using large gain amounts and applying a LFO to this destination, which will then cause the EQ to sweep through a wide range of the frequency spectrum.

### Dist Tone

Changes the value of the Distortion's tone parameter.

### Dist Dry/Wet

Changes the value of the Distortion's dry/wet parameter.

### Phaser Freq

Changes the value of the Phaser's frequency parameter.

### Phaser Spread

Changes the spread parameter value.

### Phaser Feedback

Changes the feedback value.

### Phaser Mod

Changes the amount of modulation.

### Phaser Dry/Wet

Changes the dry/wet parameter.

### Chorus Depth

Changes the chorus depth.

### Chorus Feedback

Changes the chorus feedback.

### Chorus Dry/Wet

Changes the chorus dry/wet parameter.

### Delay Feedback

Changes the feedback parameter of the delay unit.

### Delay Mod Amt

Changes the modulation amount of the delay unit.

### Delay Dry/Wet

Changes the overall effect amount of the delay unit.

### Reverb Time

Changes the reverb time.

### Reverb Mod Amt

Changes the amount of reverb modulation.

### Reverb Dry/Wet

Changes the dry/wet amount of the reverb.

### Comp Threshold

Changes the compressor threshold.

### Comp Dry/Wet

Changes the dry/wet amount of the compressor.

# 5 Arpeggiator

An arpeggiator (short: ARP) is a module that generates monophonic melodic and rhythmical patterns from sustained MIDI notes. The available modes can be chosen from a drop-down list, and are described in the following paragraphs.

- Off: The arpeggiator is off and has no effect. This is the default setting.
- Up: All MIDI notes currently sustained are traversed from the lowest to the highest note.
- Down: All MIDI notes currently sustained are traversed from the highest to the lowest note.
- Up/Down: All MIDI notes currently sustained are traversed from the lowest to the highest note, then back.
- Down/Up: All MIDI notes currently sustained are traversed from the highest to the lowest note, then back.
- Random: All MIDI notes currently sustained are played back in random order.
- Gate/Acc: A special mode which does not generate any note sequence, but merely transmits the Gate and Accent signals to the modulation matrix. Read the GATE/ACCENT section below for more information on this topic.

## OCT

When octave is set to 1, the sequence generated by the arpeggiator matches the sustained MIDI keys. More variation is obtained by increasing the number of octaves, which will cause the arpeggiator to extent the note sequence additional octaves above the pressed keys.

## PAT

The arpeggiator can use one of 20 different patterns, to create a rhythmic feel and make the generated note sequences more interesting. The first pattern, which is default, consists of sixteenth notes only. Use this setting if you do not want to enforce a particular rhythm.

## LENGTH

This control changes the length of all notes in the pattern currently selected, from half to twice the length. At center position, the pattern remains unchanged.

## GATE/ACCENT

The arpeggiator constantly emits a gate and an accent signal. The Gate signal is by default in low state, and goes high for the duration of a note.

The accent signal is by default in low state as well, and goes high on every accented step. Usually, every note is accented, but some of the preset patterns contain separate note and accent data. The most important difference between Gate and Accent is their shape, however. The accent shape has a soft slope, which is useful for growling bass sounds. The gate has a more abrupt shape, and is useful to chop a sustained sound into a rhythmic pattern.

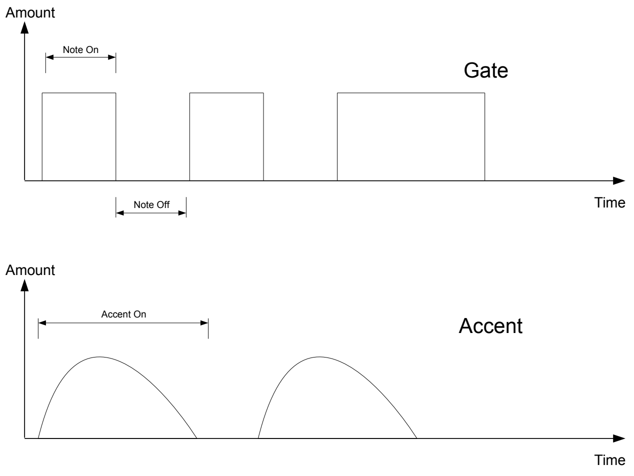



Figure 5.1: Gate and Accent.

# 6 Back Panel

The back panel hosts the audio and CV connections for Reason. A good understanding of how CVs work in Reason will enable you to obtain mind-blowing results with Antidote, and is obligatory to build complex patches which would otherwise exceed the functionality provided. A simple example is a patch requiring a third LFO. You can add one easily by connecting an external LFO to one of Antidote's CV inputs, then assign this CV input via Antidote's modulation matrix.

## CV Inputs

Antidote RE has four CV inputs, which can be freely used as modulation sources in the modulation matrix.

## Modulation CV Inputs

Standard Reason CV inputs, which are fixed to Pitch Bend, Modulation Wheel and Amplitude. Reason will automatically connect those CV inputs in some cases.

## Sequencer In

Allows to connect Reason's step sequencers (such as Matrix, RPG-8, or Thor's) to trigger monophonic note sequences using Gate/CV signals. Make sure to connect both the gate and CV cable.

## Arp Out

Sends notes from Antidote's arpeggiator to other Reason devices, in the Reason-specific Gate/CV form.

## Ext Input

A pair of audio inputs, which allow you to connect other stereo devices to Antidote and take advantage of Antidote's superior-quality effect chain. The input signal is mixed with Antidote's synthesizer output just before the first effect (EQ/Bass). The input level can be controlled and automated on the front side of the plugin.

## Output

Antidote's stereo output.

## Configuration

The configuration section sets the pitch wheel up/down range, as explained in the second chapter. Furthermore, the MIDI key velocity can be routed to the volume, panorama and filter cutoff parameters. While this could be done via the modulation matrix, using the knobs is faster, and saves modulation slots for other purposes.

# 7 MIDI Reference

Most knobs and buttons on the front panel can be remote controlled via MIDI. Antidote's default controller assignments follow common conventions and the MIDI standard as much as possible. The number of sound parameters Antidote offers, however, is higher than the amount of available MIDI controllers. Effect parameters are thus outside the MIDI range.

| **Antidote Parameter** | **CC #** |
|---|---|
| Polyphony | 15 |
| Master Volume | 7 |
| Glide | 5 |
| Ext Input Amount | 4 |
| Osc 1 Semi | 78 |
| Osc 1 Fine | 79 |
| Osc 1 Detune | 80 |
| Osc 1 Spread | 81 |
| Osc 1 Phase | 82 |
| Osc 1 Modifier | 25 |
| Osc 1 Key Track | 36 |

| **Antidote Parameter** | **CC #** |
|---|---|
| Osc 1 Pan | 37 |
| Osc 1 Waveform | 13 |
| Osc 1 Voices | 23 |
| Osc 1 Dyad | 33 |
| Osc 2 Semi | 83 |
| Osc 2 Fine | 84 |
| Osc 2 Detune | 85 |
| Osc 2 Spread | 86 |
| Osc 2 Phase | 87 |
| Osc 2 Modifier | 35 |
| Osc 2 Key Track | 76 |
| Osc 2 Pan | 77 |
| Osc 2 Waveform | 14 |
| Osc 2 Voices | 24 |
| Osc 2 Dyad | 34 |
| Osc 1 Volume | 8 |
| Osc 2 Volume | 9 |
| Sub 1 Volume | 10 |
| Sub 2 Volume | 12 |
| Drift | 70 |
| Filter Type | 44 |
| Filter Cutoff | 74 |
| Filter Resonance | 42 |
| Filter Envelope Amount | 43 |
| Filter Key Track | 46 |
| Filter Envelope Attack | 47 |
| Filter Envelope Decay | 48 |

| Antidote Parameter | CC # |
|---|---|
| Filter Envelope Sustain | 49 |
| Filter Envelope Release | 50 |
| Mod Envelope Attack | 26 |
| Mod Envelope Decay | 27 |
| Mod Envelope Sustain | 28 |
| Mod Envelope Release | 29 |
| Mod Envelope Target | 30 |
| Mod Envelope Target Amount | 31 |
| Amp Envelope Attack | 73 |
| Amp Envelope Decay | 75 |
| Amp Envelope Sustain | 71 |
| Amp Envelope Release | 72 |
| Lfo 1 Rate | 19 |
| Lfo 1 Rate Sync | 18 |
| Lfo 1 Shape | 20 |
| Lfo 1 Sync | 16 |
| Lfo 1 Free Run | 17 |
| Lfo 1 Target | 21 |
| Lfo 1 Target Amount | 22 |
| Lfo 2 Rate | 52 |
| Lfo 2 Rate Sync | 51 |
| Lfo 2 Shape | 53 |
| Lfo 2 Sync | 56 |
| Lfo 2 Free Run | 57 |
| Lfo 2 Target | 54 |
| Lfo 2 Target Amount | 55 |
| Arp Mode | 68 |
| Arp Hold | 69 |

| Antidote Parameter | CC # |
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| Arp Octave | 61 |
| Arp Rate | 62 |
| Arp Gate | 63 |
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| MM Slot 1 Amount | 106 |
| MM Slot 1 Destination | 107 |
| MM Slot 2 Source | 108 |
| MM Slot 2 Amount | 109 |
| MM Slot 2 Destination | 110 |
| MM Slot 3 Source | 111 |
| MM Slot 3 Amount | 112 |
| MM Slot 3 Destination | 113 |
| MM Slot 4 Source | 114 |
| MM Slot 4 Amount | 115 |
| MM Slot 4 Destination | 116 |
| MM Slot 5 Source | 117 |
| MM Slot 5 Amount | 118 |
| MM Slot 5 Destination | 119 |
| EQ/Bass Active | 93 |
| Distortion Active | 94 |
| Phaser Active | 124 |
| Chorus Active | 125 |
| Delay Active | 126 |
| Reverb Active | 127 |
| Compressor Active | 94 |
| EQ Low Gain | 129 |

| Antidote Parameter | CC # |
|---|---|
| EQ Mid Freq | 130 |
| EQ Mid Q | 131 |
| EQ Mid Gain | 132 |
| EQ High Gain | 133 |
| Bass Freq | 134 |
| Bass Amount | 135 |
| EQ/Bass Dry/Wet | 136 |
| Distortion Type | 137 |
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| Distortion Symmetry | 139 |
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| Distortion Level | 143 |
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| Phaser Spread | 146 |
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| Phaser Rate | 148 |
| Phaser Mod | 149 |
| Phaser LR Offset | 150 |
| Phaser Dry/Wet | 151 |
| Chorus Delay | 152 |
| Chorus Rate | 153 |

| Antidote Parameter | CC # |
|---|---|
| Chorus Depth | 154 |
| Chorus Feedback | 155 |
| Chorus LR Offset | 156 |
| Chorus Dry/Wet | 157 |
| Delay Type | 158 |
| Delay L-Rate | 159 |
| Delay R-Rate | 160 |
| Delay Color | 161 |
| Delay Feedback | 162 |
| Delay Mod Rate | 163 |
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| Delay Dry/Wet | 165 |
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# Credits



Concept: Daniel Thiel

Programming and Manual: Richard Hoffmann

Graphic Design: Marcin Lezak

## Sound Design

The last two characters of every patch name are the author's initials. The following table lists all sound designers who contributed patches to Antidote, and a website where you can learn more about their work.

| Ab. | Author Name | Website |
|---|---|---|
| EX | eXode (Daniel Thiel) | www.soundcloud.com/exodesound |
| KD | Koshdukai (Marco C.) | www.KoshdukaiMusicReason.blogspot.com |
| MG | Mike Gorman | www.combinatorhq.com |
| MK | Michael Kastrup | www.xsynth.com |
| OR | David Orbel | www.orbelmusic.com |
| RH | Richard Hoffmann | www.synapse-audio.com |
| SH | Sasha Radojevic | www.phuturetone.com |
| SC | Soundcells (Harald Karla) | www.soundcells.de |