



# DUNE 3

## User's Manual

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# 1. Introduction

Thank you for choosing DUNE 3 by Synapse Audio!

DUNE 3 is a next-generation software synthesizer plugin designed for creating music on a computer. It was developed with the highest possible audio quality in mind and offers exceptional flexibility. The latter is achieved by its modulation system, the new differential unison engine (DUNE) – hence the name. Despite the complexity of the synthesizer, fast vector processing as well as support for multiple processor cores allows DUNE 3 to run with a moderate CPU load on modern systems.

DUNE 3 was tested by experienced music producers to guarantee its ambitious design goals were met. It comes with high quality sounds, with few exceptions created entirely by professional sound designers.

DUNE 3 is available in VST®, AAX and Audio Unit formats.

## 1.1. Installation

Important note for DUNE 1 and DUNE 2 users: There is no need to uninstall DUNE 1 or DUNE 2 prior to installing DUNE 3, they all run fine in parallel. Furthermore note that DUNE 3 is patch-compatible with DUNE 2, so patches from DUNE 2 can be fully used in DUNE 3. The best way to do this is to simply move your patches/folders from DUNE 2's Soundbanks folder over to DUNE 3.

### Installation on Windows

Unzip “dune3.zip” and run SETUP.EXE to commence with the installation process. The installer will guide you through the necessary steps. During installation, you will be asked to select the location of your VstPlugins directory. Make sure to choose the correct directory for your host software. Refer to your host software's manual if you are unsure about where the host software's VstPlugins directory is located.

All content files like the factory presets, MIDI files, Wavetables and the manual will be placed in the directory you specify during installation. By default this is the user directory (Synapse Audio/DUNE 3).

The next time you start your host software DUNE 3 will appear in the VST instrument list.

## **Installation on Mac**

DUNE 3 comes with a dedicated installer application. Download and open the file named "dune3.dmg". Afterwards, double-click on the installer icon to begin the installation process. The installer will guide you through the necessary steps. The next time you start your host software DUNE 3 will appear in the AU and/or VST instrument list, depending on which format your host software supports.

## **1.2. Activation**

After installation, the next step is to provide your license key by using the Activation Tool. On PC/Windows the tool is launched on first plug-in instance insert in your host (UI must be opened). On Mac OS X the tool is launched right after installation is completed.

After typing in the key, you will be asked to activate the license. This is required just once per computer. The preferred way is Online Activation, which requires only a single click. Activation permanently enables DUNE 3 to run on your computer. You may activate DUNE 3 on three computers simultaneously, provided you are the only user of those computers (for multiple users, multiple licenses need to be purchased). Note that when choosing Online Activation, no personal data is transmitted in the process, it is a perfectly safe method of activation.

If you wish to activate DUNE 3 on a computer not connected to the Internet, choose Offline Activation. You will be given a key which you can save to a USB stick or write down on a sheet of paper. Now switch to a different computer with Internet access and log in to your account at:

<http://www.synapse-audio.com/support.html>

Click on the "Access product activations" link in the "Product activations" section on the left. Enter the key previously stored and you will receive a response code, which you can type into the Offline Activation dialog to complete the installation process.

### **1.3. Compatibility**

DUNE 3 should run on any VST or AU-compatible host and comes with both 32-bit and 64-bit versions. If you encounter any compatibility issues with your host software, do not hesitate to contact us ([service@synapse-audio.com](mailto:service@synapse-audio.com)). DUNE 3 has been tested under the following hosts:

- Ableton Live
- Apple Logic Pro
- Avid Pro Tools
- Bitwig Studio
- Cockos Reaper
- Image Line FL Studio
- Magix Samplitude
- Native Instruments Complete Kontrol
- Native Instruments Maschine
- PreSonus Studio One
- Propellerhead Reason
- Steinberg Cubase
- Synapse Audio Orion

## 1.4. System Requirements

In order to maximize sound quality, DUNE 3 employs complex DSP algorithms that are very CPU demanding. Optimized SSE vector processing, as well as support for multiple processor cores allow DUNE 3 to perform very well despite its high complexity. To achieve best performance, a modern computer is required, however:

- Macs require OS X 10.11 or later, and should be equipped with a 2.8 GHz Intel quad core processor or better. The Audio Unit runs on both Intel- and M1-based Macs.
- PCs require Windows 7 SP 1 or later, and should be equipped with a 2.8 GHz quad core CPU or better. Most importantly, Windows must be optimized for realtime audio, in order to maximize the benefits of DUNE 3's multi-threaded engine. Too many software packages or services running in the background can severely degrade performance.

Furthermore, it is important to choose a good latency/audio buffer size. We recommend to use between 10-20 ms, or 512 samples at a 44.1 / 48 kHz sample rate. On most systems, this should result in a good balance between low-latency, realtime feel and CPU performance. Note that using less than 128 sample buffers will disable multi-threaded processing, as the thread synchronisation overhead becomes too significant.

The memory requirement of DUNE 3 is approximately 100 MB per instance. Thus if you wish to run many instances, your system should be equipped with 8 GB of memory or more.

## 1.5. New features in DUNE 3

This section provides an overview of the most important changes since DUNE 2 – new users can skip this section. DUNE 3 adds many new features compared to its predecessor.

Despite many additions, DUNE 3 is still fully compatible with patches made in DUNE 2 and furthermore contains the entire factory soundbank from DUNE 2.



## Oscillators

The Oscillator section features a new type named "Swarm". This is a Supersaw-type oscillator with modulation built in, controlled by the additional RATE knob. This can make the Supersaw sound more alive. Furthermore this type usually requires less oscillators than the other types, thus saving CPU. Using say 8 oscillators only in the Swarm mode may have a similar effect to using 32 oscillators in one of the other modes.

## Filter Section

Where DUNE 2 had only a single filter plus an effect with a pre/post switch, DUNE 3 now contains two filters and an effect, with a freely adjustable order. The new Balance knob allows to blend seamlessly between both filters, the Link switch links all filter knobs to speed up the workflow in situations where the filter parameters are shared.

Apart from the structural changes, there is also new Filter and new Effect types. Of particular interest are the new analog-modeled lowpass filter types, as well as the Vocal and Formant effect programs allowing for completely new sounds not possible previously.

## LFO

A new LFO type, "Random Pulse", has been added. This type is similar to S+H (Sample and Hold), but randomizes the pulse timing instead of its amplitude.

## Effects

The two master effect busses offer the same effect types as in DUNE 2 (Chorus/Flanger, EQ, Phaser, Delay 1+2, Reverb, Distortion, Compressor), however each effect has one or more new modes that did not exist previously, thus greatly enhancing the sound design capabilities.

The Equalizer has received the largest upgrade. Using the technology from our GQ-10 Graphic Equalizer, adjusting the timbre of sounds has never been easier due to the interactive display and the new, powerful features like the low- and high-cut filters.

## **Undo/Redo**

DUNE 3 offers unlimited Undo/Redo via the PATCH menu.

## **MSEG Presets**

MSEG curves can now be stored and recalled in the MSEG editor.

## **Arpeggiator**

DUNE 3 now features two full arpeggiator units instead of just one. This greatly increases the creative possibilities, e.g. two independent sequences can be played at the same time using different voices.

## **Wavetable Editor**

Using the new comprehensive wavetable editor all factory wavetables can be modified, and new waveforms or wavetables can be created from scratch with great ease. Audio files in WAV format can be imported as well. See chapter 6. *Wavetable Editor* for a detailed explanation.

## **Skins**

Four skin sizes allow DUNE 3 to be used on many different target resolutions, in particular Full HD, 4K, 5K screens etc.

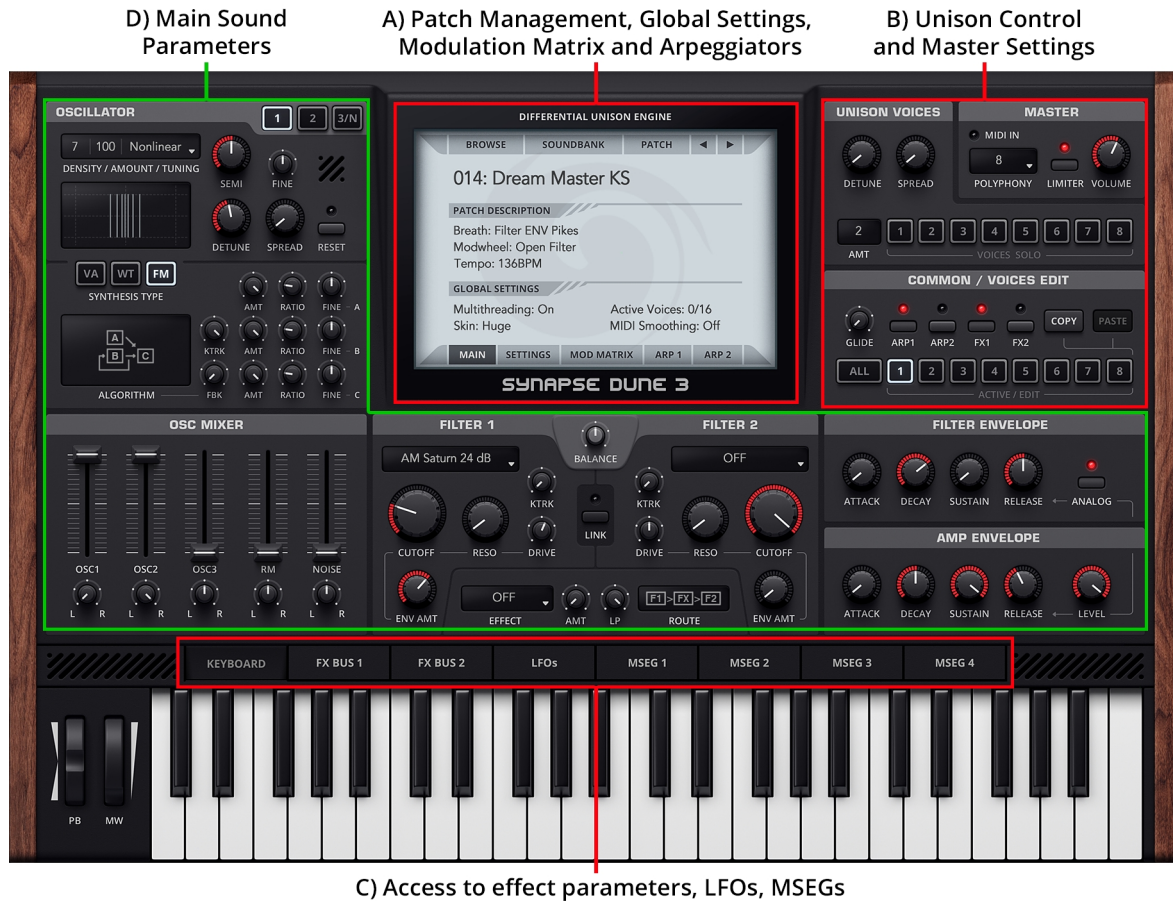
## **Performance**

DUNE 3 contains various performance improvements made to its core engine. Most of the performance gain is used by the new features that have been added. Some patches made in DUNE 2 can still see a 10-15% saving in CPU demand, however.

## 2. Basic Operation

### 2.1. Overview

The DUNE 3 interface can be divided into four parts:



The center screen (A) hosts the patch management, global and patch settings, the modulation matrix and the arpeggiator. The center screen is covered in chapter 3. *The Center Screen*.

The right side (B) contains the master section with global volume and the polyphony setting, as well as the unison voice controls. DUNE 3 allows to directly edit the eight unison voices, either all at once or individually. This makes DUNE 3 a multi-part synthesizer which facilitates synthesizing highly complex sounds.

The 8 switch buttons (C) toggle the bottom view between the keyboard, effect busses, LFOs, and four graphical envelopes (MSEG).

The remaining knobs and sliders (D) are DUNE 3's main sound parameters: The oscillator stacks, mixer, filters, and ADSR envelopes. The layout roughly follows classic synthesizers, with the oscillators on the left, followed by the filter in the middle and the envelopes on the right. Each unison voice has its own set of sound parameters. All sound parameters are covered in chapter 4. *Sound Parameters*.

## 2.2. Controlling parameters

Knobs, faders, and numerical displays are controlled by left-clicking on them, then dragging the mouse up or down in vertical direction (see fig. 2.2).

Use Ctrl+Click to set knobs to their default position, and Shift+Click to slow down the movement, in order to set precise values.

The mouse wheel is a very useful tool to speed up your workflow, and works on almost every parameter. Use the mouse wheel on numerical displays to increase or decrease the current value. Use the mouse wheel on drop-down lists, to select the previous or next parameter in the list. Knobs and faders can be fine tuned using the mouse wheel, in the graphical envelope editors it zooms the view.

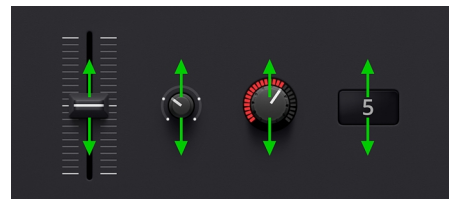


Figure 2.2. Controlling parameters.

## 2.3. MIDI Learn

Right-click on a parameter then choose "MIDI Learn" to link it to a particular MIDI controller. Afterwards, turn any knob or fader on your MIDI gear to link it to that parameter. Unlink controllers by right-clicking on a parameter then choosing "MIDI Forget". Note that without any assignments, DUNE 3 uses a default MIDI controller map (see chapter 9. *MIDI Reference*).

## 2.4. Modulation

Right-click on a parameter then choose "Modulate with" to set up a source/destination pair in the modulation matrix. Afterwards, change the amount parameter in the modulation matrix as desired. Setting up modulation in this manner is usually faster than choosing a destination from the popup menu in the modulation matrix.

Read chapter 8. *Modulation Matrix* for a more in-depth explanation of DUNE's modulation matrix.

## 2.5. Playing a patch

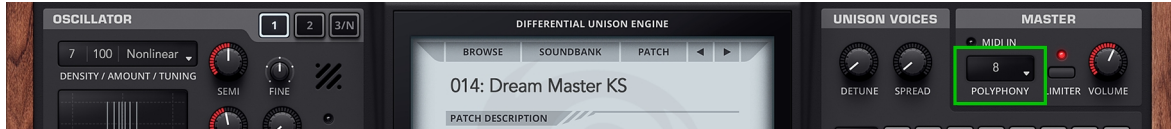
You can play individual notes directly by clicking on the keyboard at the bottom of the user interface. The vertical click position determines the velocity of the sound. A much better way, however, is to use your host sequencer or a MIDI keyboard.

Whenever you play notes via your host sequencer or keyboard, the MIDI In indicator should flash up. If this is not the case, then the required note information is not transmitted to the plugin. Refer to your host software's manual to address this issue. If you use an external MIDI keyboard, check if your host sequencer receives any data to begin with.

Select a patch by clicking on the patch name in the center screen. A more detailed description of the patch management is given in chapter 3. *The Center Screen*.

## 2.6. Adjusting Polyphony

DUNE 3 allows you to play multiple notes at the same time. The "Polyphony" drop-down menu in the master section on the right adjusts the number of notes that can be played simultaneously.



Each note you play triggers one or more unison voices (the number of active voices are displayed in the MAIN section of the center screen). Since each voice costs processor time, it is a good idea to limit the polyphony as much as possible.

When choosing Mono or Legato, only one note can be played at a time. Legato allows to smoothly go from one note to another without re-triggering the envelopes. This can be useful for bass and lead sounds, particularly in combination with the glide knob. It creates a unique playing feel and sound which can be better for monophonic lines. Using mono or legato modes also results in the smallest CPU usage possible.

Note that the voice polyphony can be adjusted per patch. While all patches should employ an adequate setting out of the box, your individual playing style or usage of sounds may require adjusting the polyphony parameter at times.

## 2.7. Pitch Bend and Modulation Wheel

At the bottom left of the user interface, the pitch bend and modulation wheels are located. The pitch bend wheel is used to temporarily shift the pitch up- or downwards. When released, it automatically snaps back to center position. The modulation wheel typically controls vibrato type effects, but can be used to modify other sound parameters as well. It remains in whatever position it was set to.

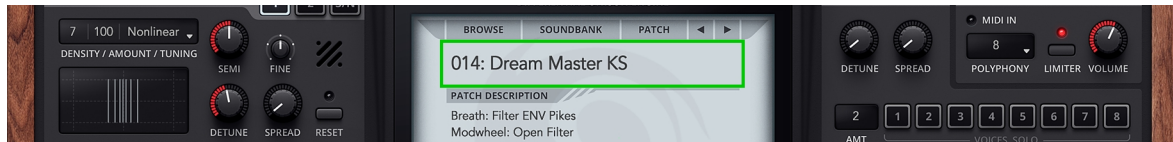
## 3. The Center Screen

This chapter explains all parameters inside the center screen. The center screen hosts the patch management, global settings, patch settings, the modulation matrix and the arpeggiator.

### 3.1. Main

#### Selecting a patch

Each musical sound that you can play is called a Patch. A patch is selected by clicking on the patch name in the center screen:



Alternatively click on the left/right arrow buttons to go to the previous/next patch inside the current bank.

DUNE 3 comes with multiple soundbanks, comprising a total of about 1000 patches. The patches are sorted into categories like Bass, Lead, Pads, SFX etc. The default bank shown is "DUNE 3 Factory", containing the latest patches made specifically for DUNE 3.

Click on BROWSER to open the patch management (see chapter 5. *Patch Browser*), which lets you search for specific patches, sort patches into categories, tag your favorite patches and so on. Click on SOUNDBANK to change the current bank/category.

Soundbanks and Patches are stored in the following directory:

- Mac OS X: /Library/Application Support/Synapse Audio/DUNE 3/Soundbanks
- Windows: Documents\Synapse Audio\DUNE 3\Soundbanks

Each patch is a single Cubase .FXP file, a common format for storing patches.

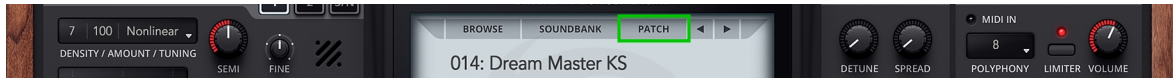
Any directory within the Soundbanks directory represents a bank.

Below the patch name, the patch description section gives some information about the currently selected patch, for example what the modulation wheel does to the sound, or whether the patch is intended for a particular tempo.

If you accidentally switch to another patch during editing a patch, do not panic. The Undo/Redo function will help you.

## Patch menu

The patch menu is invoked by clicking on "Patch":



- **Load Patch** loads a particular patch from disk. This can be useful to browse patches outside of DUNE's Soundbanks folder. Most of the time, however, you should store patches in your Soundbanks folder, in order to be able to browse a whole bank quickly.
- **Save Patch** allows you to save the current patch to disk. Always save your changes before switching to a different patch or closing your host application, otherwise you will lose your changes. Of course you can also simply save the project in your host sequencer, this will automatically save the active patch as well. Any patch that has been modified is marked with an asterisk (\*), after saving the patch, the asterisk is removed.
- **Copy FX 1/2** copies all effect bus 1/2 parameters into the clipboard, in order to apply them to a different patch or to a different DUNE 3 instance.
- **Paste FX 1/2** pastes all effect parameters from the clipboard into effect bus 1/2.
- **Initialize Patch** resets the current patch to default settings. When designing new patches from scratch, it is useful to start with a "minimal" patch, i.e. a patch with an empty modulation matrix, all effects turned off, etc., which is what the Initialize Patch function does. Note that your current patch will be lost, so be sure to save it first if necessary.



- **Revert Patch** allows you to restore the current patch from disk, in case you are unhappy with changes made to the patch.
- **Undo** allows you to go back to previous edit states. For instance if you accidentally switch the current patch during editing, simply click Undo to restore the previous patch you have been working on.
- **Redo** reverses the Undo.

## Multithreading

DUNE 3 can use up to six processor cores simultaneously.

Enabling multithreading is recommended in most cases, as it can significantly reduce CPU usage. Depending on the computer and the host sequencer used, multithreading may have no effect or even generate CPU spikes. In this case, multithreading should either be disabled or the latency should be increased.

## Skin

Click on Skin to switch to a different skin size. The default size is intended for typical Full HD displays. Use the Large or even Huge size for UltraHD displays. If you work on an older PC with a low-resolution display, choose the Small skin.

## 3.2. Settings

The settings tab controls velocity, pitch bend and the modulation rate.

### Velocity

Each MIDI note transmits a velocity value, which can be connected to any sound parameter using the modulation matrix. Since velocity is often used to control the volume or brightness of a sound, the velocity section allows for a few common direct connections.

- **Volume** adjusts how much velocity affects the volume of a patch. Set to zero, velocity has no effect on volume. Positive values cause higher velocities to increase volume, negative values invert the effect – higher velocities decrease volume.
- **Pan** adjusts how much velocity affects the stereo position. Set to zero, velocity has no effect on pan. Positive values move the sound to the right as velocity increases, negative values do the opposite.
- **Filter** adjusts how much velocity affects the filter cutoff. Set to zero, velocity has no effect on the filter. Positive values cause higher velocities to increase filter cutoff, negative values invert the effect.
- **Env Amt** adjusts how much velocity affects the envelope amount. Set to zero, velocity has no effect on the filter envelope amount. Positive values cause higher velocities to increase envelope amount, negative values invert the effect.

### Velocity Curve

Using the velocity curve setting, you can change the effect the MIDI Velocity has on the selected patch. To change the velocity curve, left-click on the curve, then drag the mouse up or down in vertical direction.

By default, all MIDI velocity values control the destination in a linear manner (1:1). For example, an incoming velocity value of 80 has twice the effect compared to a velocity of 40.

By setting the curve parameter, you can change this 1:1 mapping to a nonlinear behavior. A value of zero corresponds to the default, linear scale.

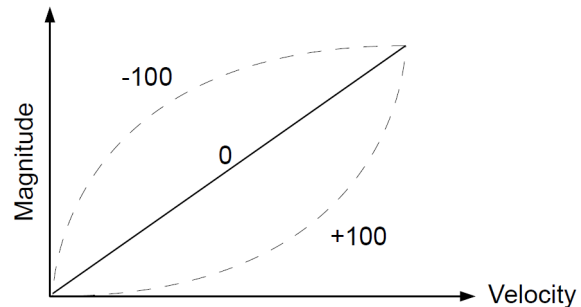
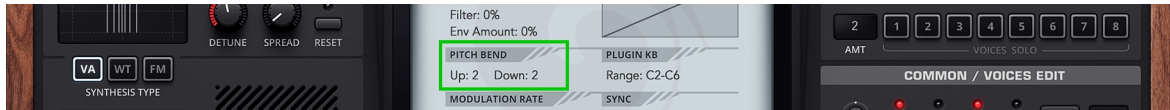


Figure 3.2. Velocity Curve.

The curve parameters is useful to fine tune the velocity response towards your personal playing style, as well as to your MIDI keyboard.

## Pitch Bend

The Bend up/down parameters specify by how many semitones a sound is pitch shifted up or down, when the pitch bend wheel is turned all the way up or down.



Click on the numbers and drag the mouse up or down to increase/decrease the pitch bend range.

If a MIDI keyboard is hooked up to your computer, turning the physical pitch bend or modulation wheel should turn the same wheel in DUNE automatically. If this is not the case, then the required MIDI information is not transmitted to the plugin, and you should refer to your host software's manual to address the issue.

## Modulation rate

The modulation rate determines how frequently the modulation matrix is evaluated.

- The **Normal** rate is almost always sufficient, and the best choice for the vast majority of sounds.
- The **(Very) Fast** rate is useful when you use rapid LFO modulations (>100 Hz), or when you use MSEGs with very fast envelopes.
- **Audio Rate** mode processes the entire synth engine sample by sample. This allows to e.g. use oscillators as modulation sources and modulate any (!) sound parameter with them accurately.

Note that audio rate modulation is very CPU-intensive and should be only used when really needed. Also note that a higher modulation rate does not equal better sound – if you cannot hear a difference, do not use a higher modulation rate, this will just waste precious CPU cycles.

### 3.3. Modulation Matrix

The modulation matrix is accessed by clicking on the Mod Matrix button (see fig. 3.3). The purpose of a modulation matrix is to connect MIDI controllers, LFOs, and envelopes with DUNE's sound parameters. In a nutshell, the modulation matrix is largely responsible for making sounds come alive.

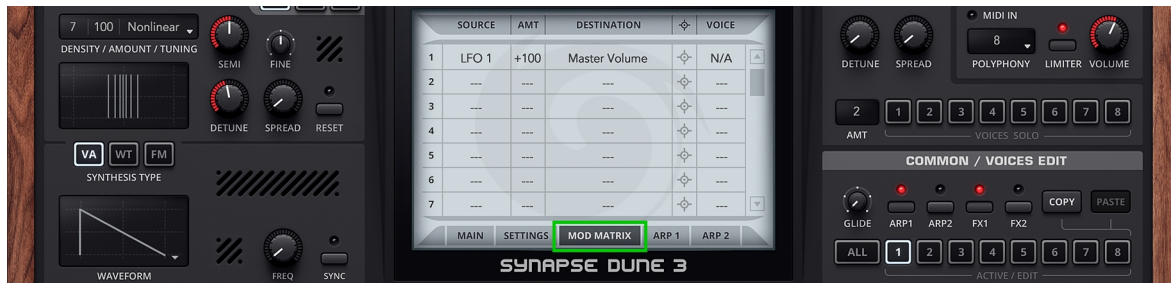


Figure 3.3. Mod Matrix view.

Click anywhere in the source column to choose the modulation source, and anywhere in the destination column to select the sound parameter to modulate. In the amount column, click and drag vertically to change the modulation amount for a specific slot. In the example above (fig. 3.3), LFO-1 modulates the amplitude with an amount of 100%, giving a tremolo effect.

The voice column on the right allows to limit the modulation to specific unison voices. By default, all voices are modulated, the column is empty ("---").

Clicking on the SOURCE or DESTINATION label sorts the modulation matrix. To clear the modulation matrix, right-click and choose "Initialize".

To get the most out of the modulation matrix, it is required to first understand how LFOs and envelopes work, and what the sound parameters do. Hence, the modulation matrix with all its sources and destinations is described more thoroughly in chapter 8. *Modulation Matrix*.

### 3.4. Arpeggiator

An arpeggiator (short: ARP) is a module that generates melodic or rhythmic patterns from one or more keys pressed. This is achieved with the help of a little step sequencer containing note length, velocity and pitch information. For additional flexibility, standard MIDI files can be loaded as well.

DUNE 3 features two Arp modules that can be used independently of each other. The arpeggiator parameters can be accessed by clicking on ARP-1 or ARP-2 in the monitor.

The arpeggiator is enabled or disabled using the ARP-1 or ARP-2 switch in the voice edit section, individually per unison voice.

#### Type

The arpeggiator can use either the built-in step sequencer or midi files as a basis, which can be chosen using the TYPE popup field in the top left of the monitor.

In Step Sequencer mode, monophonic patterns of up to 32 steps can be programmed directly within DUNE. Click on buttons 1-8, 9-16, 17-24 or 25-32 to edit eight steps simultaneously.

Each step has three parameters, note, velocity, and tie. A nonzero note value transposes a step up or down by the given number of semitones, e.g. +12 corresponds to one octave up. The tie button glues two steps, which allows to either slide from one note to another seamlessly, or to simply increase its length. A rest can be programmed by turning down the note value all the way down, until three dashes ("---") appear.

The Midi mode allows to use standard MIDI files as a basis. The files should contain only a single track, and they should be monophonic. An exception is when the "Playback" mode is chosen, in this mode polyphonic MIDI files can be played back too.

## Arpeggiator Modes

The arpeggiator module in DUNE 3 comprises 9 different modes described below.

- **Up** mode successively triggers notes for all keys currently pressed, from the lowest to the highest note, then repeats the sequence starting with the lowest note again.
- **Down** mode does the same, but in reverse order.
- **Up/Down** mode triggers all keys pressed from the lowest to the highest note, then back.
- **Down/Up** works like Up/Down, except in reverse order.
- **Alt Up** is an alternative to Up mode, employing a different pattern when moving up.
- **Alt Down** mode is identical to Alt Up, except it operates in reverse order.
- **Random** mode traverses through all keys pressed, in random order.
- **Chord** mode chops up a chord into a rhythmic pattern, according to what is programmed in the step sequencer or MIDI file.
- **Playback** mode simply plays back the sequencer notes (or MIDI file), transposed according to the current MIDI key pressed. Pressing more than one key simultaneously has no effect in this mode. Playback mode is the only mode that allows to play back polyphonic MIDI files.
- **Silent** is a special mode in which the arpeggiator does not trigger any notes. The purpose of this mode is to use the arpeggiator exclusively for modulation, by choosing "Arp Note", "Arp Velocity" or both as a source in the modulation matrix.

## Octaves

This parameter specifies how many octaves the arpeggiated sequence spans. Set to one octave, the arpeggiator will use exactly the keys pressed. Set to two octaves, the arpeggiator will use all keys pressed, plus the very same keys one octave higher, etc.

## **Steps/Bars**

When in Step Sequencer mode, this parameter sets the pattern length in steps. When the arpeggiator reaches the end of the pattern, it will automatically jump to the beginning and start over. When MIDI mode is selected, this parameter sets the number of bars to use, up to 32 bars are possible. The arpeggiator loops MIDI sequences, too.

## **Rate**

Adjusts the tempo in which the arpeggiator generates sequences from incoming MIDI notes. When SYNC is enabled, the tempo is slaved to the host sequencer and can be specified in musical intervals such as 8th note, 16th notes, dotted notes (\*), triplets (T) etc. When SYNC is disabled, the rate is specified in Hz.

## **Length**

Using the length parameter, the duration of all notes in the sequencer can be increased or decreased. Turned to the left, the notes get a staccato feel, while the opposite direction yields smoother sounding sequences.

## **Swing**

This control shuffles the position of every other 16th note by the specified amount. This parameter can be used to obtain a typical swing feel with a setting of +33% and above. In the step sequencer, the shuffled notes are located at positions 2, 4, 6, 8, ..., 32.

## **Slide**

This control adjusts the time it takes from the beginning of a slid note to reach its target pitch. In order to get any audible effect, TIE must be used on a note, and at least two different note numbers must be used during the note. Note that Slide only works in Step Sequencer mode, not in MIDI mode.

## 4. Sound Parameters

This section describes how a patch is constructed, the operation of all front panel knobs and switches, the effect section and the arpeggiator.

### 4.1. Patch Structure

The structure of a DUNE 3 patch is shown in fig. 4.1. The block diagram shows the basic working principle of the entire synthesizer without taking into account the modulation matrix (which will be covered in detail in chapter 8. *Modulation Matrix*).

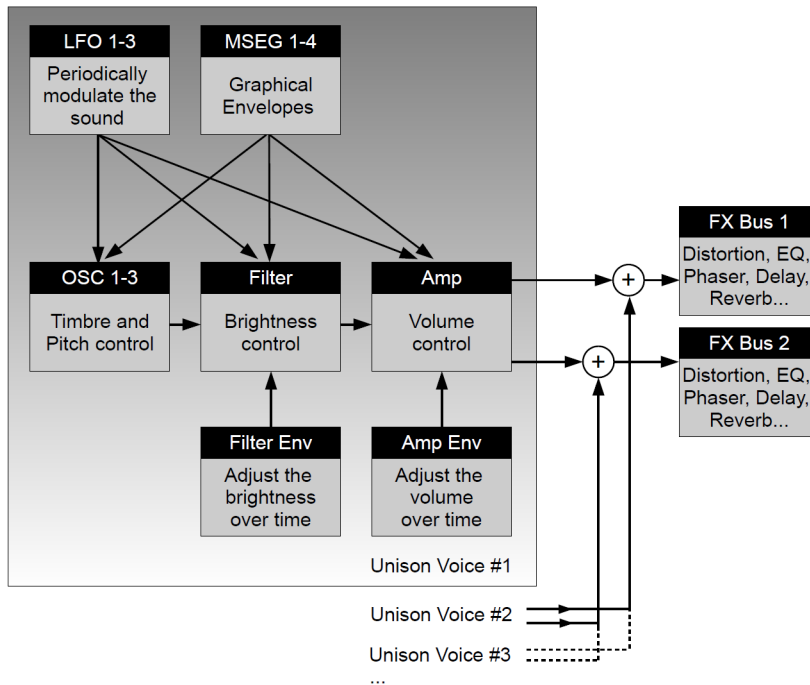


Figure 4.1. Structure of a DUNE 3 Patch.



Whenever a MIDI note is played, one or more voices are triggered to synthesize that note. Each voice has the exact same structure shown, but may use different parameters. The voices are summed and fed into the effects unit to further refine the sound with eq, delay, reverb etc.

Each voice comprises three major building blocks, an oscillator block ("OSC 1-3"), a filter block ("Filter") and a volume control block ("Amp"). 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 final block controls the volume of the signal.

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, so called envelopes 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 very important is the filter envelope ("Filter Env"), which dynamically varies the brightness and thus the timbre over time. In addition to these two envelopes, DUNE 3 offers a further four freely programmable graphical envelopes ("MSEG"), which can be linked to any sound parameter.

While envelopes nicely control the overall progression of a sound, it is sometimes desirable to add periodic modulations. Such modulations can mimic vibrato or tremolo effects known from acoustic instruments, and can be added by using one or more of the low frequency modulation ("LFO") blocks.

## 4.2. Oscillators 1+2

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. 4.2.

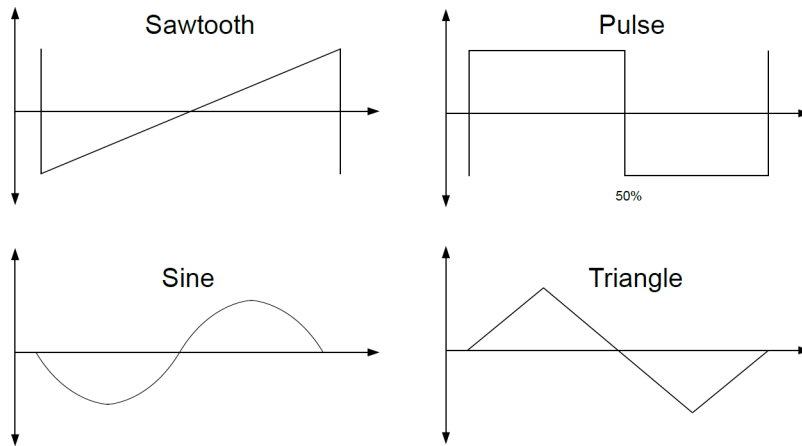


Figure 4.2. Basic oscillator waveforms.

DUNE 3's oscillator controls can be found in the top left section of the user interface:



DUNE 3 offers three oscillators and a separate noise generator. This corresponds to how many traditional synthesizers work. In DUNE 3, however, the first two oscillators are actually stacks of up to 32 oscillators each, with adjustable detune and stereo spread. This allows to obtain thick pad, bass or lead sounds with just one oscillator alone. With DUNE 3's 8 unison voices, up to 200 oscillators per key can be synthesized.

### 4.2.1. Common parameters

#### Density

The number of oscillators can be changed by left-click/drag on the Density field in the top-left. When set to zero, the oscillator stack is turned off, which saves processor time.

#### Amount

When choosing more than two oscillators using the density control, the amount (AMT) parameter allows to modify the level of the oscillators. The meaning of this parameter depends on the selected tuning mode, but usually adjusts the level of the oscillators around the center (which always remains at maximum).

#### Tuning

The chosen tuning mode affects the overall tuning and volume of the oscillator stack.

- **Linear** mode tunes all oscillators around the center with equal spacing.
- **Nonlinear** mode moves some oscillators closer to the center.
- **Gaussian** mode uses a Gaussian distribution on the volume of the oscillators, which can be fine-tuned by the amount parameter.
- **Alternate** mode lowers every other oscillator in volume when changing the amount parameter.
- **Random** uses a random tuning for all oscillators, whenever a new key is pressed.
- **Perfect 5th** is like Linear, except that every other oscillator is pitched up by +7 semitones (a fifth).
- **Minor** is like Linear, except that a minor chord is generated with 4 oscillators or more.
- **Major** is like Linear, except that a major chord is generated with 4 oscillators or more.

- **Sub Osc** is like Linear, except that every other oscillator is pitched down by -12 semitones (one octave).
- **Swarm** is a new oscillator stack model in DUNE 3, where all oscillators in the stack are individually modulated. This mode adds an extra knob Rate, controlling the rate of modulation.

### **Semi**

The control labeled SEMI adjust the coarse tuning of the oscillator stack, in semitones. The range spans +/- 36 semitones. A larger range can be obtained by using the modulation matrix, if required. This will be covered in chapter 8. *Modulation Matrix*.

### **Fine**

This parameter adjusts the fine tuning of the oscillator stack. A value of +/- 100 corresponds to half a semitone.

### **Detune**

When two or more oscillators are active in the stack, the detune parameter allows to spread their pitch around the center frequency.

### **Spread**

When two or more oscillators are active in the stack, the SPRD knob allows to spread the oscillators in the stereo field, around the center. Turned fully left, the oscillator stack will be mono.

### **Reset**

When reset is enabled, all oscillators in the stack start at the same initial phase (by default zero, this can be changed in the modulation matrix).

Setting the initial oscillator phase can be useful to obtain better control of the transient of a sound, for instance. When using more than one or two oscillators, note however that reset will create strong phasing effects. It is thus usually better to keep reset turned off when using multiple oscillators.

#### 4.2.2. Synthesis models

Three different synthesis models, Virtual-Analog (VA), Wavetable (WT) and Frequency Modulation (FM) are available per oscillator. When using more than one voice, it is furthermore possible to specify different modes for different voices. The synthesis model can be changed by using one of the three buttons (VA/WT/FM):



#### VA mode

The virtual-analog (VA) mode synthesizes three basic waveforms- sawtooth, pulse and triangle. Click on the waveform drawing to switch between the three different types. When the pulse waveform is chosen, you may additionally adjust its pulse width (see fig. 4.2.2). The default is 50%, corresponding to a square wave.

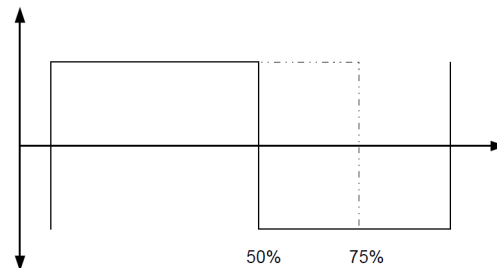


Figure 4.2.2. Pulse Width.

All three waveforms offer oscillator synchronisation, which is enabled by pressing the SYNC switch. Oscillator sync causes an oscillator to get periodically reset to zero phase, using a second "master" oscillator. Whenever the master oscillator has completed a cycle, it resets the "slave" oscillator. If the master oscillator has a lower frequency than the slave, the result is a new timbre, as the slave oscillator is not always able to complete a full cycle. In contrast to traditional synthesizers, no extra oscillator is needed to perform oscillator sync in DUNE 3. When sync is enabled, DUNE 3 automatically generates a virtual master oscillator needed to perform the synchronisation. The frequency of this oscillator is adjusted by the frequency knob (FREQ), relative to the given pitch. The sync frequency can be modulated using the modulation matrix.



### WT mode

The Wavetable (WT) mode allows to choose an arbitrary periodic waveform from a wavetable. The waveform is selected by clicking on the waveform drawing in the same manner as in VA mode. Another way of setting the waveform is by changing the position knob. The position knob traverses the entire wavetable and allows to specify positions in between two waveforms for yet more versatility. The waveform position can also be modulated via the modulation matrix, to obtain complex, dynamic timbres with constantly changing waveforms. If the waveform position is modulated, the transitions can be controlled from hard (0%) to smooth (100%) using the Interpolation knob. By clicking on the wavetable popup, different wavetables can be selected. Wavetables can have a varying amount of waveforms stored in them, between 1 and 256 waveforms. Note that you can select different wavetables for different unison voices.



### FM mode

Frequency modulation (FM) uses three sine waves as the basis for sound generation, which are usually referred to as operators. Two operators "A" and "B" change the frequency of the operator "C".



To create interesting sounds, it is important to change the amplitude of operators "A" and "B" over time. This can be achieved by combining one or more MSEG envelopes with the modulation targets "FM Amt A" and "FM Amt B". An envelope falling to zero with the target A or B produces sounds that begin with a bright timbre and then become increasingly darker, all the way to a pure sine wave.

FM synthesis can produce a variety of interesting sounds, especially bell-like sounds and synthetic piano sounds.

### **4.3. Oscillator 3**

The third oscillator generates a simple periodic waveform, which can be chosen by clicking on the waveform display.

#### **Semi**

The control labeled SEMI adjust the coarse tuning of the third oscillator, in semitones. The range spans +/- 36 semitones. A larger range can be obtained by using the modulation matrix, if required. This will be covered in chapter 8. *Modulation Matrix*.

#### **Fine**

This parameter adjusts the fine tuning of the oscillator stack. A value of +/- 100 corresponds to half a semitone.

#### **Reset**

When reset is enabled, the oscillator will always start at the zero phase when a new key is pressed. When disabled, the oscillator starts with a random initial phase. A specific phase angle to start with can be set within the modulation matrix.

## 4.4. Noise Generator

All three oscillators generate periodic waveforms with a certain pitch. Sometimes it is useful, however, to spice up a sound with a random element that has no fixed pitch. This can be useful to synthesize percussive sounds, to recreate the behavior of wind or plucked string instruments during transients or to synthesize nature sounds such as fire, water or wind. To facilitate synthesizing such sounds, DUNE 3 offers a noise generator.

### Lowpass

The white noise emitted by the noise generator passes through a low-pass and high-pass filter, to further shape the timbre of the noise. The LP knob controls the cutoff frequency of the low-pass filter. Lower values cause a darker timbre.

### Highpass

The HP knob controls the cutoff frequency of the high-pass filter. When set to zero it has no audible effect, at higher values the sound becomes increasingly thin and bright.

### Width

Since the noise generator operates in stereo, it can emit noise for the left and right channels separately. The width parameter adjusts the stereo width – set to minimum the noise appears monophonic, set to maximum both channels have independent noise sources, resulting in a wide stereo image.

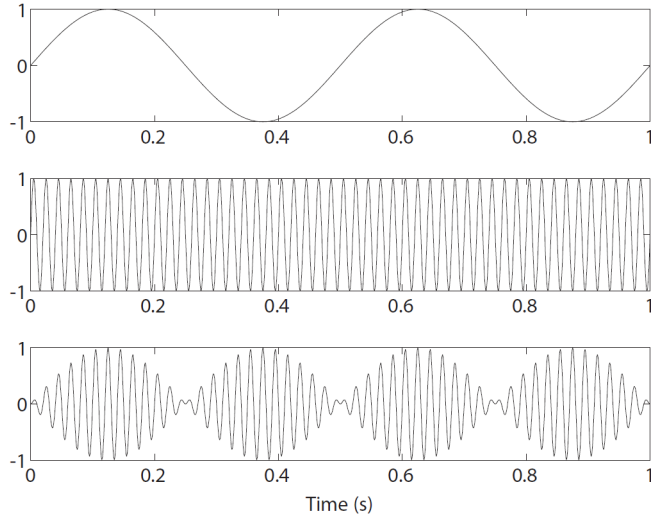
### White/Pink

The most commonly used noise source is White noise, which is the default option. White noise has a flat frequency spectrum, i.e. all frequencies are contained equally. In some situations Pink Noise may be preferable, which rolls off with 3 dB / octave, so high frequencies are attenuated somewhat.



## 4.5. Ring Modulator

A further interesting effect that can be applied to the oscillator stacks 1 and 2 is to multiply them with each other. This can be seen as one oscillator modulating the other in its amplitude:



The effect depth can be controlled using the RM fader in the mixer section. Set to zero, the ring modulator output is disabled.

Mathematically, the result of ring modulation is that the sums and differences of both signal's frequencies are generated. If the oscillators are detuned, this will lead to very inharmonic, metallic sounds.

## 4.6. Mixer

The mixer section allows to control the level and panorama of oscillators 1-3, the ring modulator and the noise generator, before they enter the filter.



All mixer parameters can be controlled via the modulation matrix too. This allows to fade in or out an oscillator, cross-fade between oscillators, modulate the panorama position, etc.

#### 4.7. Filter Section

The raw sound coming from the oscillator sections is typically too bright to be useful. While many natural instruments like a flute or guitar start with a short, bright transient, they decay quickly to a much darker timbre. This behaviour can be modelled by using a time-varying filter. The filter section is located in the middle of the user interface below the monitor.



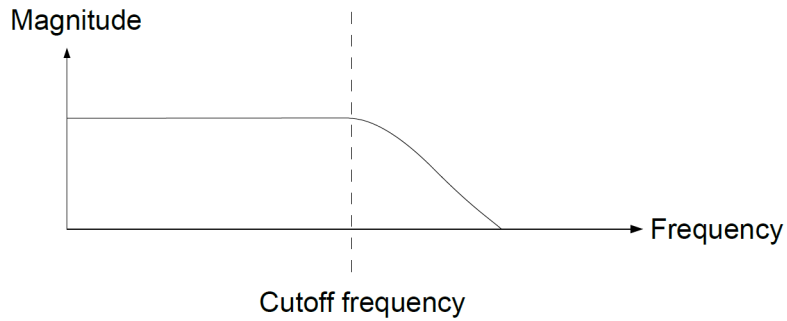
DUNE 3 features two identical filters and an insert effect. The two filters can be set to an arbitrary mode and have their own set of parameters each, for maximum flexibility. The filters, as well as the insert effect, can be routed in six different ways. The routing options will be discussed in the ROUTE section later on.

### 4.7.1. Filter Parameters

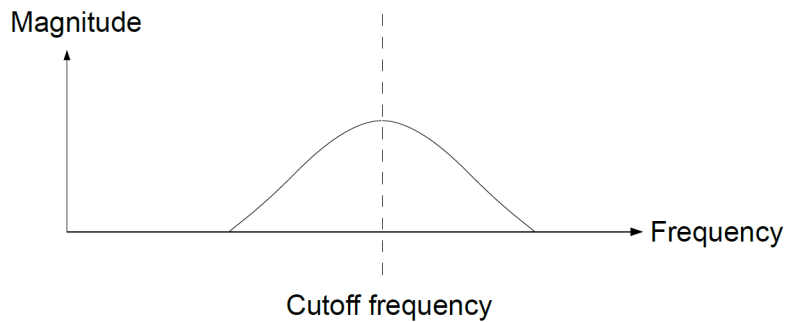
#### Cutoff

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:

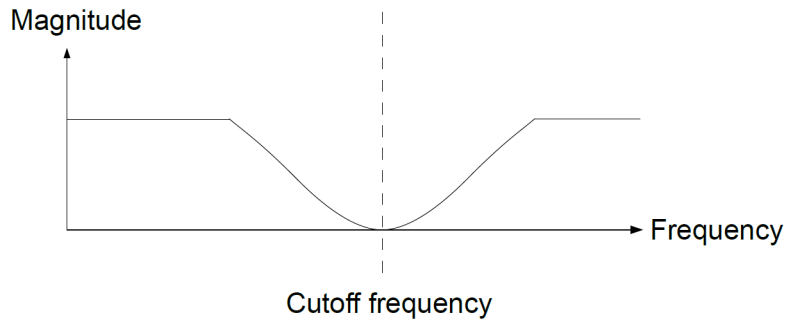
- For low-pass (LP) filter types, frequencies above the cutoff frequency are damped:



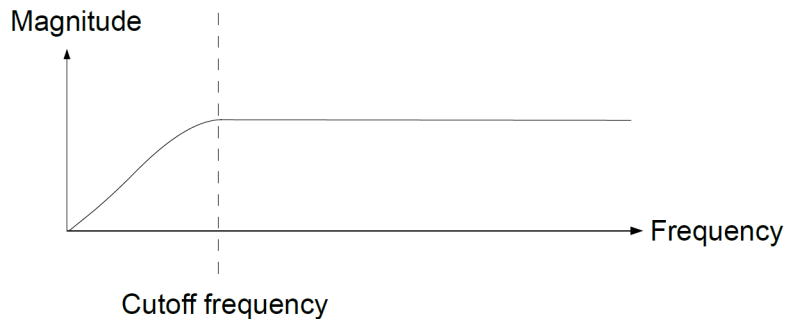
- A band-pass (BP) filter damps frequencies around the cutoff frequency. As a result, bass and treble get attenuated:



- A notch filter rejects frequencies around the cutoff frequency and passes everything else:



- A high-pass (HP) filter attenuates all frequencies below the cutoff frequency and passes the higher frequencies unchanged:

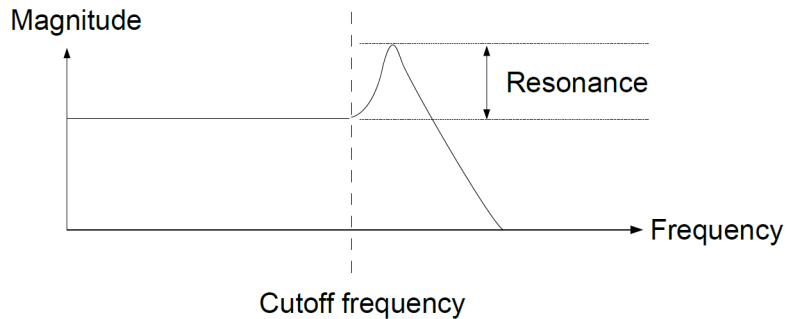


The filters described above form the basic filters encountered in most synthesizers. The low-pass filter is the most common, as it fully preserves the bass frequencies and allows the natural progression from a bright to a dark timbre when being modulated.

To modulate the cutoff frequency and produce a dynamically changing timbre, the MSEGs, LFOs and the filter envelope can be used. The most common way to control cutoff is the filter envelope, discussed later in this chapter.

## Reso

If the output of a filter is fed back to its input, resonance occurs, which is a sinusoidal oscillation near the cutoff frequency:



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 sinusoidal oscillation gets strong enough to use the filter in a similar fashion as an oscillator. This property can be useful to create special effect sounds such as laser guns, electronic bass drums etc.

## Drive

The DRIVE knob sets the input gain of the filter. When using strong resonance, drive controls the mix between the filtered signal and the resonant peak. The lower the drive setting, the stronger the resonant peak. When using any of the transistor ladder filter models (see below), drive will additionally saturate the signal, regardless of resonance.

## Keytrack

The KTRK knob determines how much the cutoff frequency is affected by the MIDI key number. 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.

At low settings, this parameter is useful to create subtle timbre variations when different notes are played. At higher settings, key tracking can be used to simulate the properties of acoustic instruments that have a varying timbre dependent on the note played.

#### 4.7.2. Filter Modes

DUNE 3 features a number of different filter models to choose from. All filters except the legacy DUNE 1 filters are zero-delay feedback designs, which more closely resemble the response of analog filters. Furthermore, most filters can reach self-oscillation when turning up the resonance.

##### Clean Multi-Mode

The clean filter models resemble analog filters with a largely clean direct path, such as OTA-based ladder filters. Only the feedback path contains mild saturation, in order to prevent the resonance from ramping to infinity at high resonant levels. The clean nature of the filters and their moderate CPU usage makes them an excellent choice for many sounds.

- **CL Lowpass 12 dB**

A two-pole lowpass filter, with 12 dB attenuation per octave above the cutoff frequency.

- **CL Lowpass 18 dB**

A three-pole lowpass filter, with 18 dB attenuation per octave above the cutoff frequency.

- **CL Lowpass 24 dB**

A four-pole lowpass filter, with 24 dB attenuation per octave above the cutoff frequency.

- **CL Bandpass 12 dB**

A two-pole bandpass filter, with 6 dB attenuation per octave around the cutoff frequency.

- **CL Bandpass 24 dB**

A four-pole bandpass filter with 12 dB attenuation per octave around the cutoff frequency.

- **CL Highpass 12 dB**

A two-pole highpass filter, with 12 dB attenuation per octave below the cutoff frequency.

- **CL Highpass 24 dB**

A four-pole highpass filter, with 24 dB attenuation per octave below the cutoff frequency.

- **CL Notch**

A four-pole band-reject filter (Notch) filter.

- **CL Deep Notch**

Like CL Notch, but produces deeper notches in the frequency spectrum.

### **Expander**

The Expander filters are analog-inspired lowpass filters. In contrast to the Clean Multi-Mode filters, the Expander filters have saturation in both the direct signal path as well as the feedback path.

- **XP Lowpass 12 dB**

A two-pole lowpass filter, with 12 dB attenuation per octave above the cutoff frequency.

- **XP Lowpass 24 dB**

A four-pole lowpass filter, with 24 dB attenuation per octave above the cutoff frequency.

### **Comb Filter**

A comb filter is a special filter that uses a delay to create a series of notches in the frequency response. This creates a special timbre that can be adjusted via the cutoff knob.

Since the comb filter does not remove high frequencies, it is often useful to add a low-pass filter as a second filter in series following the comb filter. Using keytracking, the comb filter is also tunable.

This feature is very useful and can be used to keep the timbre stable (just set the filter keytracking to 100%). The delay in the comb filter also has a feedback path. This allows for resonance, which can be adjusted using the Resonance knob. Be careful when using very high resonance values close to 100%, the sound can get very loud here.

## **Analog-Modeled**

The analog-modeled filters are the most accurate reproductions of typical analog filters in DUNE. With the exception of the Alpha filter, the analog-modeled filters use 3x oversampling to improve their accuracy and feature multiple non-linearities. Thus, they are computationally more expensive than the other filters in DUNE. Note that for many patches the CL or XP filter types above are fully sufficient, so some care should be taken when to use the analog-modeled filters.

- **Alpha 24 dB**

A four-pole lowpass filter based on a state-variable design, with 24 dB attenuation per octave above the cutoff frequency.

- **Polaris 24 dB**

A four-pole lowpass filter with 24 dB attenuation per octave, inspired by the CEM 3372 chip. A unique sounding chip which was put only into a handful of hardware synthesizers.

- **Pro 12 dB**

Enhanced version of the Transistor ladder filter, with 12 dB attenuation per octave.

- **Pro 24 dB**

Enhanced version of the Transistor ladder filter, with 24 dB attenuation per octave.

- **Saturn 24 dB**

A four-pole filter based on a frequently used custom OTA chip. This filter type can be used to simulate a wide range of vintage-analog synthesizers.



## Brickwall

The Brickwall filters in DUNE 3 are special filters featuring a very steep slope. This can be useful to filter out unwanted frequencies in the bass or treble part of the spectrum.

- **BW Lowpass 36 dB**

A six-pole lowpass filter.

- **BW Lowpass 60 dB**

A ten-pole lowpass filter.

- **BW Highpass 36 dB**

A six-pole highpass filter.

- **BW Highpass 60 dB**

A ten-pole highpass filter.

## Transistor Ladder

The transistor ladder filters are exclusively low-pass types and resemble analog transistor-based designs. The key feature of those filters is that they contain saturation in all stages, thus coloring incoming signals even without any resonance applied.

The transistor filters require substantially more CPU than the multi-mode filters.

- **TR Lowpass 12 dB**

A two-pole lowpass filter, with 12 dB attenuation per octave above the cutoff frequency.

- **TR Lowpass 18 dB**

A three-pole lowpass filter, with 18 dB attenuation per octave above the cutoff frequency.

- **TR Lowpass 24 dB**

A four-pole lowpass filter, with 24 dB attenuation per octave above the cutoff frequency.

- **TR Lowpass 30 dB**

A five-pole lowpass filter, with 30 dB attenuation per octave above the cutoff frequency.

### **Acid Lowpass**

This special filter mimics a four-pole transistor network with unbuffered filter stages. The result is a rather dark timbre, unlike any of the other filter types.

### **Sallen-Key**

The Sallen-Key filters are two-pole, multi-mode filter designs. What makes them special is the sound of the resonance. It features strong distortion, making for a very aggressive sound.

- **SK Lowpass 12 dB**

A two-pole lowpass filter, with 12 dB attenuation per octave above the cutoff frequency.

- **SK Bandpass 12 dB**

A two-pole bandpass filter, with 6 dB attenuation per octave on either side around the cutoff

- **SK Highpass 12 dB**

A two-pole highpass filter, with 12 dB attenuation per octave below the cutoff frequency.

### **Dune**

DUNE 3 incorporates two filter types from DUNE 1 ("Lowpass 12dB" and "Lowpass 24dB"), to facilitate creating patches that sound similar to DUNE 1 patches.

- **Dune 12 dB**

A two-pole low-pass filter, with 12 dB attenuation per octave above the cutoff frequency.

- **Dune 24 dB**

A four-pole low-pass filter, with 24 dB attenuation per octave below the cutoff frequency.

#### **4.7.3. Balance**

The Balance knob blends seamlessly between both filters.

If the knob is turned to the left, only Filter 1 is audible. In the middle position, both filters are audible. Turned fully clockwise, only Filter 2 is active.

#### **4.7.4. Link**

If the Link switch is enabled, the parameters of Filter 1 and Filter 2 can be controlled simultaneously using the Filter 1 knobs. Thus when turning e.g. the Filter 1 Cutoff knob, the Filter 2 Cutoff knob will move as well. Note that the Filter 2 controls keep their relative distance to Filter 1 prior to touching the knob (if there is any). This method of linking filters allows for a very flexible workflow.

#### **4.7.5. Insert Effect**

Insert effect section allows to perform processing in addition to the two main filters.

- **Light Dist**

Performs a mild distortion on the input signal. The input gain is adjustable using the AMOUNT knob, and determines the amount of distortion. The output of the distortion stage passes through a low-pass filter with adjustable cutoff frequency. Both the amount of distortion and the low-pass filter cutoff frequency can be modulated via the modulation matrix.

- **Hard Dist**

A stronger distortion effect than the light distortion, with more gain and a different timbre.

- **Foldbk Soft**

A foldback distortion with a smoother slope and less artifacts, making it a good alternative to the light & hard distortion types.

- **Foldbk Hard**

Foldback is a special kind of distortion, where the signal is distorted along a bipolar curve such as a triangle or sine wave. The result is a rather extreme distortion. It's sound is often very digital and creates strong artifacts, so this effect should be used with some care.

- **Bitcrush**

The bitcrush effect reduces the dynamic range of the signal. The amount of reduction is controlled using the AMOUNT knob. If the signal range is reduced to 2 bits, for instance, each sample passing through the filter is reduced to only 4 different states. Strong reduction makes the result sound somewhat similar to a square wave, and creates very strong digital artifacts. To tame the artifacts a little, the lowpass filter on the output can be used to roll off high frequencies.

- **Ratecrush**

Ratecrush is another effect which creates strong digital artifacts. Rather than reducing the dynamic range, Ratecrush employs a sample-and-hold circuit to reduce the sample rate.

- **Halfrect**

The half-wave rectifier effect nulls the negative half-wave of the incoming signal. Note that the signal level should be 0 dB for this effect to work best, otherwise a temporary DC offset may occur which results in a short audible "thump" noise when pressing a key.

- **Fullrect**

The full-wave rectifier effect mirrors the negative half-wave of the incoming signal, such that it becomes positive.

Note that the signal level should be 0 dB for this effect to work best, otherwise a temporary DC offset may occur which results in a short audible "thump" noise when pressing a key.

- **Phaser**

A four-stage phaser effect, creating two notches in the frequency spectrum. The AMT knob controls the frequency of the notches, the second knob the feedback in +/- 100%. A classic sweeping phaser effect can be obtained by modulating the AMT knob position from the modulation matrix.

- **Highpass 1p**

A one-pole highpass filter with 6 dB attenuation per octave.

- **Highpass 2p**

A two-pole highpass filter with 12 dB/oct attenuation.

- **Lowpass 1p**

A basic one-pole lowpass filter with 6 dB attenuation per octave. The cutoff frequency is controlled via the AMOUNT knob, and can be modulated via the modulation matrix. The extra filter can be used to fine-tune the response of the main filter, for example a 2-pole bandpass filter could be turned into a 3-pole filter. Another option is to apply different modulation to the main filter and the extra filter.

- **Lowpass 2p**

A two-pole lowpass filter with 12 dB attenuation per octave.

- **Lowcut+KT**

A low-cut filter with keytracking, i.e. the filter will adjust its cutoff frequency according to the current key pressed. This filter type is useful to shape the first partials of a sound, in particular the fundamental.

- **Notch**

A two-pole notch filter, where the AMT knob controls the frequency of the notch.

- **Comb**

The Comb filter creates a plurality of notches in the frequency spectrum. The spacing between notches can be controlled via the AMT knob.

- **Formant**

Using the AMT knob, the Formant filter blends between the two vowels "A" and "O".

- **Vowel**

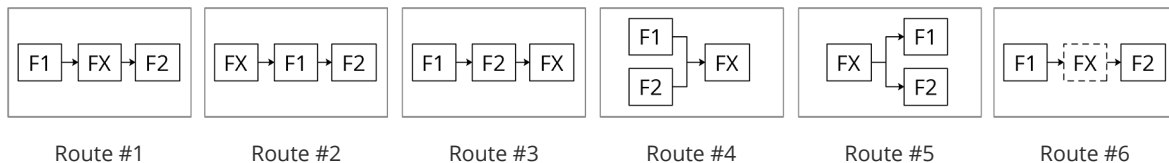
The Vowel filter sweeps between all vowels (A,E,I,O,U) by using the AMT knob. Interesting effects can be obtained by modulating the AMT knob position from the modulation matrix.

#### 4.7.6. Filter Route

The order in which the two filters, as well as the insert effect is processed, is determined by the ROUTE pop-up.

Note that the filter routing has a strong effect not only on the sound, but also on the behavior of the Filter Balance knob described above.

The following filter route setups are possible in DUNE 3:



## 4.8. Filter Envelope

An envelope controls a sound parameter over time, starting from the instant a key is pressed (see fig. 4.8). The filter envelope is designed specifically to control the filter cutoff frequency, but can be used to control other parameters too (via the modulation matrix, see chapter 8. *Modulation Matrix*).

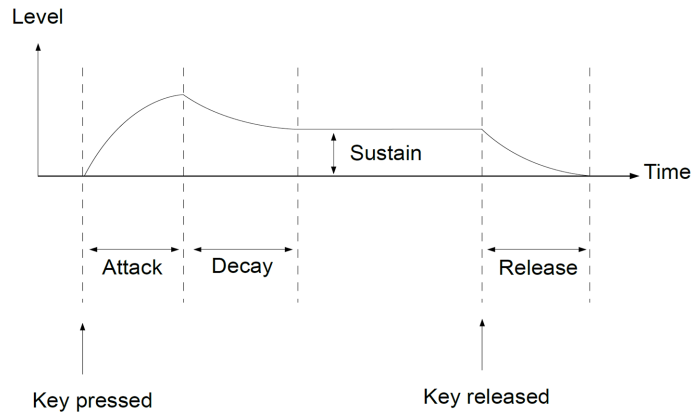


Figure 4.8. The filter envelope.

### Amount

The AMT knob controls how much the filter envelope 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. Note that when a dual filter is used, the filter envelope affects both filters simultaneously.

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.

### **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 the sustain level is set to zero, the release parameter may have no effect if the envelope has previously reached zero already.

### **Analog**

By default, DUNE 3 employs standard envelopes common to most soft- or hardware synthesizers. The analog switch can be used to create more punchy envelopes, typical for some vintage-analog synthesizers. The effect is particularly noticeable for short attack times.



## 4.9. Amplitude Envelope

Located right below the filter envelope, the amplitude envelope controls the progression of the volume of a sound (see fig. 4.9). It works in the same manner as the filter envelope.

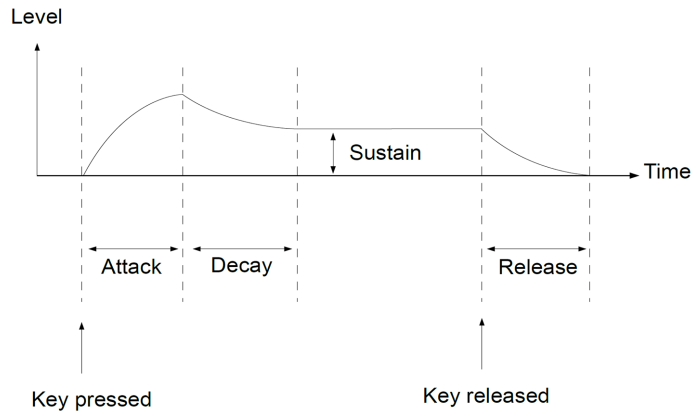


Figure 4.9. The amplitude envelope.

### Attack

The ATTACK parameter specifies the duration it takes for the amplitude envelope to go from zero to its maximum level.

### 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.

## **Level**

Controls the level of the amplitude envelope, which simply boils down to the overall level of the current voice. If you wish to modulate the volume of a patch, this is usually the right destination parameter to choose.

## **4.10. Unison Voices**

The "Unison Voices" section is located in the upper right corner of the user interface between the monitor and the master area.

DUNE 3 can fire multiple voices simultaneously for each note played. Using multiple voices instead of just one can produce thick and spacious sounds, another application is to mix completely different sounds.

In order to use several voices sensibly, these voices must differ from each other. If you just want the sound to become wider or fatter, simply increase the number of Unison voices then use the Detune or Spread parameters to widen the sound.

A more sophisticated way of working with Unison voices is to directly edit the Unison voices, by selecting a voice in the Voice Edit section and then making changes to the sound parameters. This way of working with Unison voices allows to stack completely different sounds.

## **AMT**

This knob controls the number of voices to use for each note played.

When set to two voices, for instance, each key press will trigger two unison voices. Note that the CPU usage doubles as the number of unison voices is doubled, since two voices per key mean twice the processing. In contrast to DUNE 1, the number of unison voices does not affect the polyphony in any way – so two unison voices at a polyphony of four means that up to eight voices will be needed. The MAIN monitor displays the current number of active voices, as well as the maximum possible.

### **Detune**

This parameter detunes the unison voices. Higher settings correspond to more variation. A minimum of two voices is necessary to get an audible change. The detuning is centered around the main note pitch. When playing e.g. A4 (440 Hz) and detuning two voices by 1 Hz, the two pitches would be 339 Hz and 441 Hz, thus still creating the sensation of a 440 Hz note.

### **Spread**

Spreads the unison voices in the stereo field. A minimum of two voices must be dialed in to create an audible effect. At maximum position, the voices will be fully spread in the stereo field – in the case of using two voices, one voice will be panned hard left and the other one hard right. In case of three voices, one voice will be panned hard left, one dead center and one hard right, etc.

## **4.11. Common / Voices Edit**

The voice edit section is located just below the unison voice section. This section allows to edit all unison voices combined (ALL), or to select individual voices. Each unison voice has its own set of parameters, which means you can stack up to 8 different sounds. When a specific voice is selected, it can be copied by pressing the COPY button, selecting the destination voice then pressing PASTE.

### **Arp 1**

This button enables or disables the first arpeggiator.

Note that each unison voice can have the arpeggiator enabled or disabled for increased versatility.

### **Arp 2**

This button enables or disables the second arpeggiator.

### **Glide**

The glide knob controls the pitch glide effect for successive notes. Turned fully to the left, there is no glide. Towards higher settings, the pitch glide time increases. Each unison voice can have its own glide time.

### **FX 1**

Enabling this button sends the voice through the first FX Bus.

The FX bus can be enabled or disabled for each voice separately.

### **FX 2**

Enabling this button sends the voice through the second FX Bus.

The FX bus can be enabled or disabled for each voice separately.

### **Solo**

Allows to solo individual voices. Note that only voices that are actually in use can be solo'ed. Pressing solo on the 4th voice of a 3-voice patch will thus lead to silence.

## 4.12. Master

The Master section contains basic performance parameters and allows to set the global volume.

### Polyphony

The polyphony popup menu allows to specify how many notes can be played simultaneously. The main purpose of limiting the polyphony is to limit the CPU usage, as less voices need to be rendered.

A polyphony of 4 means that 4 keys can be held down simultaneously, pressing a fifth key will cut one of the existing voices. Note that when sounds have long release times (e.g. pads, strings), it is a good idea to choose a polyphony of 12 or 16 even when only playing a few notes at a time. Otherwise, old notes get cut which can lead to clicks.

When choosing Mono or Legato, only one note can be played at a time. Legato allows to smoothly go from one note to another without re-triggering the envelopes. This can be useful for bass and lead sounds, particularly in combination with the glide knob. It creates a unique playing feel and sound which can be better for monophonic lines. Using mono or legato modes also results in the smallest CPU usage possible.

### Limiter

DUNE 3 features an optional brick-wall limiter at its output, placed after the Master Volume control. The limiter lowers spurious peaks such that the output signal will never exceed a level of 0 dB. Note that the limiter is a zero-delay type, thus it should be used with care and not driven too hard. Otherwise the limiter may degrade the quality of the output signal.

### Master Volume

Sets the overall volume of the entire synthesizer.

## 4.13. MSEGs

DUNE 3 features four graphical envelopes called MSEG (Multiple Segment Envelope Generator). Graphical envelopes allow precise and customizable adjustment of sound parameters over time, and thus serve as an important modulation source. MSEGs can be looped, which allows to create rhythmic gate effects (for instance the classic "trance" gate sound), or they can act as low-frequency oscillators with custom shapes.



### Using the editor

- Create new points either with the left mouse button (double-click), or with the middle mouse button (single-click).
- Delete points with the right mouse button.
- Move points by clicking on them with the left mouse button, then moving them in any direction.
- Change the curve shape of a segment by clicking on it with the left mouse button, then drag the mouse up or down.

Four buttons are located on the left side of the editor:

- **Preset** opens a factory MSEG preset, or saves a custom user preset for later use.
- **Copy** copies the MSEG into the clipboard.
- **Paste** replaces the current envelope with an envelope stored in the clipboard.

- **Invert** mirrors all points vertically.
- **Reverse** mirrors all points horizontally.
- **Clear** deletes all points in the envelope.

## Mode

MSEGs can be used in four different modes:

- **Note On** starts the MSEG when pressing a key, then traverses the envelope until the last point is reached. Note that MSEGs are polyphonic in this mode when modulating voice parameters (such as pitch), ie each voice keeps its own envelope position.
- **Note Off** triggers the MSEG when releasing a key. MSEGs are polyphonic in this mode too.
- **Key On** mode is identical to Note On, except when the arpeggiator is enabled. Note On will react to every note that is triggered by the arpeggiator, while Key On will only react to actual MIDI keys pressed. Key On mode thus allows modulation of an arpeggiated sequence as a whole, for example the mode can be used to fade in an arpeggiated sequence.
- **Loop** mode periodically loops through the envelope. MSEGs are monophonic in this mode, i.e. all MSEG destinations receive the same signal. This is important when using the envelope for trance gate and other rhythmic effects, synchronized to the song tempo.

## Sync

When the sync switch is off, the envelope operates in seconds. The time line shows seconds, and the maximum length can be adjusted in seconds.

If the sync switch is on, the MSEG operates in sync with the host tempo. The time line is switched to a musical notation in the format "bars.quarters" (e.g. 4.2 means the second quarter note in the fourth bar).

## Rate

Adjusts how fast the envelope is traversed. When the SYNC switch is disabled, the rate knob simply scales the entire envelope from 1/10th to 10 times the duration. Set to center, the envelope time is unaffected by the rate knob – an envelope spanning 1 second takes exactly 1 second to complete.

When sync is on, the envelope duration is scaled with musically meaningful values, in order to keep the sync with the host tempo. Setting the rate knob to half a bar (1/2) will traverse the envelope twice as fast as the default of one bar (1/1), for instance. For more sophisticated effects, Triplet (T) and dotted (\*) values can be dialed in.

The MSEG rate can be modulated via the modulation matrix. Note that this only works when sync is disabled.

## Length

The maximum duration of the envelope. The meaning depends on the rate knob and the sync switch. If sync is disabled and rate set to center, the maximum duration is given in seconds.

## Velocity

With the Velocity knob, the envelope amount can be varied according to the velocity of the MIDI key pressed. At default center position, the velocity has no effect. Set to negative values, low velocities will have a stronger effect than high velocities. Set to positive values, higher velocities increasing the envelope amount.

Making envelope amounts velocity-dependent is particularly important for FM sounds. Low velocities typically correspond to low envelope amounts (resulting in a soft sound), while high velocities typically correspond to high envelope amounts (causing a brighter, more aggressive sound).



## Keytrack

The key track knob allows to make the envelope rate dependent on the MIDI key number pressed. Usually key tracking is set to zero, so the MIDI key number has no effect on the envelope. The higher the key track value, the shorter the envelope when pressing higher keys. The purpose of key tracking is to imitate the behavior of acoustic instruments such as the guitar – high notes on the guitar decay more quickly than low notes. Key tracking requires the sync switch to be turned off, otherwise it has no effect.

## 4.14. LFOs

The three LFOs are accessed by clicking on the LFOs tab.



Using oscillators, the filter and envelopes, it is possible to control the basic properties of a sound, such as timbre, volume and pitch. For bass and percussive sounds this may be enough to get good results, but for pad or lead type sounds, the sustain stage can 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 similar to ordinary oscillators, and usually generate a periodic signal using similar waveforms (see fig. 4.14). LFOs are inaudible, however, 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. DUNE's three LFOs are much more capable than that, however, as almost any sound parameter can be used as a modulation destination. Additionally, LFOs can modulate each other in volume or frequency to obtain yet more interesting variations.

Assigning destinations to the LFOs and adjusting the modulation depth is performed in the modulation matrix, which is covered in the chapter 8. *Modulation Matrix*.

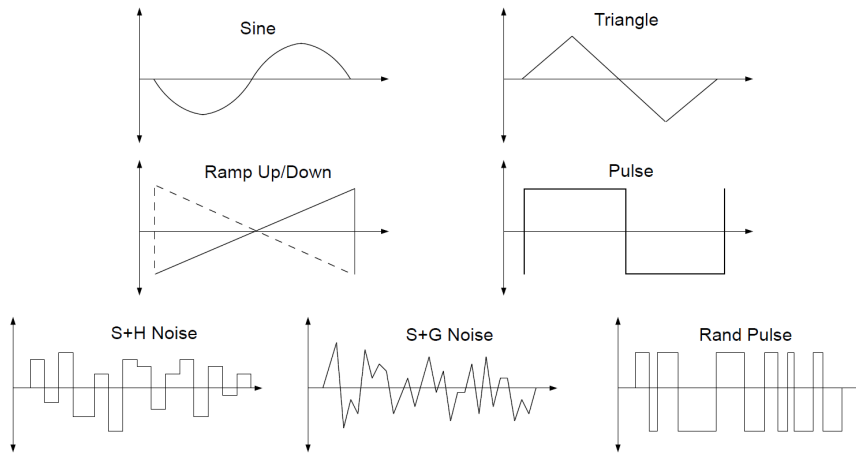


Figure 4.14. LFO waveforms.

## Shape

Clicking on the shape popup menu allows to select one of the waveform shapes depicted in fig. 4.14. Ramp, Pulse, Triangle and Sine are periodic waveforms, while S+H noise (Sample-and-Hold), as well as S+G noise (Sample-and-Glide) are random signals. Random modulation signals are useful for special effects or to simulate the behavior of old analog hardware, by choosing a very slow and subtle modulation.

## 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, eighths 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 eighth 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.

## **Poly**

By default LFOs operate in a global fashion, ie all voice destinations receive the same LFO signal. When the POLY switch is enabled, however, each voice uses its own, local LFO. This allows to set up modulations which always start at zero phase, for instance.

## **Fade-in**

Usually modulations start immediately when a key is pressed and last for the entire duration of the sound. Sometimes, however, strong modulation can be objectionable in the early attack stage of the sound. To preserve the transients, the fade-in parameter can be used to gradually increase the modulation from zero to its maximum value, for a duration specified in seconds.

## **Offset**

Adds or subtracts an offset from the LFO Output, while simultaneously adjusting the LFO amplitude. This can lead to interesting effects when controlling this parameter via the modulation matrix.

## **Amount**

This knob adjusts the amount of LFO modulation that takes place. This is useful to fine-tune the overall amount of modulation in combination with the modulation matrix. Furthermore, the amount knob itself can be modulated via the modulation matrix. This allows to vary the modulation amount over time, for instance with a second LFO or an envelope.

## 4.15. Effect Bus 1+2

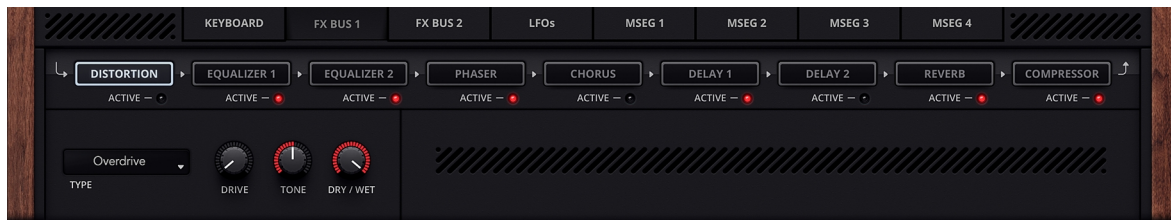
DUNE 3 offers two effect busses with nine effects in each to further enhance the sound. All of them may be used simultaneously, and most units offer several sub-types to further increase their versatility.

Note that all effects are global, that is all voices enabled for a specific effect bus are first summed and then processed by the effect bus.

The effects are processed from left to right in the order they appear. By default, the distortion is applied first and reverb last. You can modify the effect order, however, by dragging the individual sections. When you click on an effect label (e.g. "Delay") a red border will appear. Keep the mouse button pressed, and move the effect into any spot you like. Note that changing the effect order is not necessarily audible. The most significant change usually occurs when moving an active distortion or compressor effect to a different position.

### 4.15.1. Distortion

A distortion effect changes the signal in a nonlinear manner, which creates new overtones and usually results in a harsher sound.



DUNE 3 offers a variety of different distortion types:

- **Crunch** simulates a typical guitar pedal distortion effect.
- **Dirt** is similar to Crunch, but with a brighter timbre.

- **Fuzz** simulates a fuzz pedal with silicon transistors.
- **Grunge** is a distortion effect which specifically simulates the vintage DS-1 pedal, popularized by its usage in Grunge music.
- **Hard Clip** boosts then simply hard-clips the signal at 0 dB.
- **Overdrive** simulates a typical guitar overdrive effect.
- **Saturation** is based on a smooth saturation curve, ideal for the more subtle effects.
- **Screamer** is a classic stomp box emulation. There is two variations of this algorithm, one with 4x and one with 32x oversampling. The latter requires more processing power, but can provide higher quality particularly when distorting high notes.
- **Triode** models a typical valve-based guitar amplifier head.
- **Bitcrush** reduces the dynamic range of the signal to a low bit depth.
- **Dynacrush** is a variation of Bitcrush, however with a different timbre that depends on the input signal. Sometimes the results are more subtle than Bitcrush, while at other times they are harsher.
- **Ratecrush** reduces the sample rate of the signal, using a sample-and-hold circuit.
- **Exciter** is an effect which distorts only the high frequencies of the signal, leaving other frequencies largely unaffected.

Note that the Hard Clip, Bitcrush and Ratecrush effect can sound very digital and harsh at high drive settings, so those effects should be used with some care.

## Tone

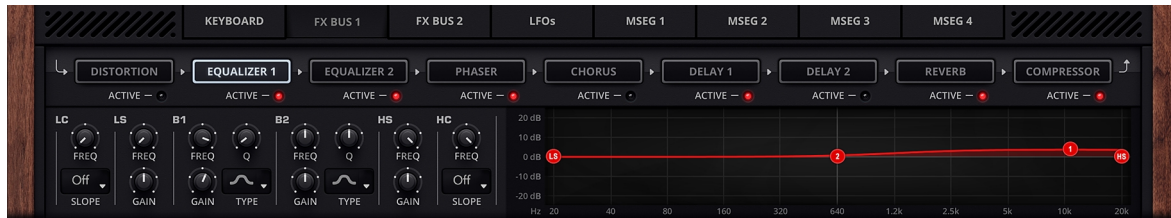
Tone usually controls the mid frequencies of the output signal, with the center position resulting in a largely neutral sound. The implementation differs dependent on which distortion type is chosen, however, so the effect can vary.

## Dry/Wet

Blends between the dry and processed signal. For guitar-type distortion effects, this parameter should be typically set to 100%. For all other types, it is often a good idea to try smaller values.

### 4.15.2. EQ-1 and EQ-2

An equalizer (EQ) is used to boost or attenuate a certain frequency range.



The EQ in DUNE 3 features five basic types:

- Peaking (Bell) 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.
- Low cut attenuates all frequencies below the chosen frequency using a constant slope given in dB per octave.
- High cut attenuates all frequencies above the chosen frequency using a constant slope given in dB per octave.

The EQ contains an interactive display on the right side of the interface.

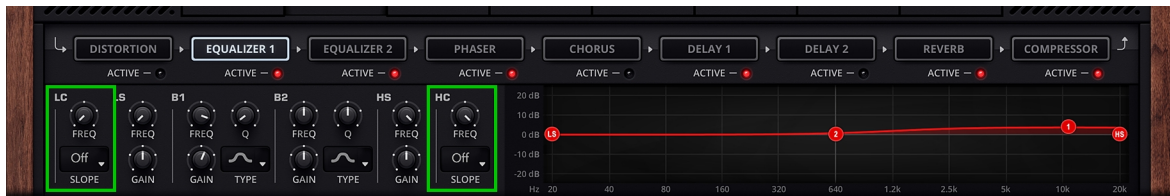
This display plots the frequency response from 20 Hz to 20 kHz according to the current EQ settings, thus giving instant visual feedback of how the audio signal is going to be processed.

The gain and attenuation of all frequencies are displayed along the vertical axis of the display. The gain range of the display is  $\pm 20$  dB.

Using the interactive display is straightforward: By left-clicking and dragging the labeled circles, the frequency and gain of the respective EQ band can be changed directly. The EQ bands B1 and B2 additionally feature a Q parameter, which can be changed by right-click and dragging a circle.

## Low and High Cut Filters

The Low Cut filter (LC) and the High Cut filter (HC), placed at the leftmost and rightmost side of the EQ, are offered in addition to the usual bell and shelf bands. Both the LC and the HC are turned off by default, in order to keep the EQ response flat and to reduce the CPU usage.



The filters are enabled by selecting the desired slope from the popup menu.

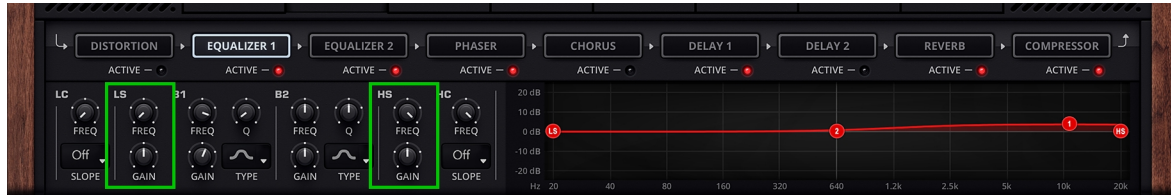
The filter parameters comprise the corner frequency given in Hz, as well as the slope given in dB/octave. The slope can be adjusted from 12 dB to 48 dB, thus allowing to go from a fairly gentle response to a very steep response. Both the low- and high cut filters are an excellent choice for cleaning up an instrument track, a recording, or even an entire mix.

The Low Cut filter, for example, can be used to remove everything below 30 Hz, choosing a steep slope like 48 dB/octave.

The High Cut filter can be used to remove high-frequency hiss. Shelf type filters are often not suitable for this purpose, since their slope is too gentle. The low- and high cut filter frequency can be modulated via the Modulation Matrix (see chapter 8. *Modulation Matrix*).

## Low and High Shelf Filters

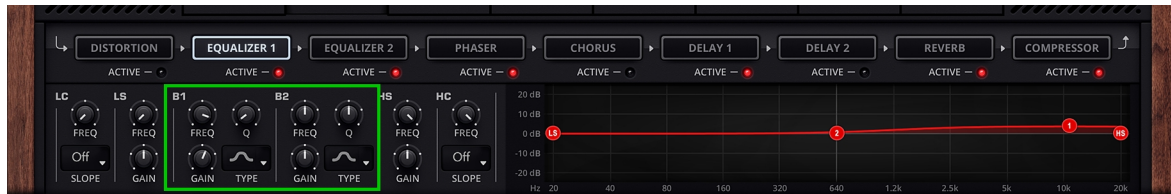
The low shelf (LS) filter is designed to boost or attenuate bass frequencies, although it can be used across the entire spectrum.



The gain parameter determines by how many dB the bass frequencies should be boosted or attenuated. The frequency parameter determines the corner frequency, where the filter starts to work on the signal. The typical range for the low shelf filter is 50-100 Hz. When attenuating bass frequencies, the low shelf filter only attenuates by the specified dB value. Thus it is preferable to the low cut filter whenever a subtle change in the bass area is desired, for instance lowering all bass frequencies below 100 Hz by -3 dB. The high-shelf (HS) filter is a mirrored version of the low shelf filter, designed to boost or attenuate high frequencies. The filter has the same parameters as the low shelf filter.

## Parametric EQ

The EQ bands B1 and B2 feature three parameters - frequency, gain and Q. All three parameters can be changed either from the interactive display as mentioned above, or via the corresponding knobs on the panel.





The frequency parameter sets the location of the peak of the bell curve on the frequency axis. Gain defines how much to cut or boost that frequency, and Q adjusts the bell shape from narrow to very broad.

Bell-shaped EQs are the most common type of EQs, due to their versatility. For instance, parametric EQs can be used to gently boost or cut any frequency region using a low Q factor, or they can be used to eliminate problem frequencies using a high Q and strong attenuation. They can also be used to sweep through a sound using a high Q factor, in order to find problematic frequencies.

The Type button allows to switch the default bell-shaped EQ to a shelf type. Due to the extra Q control, this results in a more flexible version of the low/high shelf filters.

### 4.15.3. Phaser

A phaser modifies the phase of the signal around some frequency, then adds it back to the original signal. The result is a notch in the frequency spectrum. By varying the operation frequency periodically with a LFO, the typical sweeping sound of a phaser is achieved.



### Type

The phaser can use between 2 and 12 stages. Each pair of stages produces one notch in the frequency spectrum, so e.g. two stages produce one notch. More stages result in more notches, which increases the phasing effect. The "Super Phaser" type is an analog inspired six-stage phaser with its own distinct timbre.

## Rate

Sets the modulation rate of the effect in Hertz (Hz). 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 eighth note.
- 1/16T sets the modulation rate to a sixteenth triplet.
- 1/1 sets the modulation rate to span one bar.

## Spread

Adjusts the width of the notches produced by the phaser. Usually 100% sounds best.

## Feedback

The output of the phaser can be fed back to its input, thus creating a resonant sweep. Both positive and negative feedback is possible, at center position there is no feedback.

## Freq

Sets the lowest frequency the Phaser. For the Chorus/Flanger types, this parameter specifies the frequency range to include. Usually this should be set to 100% to include the entire frequency spectrum.

## Depth

Sets the modulation depth of the LFO. Set to zero, there is no modulation and thus no sweeping effect, the sound will be static. When increasing the modulation depth, the phaser operates on a larger frequency region, which increases the sweeping effect.

## LR Offset

Since the phaser operates in stereo, both channels have their own LFO. With the offset knob, the relative phase difference between both channels can be adjusted from 0 to 180 degrees. Set to zero, both channels produce an identical sweep. Set to nonzero, both channels will operate on different frequency regions, making the effect more spatial.

## Dry/Wet

Mixes the dry and processed signal. Since the phasing effect is achieved by mixing the original and processed signal, the mix ratio should be typically around 50%. Lower values can be useful to make the effect more subtle, however, and higher values can be useful with strong feedback.

### 4.15.4. Chorus

A chorus or flanger effect is obtained by summing a signal with a delayed copy of itself.

When the delay is continuously varied with a low-frequency oscillator and separately for the left and right channel, the sound becomes thicker and more spatial. Using smaller delay times and adding feedback results in a flanging effect.



## Mode

- **Dimensional**, **JP Chorus** and **JP Flanger** are vintage chorus emulations, creating a warm and spatial stereo image.

- **Single** simulates a standard chorus effect with one delay line for each channel.
- **Dual** uses two delay lines per channel, to obtain a yet thicker sound.

### **Offset**

Adjusts the minimum delay time. Medium to high settings are useful to create a typical chorus affect, while low values are useful for flanging.

### **Rate**

Sets the modulation rate of the effect in Hertz (Hz).

### **Depth**

Sets the modulation depth of the LFO. Set to zero, there is no modulation and thus no sweeping effect, the sound will be static.

### **LR Offset**

Since the chorus operates in stereo, both channels have their own LFO.

With the offset knob, the relative phase difference between both channels can be adjusted from 0 to 180 degrees. Set to zero, both channels produce an identical sweep, while nonzero values make the chorus effect more spatial.

### **Feedback**

The output of the chorus delay lines can be fed back to their input, creating a resonance effect. Both positive and negative feedback is possible, at center position there is no feedback. For a classic chorus sound, the feedback should be set to zero. For flanging effects, use strong positive or negative feedback.

## Low Cut

The Low-Cut knob cuts bass frequencies of the chorus effect. This only affects the processed signal. Especially for low-pitched sounds, the Low Cut filter is useful to avoid a frequency mash caused by the chorus effect.

## High Cut

The High-Cut knob cuts treble frequencies of the chorus effect. Again this only affects the processed signal.

## Dry/Wet

Mixes the dry and processed signal. Since the chorus effect is achieved by mixing the original with a delayed copy of itself, the mix ratio should be typically around 50%. Lower values can be useful to make the effect more subtle, however, and higher values can be useful with strong feedback.

### 4.15.5. Delay 1 and Delay 2

A delay effect produces a series of echoes.



The duration of the echoes is usually locked to the host tempo. A total of eleven delay programs are available.

## Type

- **Simple** creates a series of echoes centered in the stereo field.
- **Simple+Offset** is identical to the Simple delay, except that the right channel is delayed. When both channels share the same delay time, this sounds like the ping-pong delay except for a missing first echo.
- **Ping-Pong** creates echoes alternating between the left and right channels.
- **Dual Ping-Pong** uses two ping-pong delay units, one for each channel. The result is a pair of echoes alternating between the left and right channel.
- **Diffuse** works similar to the Simple delay, except that each echo becomes increasingly diffuse.
- **Tape** works similar to the Simple delay, except that each echo becomes increasingly saturated.
- **L-C-R** (Left Center Right) is a special program where the echoes alternate between the left channel, the center, the right channel and the center again etc.
- **Filtered** created echoes with a changing timbre. This delay program features Cutoff and Q knobs to control the filter sound.
- **Rhythmic** works like the Simple delay, but uses a specific pattern.
- **Swing** allows to adjust the timing of every other echo (using the Swing knob), thus creating swing-type effects.
- **Vintage** is a special delay program, creating warm and spacious echoes.

## L/R Rate

The delay time can be specified independently for the left and right channels.

By default the delay times are synced to the host tempo and are 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.

Turning off SYNC allows to specify the delay time in milliseconds (ms), for each channel separately. When using very short delay times (e.g. 1 ms), the ear can no longer perceive the individual echoes as such. The result is a comb-filter effect which can sometimes sound interesting. Turning off sync also allows to modulate the delay time via the modulation matrix, useful to obtain special effects.

### **Low Cut**

The echoes can be processed by a 6 dB/oct lowpass filter, making each subsequent echo darker. Set to zero, the lowpass filter is disabled.

### **High Cut**

The echoes can be processed by a 6 dB/oct highpass filter, making each subsequent echo brighter than the previous one. Set to zero, the highpass filter is disabled.

### **Width**

By default, all delay programs operate in full stereo. When using the ping-pong delay, for instance, the generated echoes will alternate between the left and right speakers, with no signal in the center. If this effect feels too strong, the width knob can reduce the stereo image, all the way down to mono.

## Feedback

The feedback parameter adjusts how long 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 delay modulation rate in Hertz (Hz).

## Mod-Amt

All delay programs allow to modulate the delay time with a LFO. The amount knob adjusts the modulation depth. Set to zero, no modulation takes place, the echoes sound rather static. Small modulation amounts cause the delay to sound slightly more organic, as the delay constantly varies a little over time. Large modulation amounts cause a noticeable strong pitch modulation, which can be used for special effects.

## Dry/Wet

Blends between the dry and processed signal.

### 4.15.6. 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.





Eleven different algorithms are offered in DUNE 3.

### Type

- **Ambience** is useful to add a subtle ambience to a sound.
- **Cathedral** is an algorithm designed for large spaces, with a reverb time of 4-5 sec. or more.
- **Chapel** simulates a larger chapel, good for reverb times of 2-3 sec.
- **Big Room** simulates a room, designed for a reverb time around 1 sec.
- **Gated** and **Studio Room** are special programs crafted for (very) short reverb times.
- **Plate** is a highly diffuse reverb program, it does not simulate a particular space.
- **RutaVerb** simulates a concert hall. The algorithm produces a dense reverb with high realism and a nice reverb tail.
- **Hall** simulates a space of various sizes, small, medium, or large. The algorithm works best with reverb times of 3 secs and above.
- **Slapback Hall** simulates a hall with strong early reflections occurring after a few milliseconds.
- **Shimmer Room** and **Shimmer Hall** are special programs which create a "shimmer" effect on top of the reverb sound. The amount and brightness of the shimmer effect is controlled by the Shimmer and Regen knobs, respectively.
- **Stadium** simulates a very large reverberant space.

### 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**

This parameter adjusts the reverb time in seconds. Reverb time is defined as the time it takes the wet signal to reach -60 dB. Note that the displayed value is an approximation, and that other parameters like high-frequency damping can shorten the perceived reverb time.

## **HF Damp**

The high-frequency damp parameter changes the simulated room's wall materials. Low settings correspond to reflective walls (bright, uncolored sound), high settings to very absorbent walls (progressively darker timbre).

## **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 parameter, only the reverberated signal.

## **Color**

The color parameter can be used to equalize the mid frequencies of the reverberated signal. At center position the reverberated signal is unaffected. Turning the knob to the left cuts the mid frequencies, turning it to the right boosts the mid frequencies.

## **Mod-Amt**

All reverbs in DUNE 3 employ modulation, to generate a rich timbre. The modulation amount parameter adjusts how strong this modulation is.

## **Width**

Adjusts the width of the reverb. By default the reverb width is 100%, thus creating a maximally wide stereo image.

## Dry/Wet

Blends between the dry and processed signals.

### 4.15.7. Compressor

The purpose of a compressor is to reduce the dynamic range of a signal, which increases the perceived loudness and can reduce clipping problems in a digital environment. A compressor performs dynamic range reduction by lowering the level of loud passages. What is considered "loud" is specified by the user, in form of a threshold level given in decibel (dB).



Four different kinds of compressors are available in DUNE 3.

#### Mode

- **Air** is a multi-band compressor designed to maximize loudness. The compressor typically boosts treble/presence to some degree, hence the name.
- **Opto** is a single-band compressor inspired by hardware devices using optical attenuators to control the dynamics of the signal. Unlike the Air compressor, it yields a rather transparent signal, as long as the amount of compression is low.
- **Punch** is based on a classic analog solid-state design, useful to give more punch to some kick or drum sounds.
- **Vintage** is a classic single-band compressor, which adds a small amount of smooth saturation on the output signal.

## Input

Adjusts the input level. The input level affects the amount of compression, and should be adjusted such that the level meter peaks around 0 dB.

## Ratio

Ratio tells the compressor how much to lower those passages that are louder than the threshold. A ratio of 1:1, for example, means that the sound comes out 1 to 1 - as it is - the compressor does not do anything. At a ratio of 2:1, the loud passages will be lowered by a factor of 2 - they will come out only half as loud as they were when they came in. At a ratio of 4:1, they will come out 4 times softer, and so on.

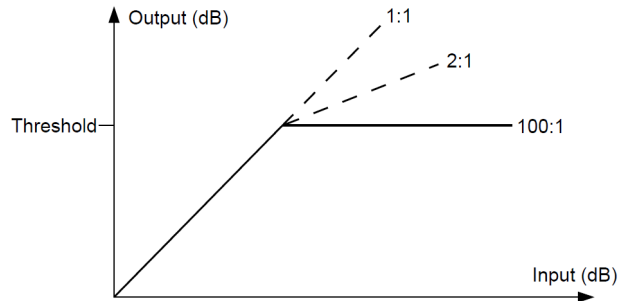


Figure 4.15.7. The compression input/output ratio.

So the Ratio setting effectively decides how much the passages above the threshold will be compressed.

Refer to fig. 4.15.7 for a graphical representation of the compression input/output ratio. The image shows ratios 1:1, 2:1 and 100:1. At the highest setting 100:1, the compressor will act as a limiter, and not pass any signal levels above the threshold.

## Thres

The threshold knob sets the level at which a passage is considered "loud" - that's the point at which the compressor will start working. For example, if you set the threshold to -16 dB, then everything below that level will pass straight through, while anything above -16 dB will be attenuated according to the ratio setting.

## **Attack**

The attack parameter adjusts the time it takes for the compressor to react to a signal exceeding the threshold.

The attack time involves a trade-off: Short attack times usually sound more transparent, but can also cause distortion artifacts on some signals when the compressor reacts too quickly.

Long attack times usually do not cause distortion problems, but fast transients pass through uncompressed which sometimes sounds objectionable.

## **Release**

The release parameter adjusts the time it takes for the compressor to recover when the threshold is no longer exceeded.

## **Output**

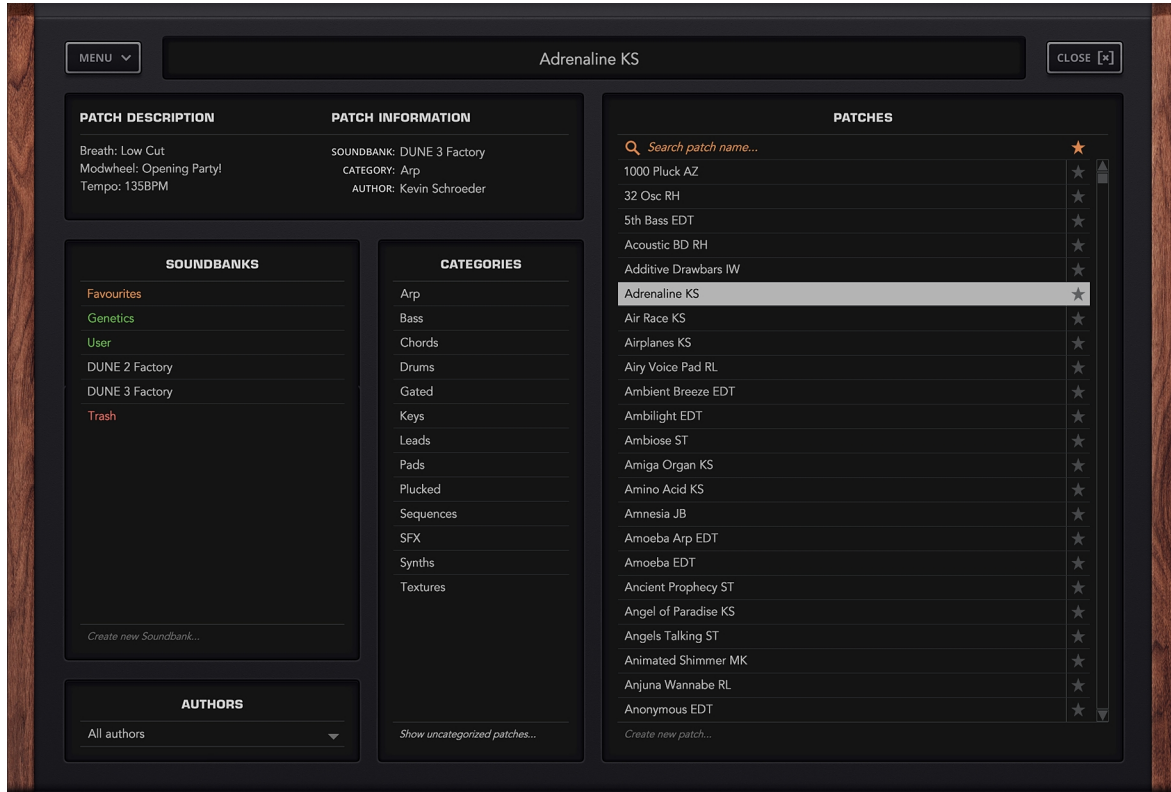
Adjusts the compressor output level in decibel (dB).

## **Dry/Wet**

Blends between the dry and processed signals. While a compressor is traditionally used 100% wet, mixing in the dry signal has become a popular trick known as "parallel" or "New York" compression.

## 5. Patch Browser

Since version v3.5, DUNE 3 contains an extensive patch browser to search for specific sounds, manage soundbanks, mark favourites, or even create completely new sounds with the Genetics function. The patch browser is opened by left-clicking on "BROWSER" in the center monitor.



Important: Since the patch browser is typically used to change the currently selected patch, any changes made to the active patch should be saved before using the patch browser.

The functions of the patch browser are explained in more detail below.

## Soundbanks

When the browser is opened for the first time, all installed patches from all soundbanks are shown on the right column. By left-clicking on a soundbank, the displayed patches can be limited to the selected soundbank. By right-clicking on a soundbank, that soundbank can be renamed, or its location in the file system can be revealed.

In addition to the factory soundbanks, there are special banks highlighted in color:

- **Favourites:** This displays all patches that have been marked as favourites.
- **User:** Own patches can be stored in this soundbank.
- **Trash:** Displays patches that have been moved to the trash can. These patches can then be permanently deleted by right-clicking and selecting "Clear Trash".

With the function "Create new soundbank..." a new soundbank can be created with the desired name.

## Authors

This popup menu can be used to limit the patches displayed on the right to a specific author. By default, all authors are displayed.

## Categories

The patches shown can be limited to specific categories. Multiple selection is possible by holding the CTRL key.

The "Show uncategorized patches" function displays all patches that have not yet been assigned to a category yet. This is useful for sound designers who are creating a new soundbank.

## Patches

The right column shows all patches that match the current selection criteria. Left-clicking on a patch selects and makes it the current patch. Multiple selection is possible by holding either the CTRL or SHIFT keys. Using Drag+Drop, selected patches can be assigned to a specific soundbank or category.

The search box on top can be used to search for specific patch names. Entering a string will immediately show all patches containing this string.

With the "Create new patch..." function at the bottom, a new Init Patch can be created in the current soundbank.

## Menu

The MENU button provides the following functions:

- **Genetics:** Opens the Genetics tab, which can be used to create new sounds (see below).
- **Import Patch/Folder:** These functions can be used to add a single DUNE 3 patch or a directory of patches to the sound banks. This can be useful to quickly integrate patches from external hard drives or USB sticks.
- **Import ZIP:** With this function a zipped soundbank can be imported directly. The soundbank will be named like the ZIP.
- **Reload Soundbanks:** Reloads all patches, in case patch files have been added or deleted manually via the file system.

Note: For the folder and ZIP import, if there are wavetables found besides the patches, these are also automatically copied into the wavetables. In this way, third-party soundbanks can be installed quickly and conveniently.



## Genetics

With the Genetics function, new sounds can be created from 2-3 already existing patches. In doing so, properties from these patches are randomly thrown together.

After selecting at least two patches in the patch window on the right (hold down the "CTRL" key for multiple selection), a new patch can be created using the "Generate" function. If the selected patches fit well together, usually only a few clicks on "Generate" are necessary to get a usable sound.

Important: When pre-listening, the listening volume should be relatively low, since the generated patches are random and can therefore sometimes become very loud.

The "Auto" function is similar to the "Generate" function, with the difference that here two new patches are randomly selected from the current directory with each click.

Successful sounds can be quickly saved with SAVE. These sounds are then stored in their own soundbank "Genetics" and are date and time stamped.

The Genetics function can weight the selected patches differently. This is controlled by the "Amount" knob. The larger the value, the more patches 2 and 3 are factored into the final sound. At the minimum value of 0%, patches 2 and 3 are not considered at all, i.e. the first patch remains as is.

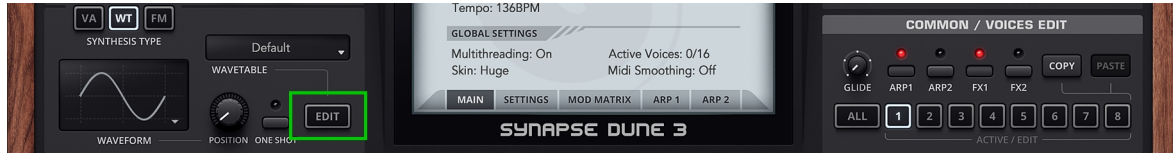
## Keyboard shortcuts

To speed up the workflow, the browser supports the following keyboard shortcuts:

- **Arrow keys** select the next/previous patch.
- **F2** renames the currently selected patch.
- With **ESC** or **RETURN** the browser can be closed.

## 6. Wavetable Editor

DUNE 3 offers the possibility to create your own waveforms and wavetables. To use the Wavetable Editor, one of the oscillator blocks must be set to Wavetable mode (WT). Then you can open the Wavetable Editor with the EDIT button:



The Wavetable Editor is divided into three sections.

- In the upper bar there are tools for editing and a small window for navigation within the waveform.
- In the middle is the actual editor in which waveforms can be drawn directly with the mouse.
- In the lower area you can select the waveform you want to edit. A wavetable can contain up to 256 waveforms, with the +/- buttons the number of waveforms can be adjusted. A formula editor allows to create a waveform or even a complete wavetable by entering mathematical formulas.

### 6.1. Toolbar



#### Menu

The MENU popup contains a number of useful functions for creating wavetables.

### **Import WAV (Single-Cycle)**

This function loads a wave file (WAV) that replaces the currently selected waveform. The wave file is automatically adjusted to the required length of 2048 samples.

### **Import WAV (Fit To Wavetable)**

Loads a wave file (WAV) as a complete wavetable. The number of waveforms remains constant, so the wave file is "squeezed" into the wavetable.

### **Import WAV (Adjust Wavetable)**

Loads a wave file (WAV) as a complete wavetable. With this variant, the number of waveforms is automatically adjusted to the length of the wave file. This function is useful if the wave file consists of several waveforms with 2048 samples each.

### **Export Wavetable (DUNE 3)**

Saves a wavetable in DUNE's WT format. If the wavetable is saved in the \Wavetables folder, it will appear in the wavetable list the next time DUNE 3 is opened.

Exporting wavetables is only necessary if the wavetable is to be reused in other patches or projects. Otherwise it is sufficient to save the current patch.

### **Export Wavetable (WAV)**

Saves a wavetable in WAVE format.

### **Morph Wavetable**

The Morph function creates a seamless transition from the first to the last waveform within a wavetable. All intermediate waveforms are overwritten during this process.

### **Normalize Wavetable**

Normalizes the amplitude of the entire wavetable to 0dB.

### **Clean Wavetable**

Creates a new, empty Wavetable.

### **Copy Waveform**

Places the current waveform in the clipboard.

### **Paste Waveform**

Copies the waveform from the clipboard to the current slot.

### **Invert Waveform**

Mirrors the waveform vertically.

### **Reverse Waveform**

Mirrors the waveform horizontally.

### **DC Offset Waveform**

This function removes the DC offset of a waveform, i.e. it ensures that the energy above and below the zero line is balanced. Removing DC offsets avoids clicks during the transients.

### **Normalize Waveform**

Normalizes the amplitude of the current waveform to 0dB.

### **Clean Waveform**

Deletes the waveform in the editor.

### **Add Waveform**

Adds an additional waveform to the wavetable at the end of the wavetable.

### **Insert Waveform**

Inserts an additional waveform at the current wavetable position.

### **Remove Waveform**

Removes the selected waveform from the wavetable, the wavetable becomes smaller.

### **Tools**

- The **Selection** tool can be used to select a horizontal section in the waveform view.
- The **Pen** tool can be used to draw waveforms directly in the Editor.
- The **Line** tool can be used to draw straight lines in the editor.
- The **Segment** tool can be used to draw several contiguous lines or curves in the editor. In this mode, the left mouse button is used to set curve points, which can then be moved by dragging the mouse.
- The **Sine** tool open the additive editor, which allows to construct a waveform using individual partials. Each partial has a magnitude (upper part) and a phase (lower part). The additive editor is useful to create e.g. organ or vocal-like waveforms.

### **Undo/Redo**

Use the Undo and Redo buttons (left/right arrow) to return to previous editing steps.

## Navigation panel

The navigation panel shows the current waveform and the currently selected section of it. If you zoom into the waveform, the section can be moved using the navigation panel.

## Zoom

With the zoom buttons  $-/+$  on the right side you can zoom in and out of the waveform. Zooming is also possible using the mouse wheel, if the mouse is either in the Navigation Panel or the Edit Area.

## Close

Closes the wavetable editor.

## 6.2. Editor

The editing area is located in the middle. Here you can use the mouse to edit the currently selected waveform. If, for example, the Pen tool is selected, you can draw the waveform directly.



## 6.3. Using the editor

### Selection tool

If the selection tool is selected, a selection can be created by left-click/drag. Alternatively, the entire waveform can be selected by double-clicking. With a right click, operations such as copy/paste, fade in/out etc. can be applied to the selection.

For real-time editing, Ctrl+Left-click and Alt+Left-click are useful tools. They can be used to directly change the phase or volume of the selection.

### Pen tool

Waveforms can be drawn directly with the left mouse button. As with the selection tool, Ctrl+left mouse button and Alt+left mouse button can be used to adjust the phase or volume of the waveform. A click on the right mouse button is very practical, zooming directly to the selected sample.

### Line tool

The Line tool allows you to draw a straight line by left-click/drag. An important trick is Shift+Left-click. This draws the line in such a way that it always runs along the curve. In this way, there are no hard transitions to the existing waveform.

### Segment tool

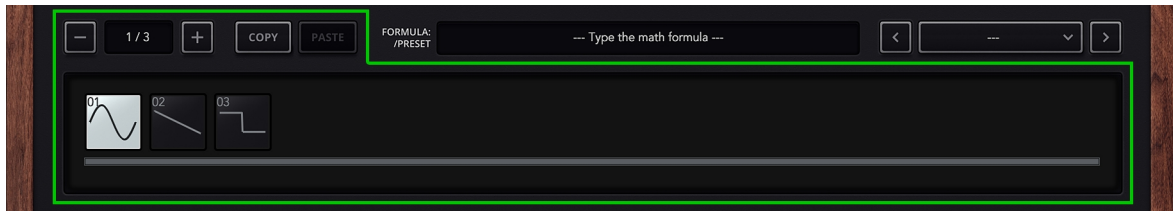
While the Pen and Line tools are suitable for drawing waveforms directly, the Segment tool can be used to create waveforms indirectly along a set of points. With the left mouse button new points can be created or existing points can be moved. By default, a straight line is drawn between two points, which can be changed into a curve by left-click/drag. An important trick is Shift+Left-click. This draws the points in such a way that it always runs along the curve. In this way, there are no hard transitions to the existing waveform.

## Additive tool

The Additive Editor can be used to construct waveforms from individual partials.

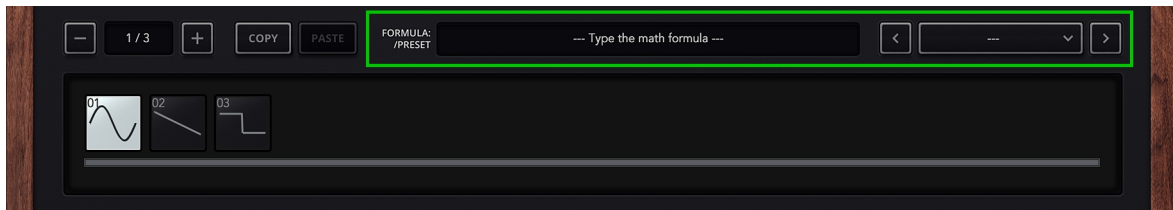
Use left-click/drag to set the magnitude (upper part of the edit window) or the phase (lower part) of a partial. The most useful partials are the first 10-20 partials, which largely shape the sound of a waveform. The additive editor is useful to create organ, bell or vocal-type waveforms.

### 6.4. Waveform selection



In the lower part of the wavetable editor, all waveforms of the wavetable are displayed in thumbnail view. Left-click to select a waveform for editing, and drag and drop to swap individual waveforms with each other. The latter is important if you are using modulation to pass through the wavetable, as the order of the waveforms has a major influence on the sound.

### 6.5. Formula Editor



By entering a formula such as  $\sin(x*\text{PI})$  you can directly generate a complete waveform.



Next to the input field on the right you will find some presets for inspiration. The formula editor can also be used to create entire wavetables at once. If you insert a second parameter "y" into the formula, it represents the ascending position of the waveform within the wavetable.

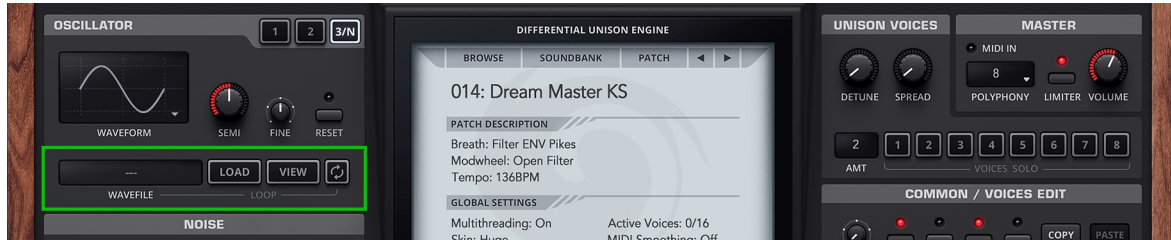
Example: The formula  $\sin(x*\text{PI})*y$  would slowly fade in a sine wave.

The following list contains all supported functions that can be used:

- $\sin(x)$ ,  $\cos(x)$ ,  $\tan(x)$  Trigonometric functions
- $\arcsin(x)$ ,  $\arccos(x)$ ,  $\arctan(x)$  Inverses of trigonometric functions
- $\sinh(x)$ ,  $\cosh(x)$ ,  $\tanh(x)$  Hyperbolic functions
- $\text{sgn}(x)$  or  $\text{sign}(x)$  Sign (returns 1 when x is positive, -1 if negative)
- $\text{rnd}(x)$  Random value
- $\text{exp}(x)$  Exponential Function
- $\text{sqrt}(x)$  Square Root of x
- $\text{abs}(x)$  Absolute value of x
- $\text{min}(x1,x2)$  Minimum of 2 values
- $\text{max}(x1,x2)$  Maximum of 2 values
- $\text{pow}(x1,x2)$  x1 to the power of x2
- $\text{mod}(x1,x2)$  x1 Modulo x2

## 7. Playing WAV files

Since version 3.4, WAV files can be included in sounds in DUNE 3. This is done via oscillator 3:



After pressing the LOAD button, a directory window opens where you can select your own WAV files. Supported are 16, 24, and 32-bit WAV files between 44.1 and 192 kHz sample rate, mono and stereo.

If a WAV file is loaded, the sawtooth/pulse/triangle oscillator is automatically switched off and replaced by the WAV file instead. Note if no sound is heard after pressing a key, the reason might be that the level of oscillator 3 in the mixer is set to zero.

The VIEW button can be used to view and edit the loaded WAV file. In particular, the loop points can be set or adjusted here.

The WAV editor is divided into two areas:

- In the upper bar there are tools for editing and a small window for navigation within the waveform.
- In the middle is the actual editor. Here the loop points can be set or adjusted. Waveforms can be drawn directly with the mouse or areas can be selected and changed (e.g. Fade In and Fade Out).

## 7.1. Toolbar



### Menu

The MENU popup contains a number of useful functions for editing WAV files.

### Loop

Turns the loop function on or off. When loop is enabled, the selected segment is repeated continuously as long as a note is played.

### Draw Ghost

If Draw Ghost is activated, the loop end is overlaid with the loop start when zooming in. The beginning of the loop is displayed as a dark gray waveform. To get a smooth loop, both waveforms should be nearly congruent.

### Find Loop Region

This function can be used to automatically search for favorable loop points in order to obtain a loop without clicks. The simpler the waveform, the better the loop usually sounds.

### Show Samples / Seconds

When setting the loop points, the current position is displayed either in samples or in seconds.

### DC Offset

This function removes the DC offset of the audio file, i.e. it ensures that the energy above and below the zero line is balanced.

## Normalize

Normalizes the amplitude of the current waveform to 0dB.

## Revert

Reverses changes made to the WAV file.

## Tools

- The **Loop** tool can be used to select a region that will be repeated continuously during playback. To do this, use the left mouse button to determine the start point and drag to the desired end point. Then both the start and end points can be moved separately to define the loop region more precisely. If clicked outside the loop region, the loop is completely removed and a new region can be defined.
- The **Selection** tool can be used to select and modify a horizontal section in the audio file.
- The **Pen** tool can be used to draw directly in the editor.

## Navigation panel

The navigation panel shows the current audio file and the currently selected section of it. If you zoom into the audio file, the section can be moved with the help of the navigation area.

## Zoom

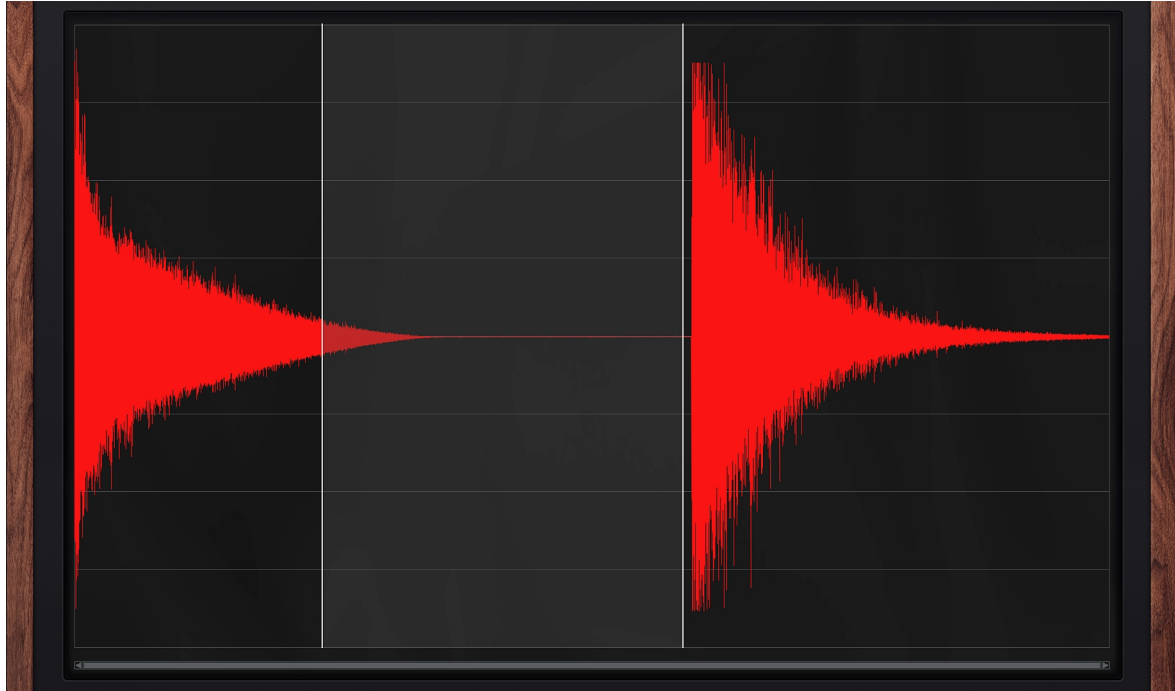
With the zoom buttons -/+ on the right side you can zoom in and out of the audio file. Zooming is also possible using the mouse wheel, if the mouse is either in the Navigation Panel or the Edit Area.

## Close

Closes the WAV editor.

## 7.2. Working with the WAV Editor

The editing area is located in the middle. Here you can use the mouse to edit the currently loaded WAV file. If, for example, the Loop tool is selected, you can select a region that will be repeated continuously during playback.



### Loop tool

With this tool, a loop can be created by left-clicking/dragging in the editor. Once a loop is created, clicking outside the loop area will erase the loop. The entire waveform can be selected by double-clicking.

## **Selection tool**

Using this tool, a selection can be created by left-clicking/dragging. Alternatively, the entire waveform can be selected by double-clicking. Right-click to apply operations such as copy/paste, fade in/out, etc. to the selection. Furthermore by right-click you can set up the loop region directly from the selection.

For real-time editing, Ctrl+left click can be used to adjust the volume of the selected area.

## **Pen tool**

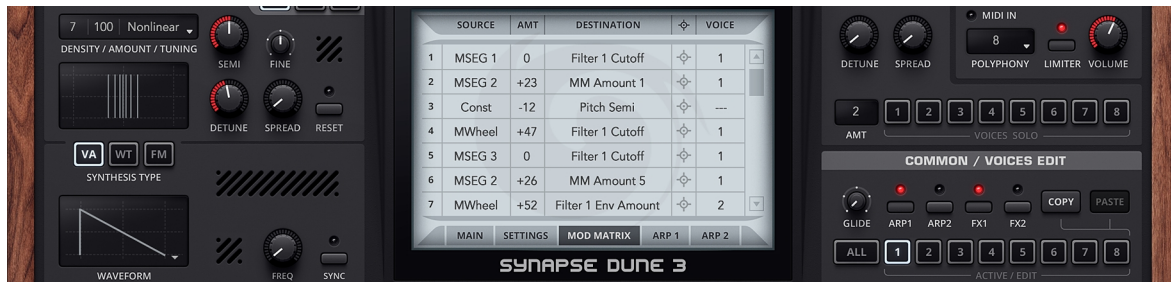
Use the left mouse button to directly modify individual points of the audio file. By right-clicking, it is possible to zoom directly to the current sample position.

For real-time editing, Ctrl+left click can be used to adjust the volume of the waveform.

## 8. 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:



The modulation matrix in DUNE 3 is located in the center of the interface, and accessed by clicking on the Mod Matrix button. Up to 32 source/destination combinations are possible per patch.

The source and destination parameters can be set by left clicking, the amount parameter (AMT) is moved like a knob. The destinations can also be dragged directly to knobs or switches using the drag+drop icon next to the "DESTINATION" column.

The modulation sources comprise the LFOs. The ability to link a LFO to any sound parameter makes the traditional LFO destination parameter obsolete and offers far greater flexibility. Classic destination parameters include Pitch (to obtain a vibrato effect), Volume (to obtain a tremolo effect) as well as Filter Cutoff. Further sources include the graphical envelopes (MSEG 1-4), which also must be assigned to a destination in the modulation matrix to work. Typical destinations for the graphical envelope generators include pitch, volume, or the amount of frequency modulation (FM) when using FM synthesis.

The modulation matrix is also used to assign MIDI controllers to sound parameters. The modulation wheel or expression pedal, for instance, can be chosen as a source and linked to any destination parameter. The effect of note velocity is also controlled from within the modulation matrix. While typically linked to volume, it can be easily set to affect the filter cutoff frequency or other sound parameters.

Advanced users will appreciate the ability of the modulation matrix to modulate itself, by choosing any of the 32 modulation slots as a destination.

### 8.1. The differential unison engine (DUNE)

The eight Unison voices in DUNE 3 can be modulated separately via the modulation matrix if desired. This makes the Unison function in DUNE much more flexible and powerful than the Unison function found in traditional synthesizers, which often comes with no or very few parameters. This concept is referred to as the differential unison engine, hence the name "DUNE".

To modulate unison voices, the modulation matrix offers a fourth column "Voice", which can be used to target specific unison voices. This concept can best be explained using a simple example matrix:



In this example, we have assigned all three LFOs to three different destinations: Filter Cutoff, Pan (Panorama) and Volume. In the first row, the Voice field is left blank, which simply means to apply the modulation to all unison voices in use. If the number of unison voices is set to e.g. five voices, then the filter cutoff will be modulated in the same manner for all five voices. The second and third column, set to modulate the volume and panorama, are restricted to affect the second and third unison voice, respectively. As a result, only those voices will be modulated.



To experiment with and understand the unison engine, try the example above (or something similar) on an empty patch (use the "Initialize Patch" function in the Patch menu). Make sure that the number of unison voices is set to at least three. The number of unison voices can be set from the Unison panel, located above the modulation matrix. Use the SOLO switches in the same panel to toggle between individual unison voices and see how the modulation affects them.

A further example will demonstrate how to set two unison voices to different settings, and conclude the introduction of the differential unison engine:



Here we modulate two parameters, filter cutoff and filter resonance. The modulations are set to affect the second unison voice. As a source, we chose "Const", which means constant. The result of those modulations is to simply adjust the two filter parameters relative to the knob settings. For example, if the Cutoff knob is set to 50%, the second voice will have a cutoff setting corresponding to 100% (50% + 50%).

Note that in DUNE 3, there is two ways to change the parameters of unison voices. You can use the modulation matrix as described in the second example above, or you can simply edit the voice parameters directly, by using the Voice Edit section.

By editing voice parameters directly, you perform absolute changes to the unison voices. This is required if you want to use different filter types, different synthesis models or different wavetables in the unison voices, and is generally a good idea if you want to stack completely different sounds.

Using the modulation matrix instead, you perform relative changes to the unison voices. This is great to keep common controls: Suppose you build a C-minor chord stab with three unison voices using the modulation matrix, then all voices can still be edited together. Tweaking the cutoff knob or changing pitch will always affect the entire chord and not just a single voice.

## 8.2. List of Sources

The following section lists all available modulation sources with a brief explanation. All sources, whether it is MIDI data or synth parameters, are converted to the same range, which is [0,+1] for unipolar and [-0.5,+0.5] for bipolar sources. The LFOs, the Pitch bend wheel, the Random and the Alternate modifier are bipolar sources, all other sources are unipolar.

The current value of a source is multiplied with the amount value [-100 to +100] in the same modulation slot. The result of the multiplication is then added to the selected destination parameter.

### Velocity

The MIDI Note-On velocity information, which is transmitted once at the instant a key is pressed. The harder a key is hit, the higher the transmitted value.

### Vel>100

Emits maximum value if the MIDI key velocity is greater than 100, and zero otherwise. This source can be used to obtain additional effects when a key is hit hard, for example an extra layer could be triggered.

### Keytrack

The MIDI Note number becomes the source of modulation, relative to C0. This means low keys emit a low value while high keys emit a high value. This source can be used to make a patch key-dependent, for instance shorten the envelopes when pressing a higher key.

### Freqtrack

Identical to key track, except that this source is based on the pitch of the MIDI key, rather than its note number.

## **MWheel**

The MIDI modulation wheel data (controller CC#01).

## **PWheel**

The MIDI pitch wheel data. Note that the pitch bend wheel always changes the pitch up or down dependent on the Bend up/down setting in the main panel. Set both to zero if you wish to use the pitch wheel exclusively for a different purpose.

## **ATouch**

Quality MIDI keyboards not only transmit velocity, but send pressure information as well. This parameter is called Aftertouch. In contrast to velocity, the aftertouch information is sent permanently and for the entire keyboard, not per key. It is transmitted for as long as any key(s) are being pressed. Note that there is a few keyboards which support sending pressure information per key. This is called polyphonic aftertouch. While polyphonic aftertouch is supported by the MIDI standard, such keyboards are very rare and never found wide usage. Hence, polyphonic aftertouch is not supported.

## **Const**

This source simply sends a constant value of 1. Hence, the amount setting is directly added to the destination parameter. This can be useful for a wide range of tasks, for instance to offset a parameter for a specific unison voice or to set parameters only available in the modulation matrix.

## **Random**

Sets a random value whenever a voice is triggered. Very useful with pan as a destination, or to add subtle pitch modulations simulating the behavior of acoustic instruments or vintage analog synthesizers (both of which have a slightly inconsistent pitch each time a note is played).

## **Alternate**

The alternate destination switches polarity every time a key is pressed. Thus this destination can be used e.g. to let sounds alternate between the left and right channel: *Alternate +100 Amp Pan*.

## **Arp Note**

The note information sent from the arpeggiator. This is a special purpose parameter, which should be used in combination with Pitch Semi as a destination and with the Arp Mode set to "Silent". This will result in direct pitch changes within a voice, an effect common in old computers such as the C64. The resulting sound is different from the regular arpeggiator, which constantly triggers and releases voices.

## **Arp Vel**

The velocity information sent from the arpeggiator. This can be used to modulate any parameter rhythmically, and sync'ed to the host tempo if desired.

## **Breath**

MIDI Breath controller (CC#02).

## **Foot**

MIDI Foot controller (CC#04).

## **Expression**

MIDI Expression controller (CC#11).

## **Osc 1**

The output of oscillator 1. When using this source, make sure to adjust the modulation rate to "Fast" or "Audio-Rate".

## **Osc 2**

The output of oscillator 2. When using this source, make sure to adjust the modulation rate to "Fast" or "Audio-Rate".

## **Osc 3**

The output of oscillator 3. When using this source, make sure to adjust the modulation rate to "Fast" or "Audio-Rate".

## **Noise Gen**

The output of the noise generator. When using this source, make sure to adjust the modulation rate to "Fast" or "Audio-Rate".

## **LFO 1**

Current value of LFO-1.

## **LFO 2**

Current value of LFO-2.

## **LFO 3**

Current value of LFO-3.

## **LFO\*MW**

The value of LFO-1, LFO-2 or LFO-3 multiplied by the modulation wheel data. Use this to create vibrato or tremolo effects with the depth controlled by the modulation wheel.

**LFO\*AT**

The value of LFO-1, LFO-2 or LFO-3 multiplied by the aftertouch data. Useful to create vibrato type effects when applying pressure on the keyboard.

**Voice #**

The unison voice number counting from zero, i.e. the first unison voice sends 0, the second voice sends 1, the third 2, and so on. This source can be useful to quickly set a parameter of each unison voice to a different value.

**Filter Env**

Current value of the filter envelope.

**Amp Env**

Current value of the amplitude envelope.

**MSEG 1**

The output of envelope generator 1.

**MSEG 2**

The output of envelope generator 2.

**MSEG 3**

The output of envelope generator 3.

**MSEG 4**

The output of envelope generator 4.

## 8.3. List of Destinations

The destinations of the modulation matrix comprise almost all sound parameters DUNE has to offer (as described in the chapter 4. *Sound Parameters*), plus most effect parameters and a few helper functions not available on the front panel.

### 8.3.1. Common

#### Pitch Semi

The overall tuning of in semitones. An amount value of +1 corresponds to one semitone, +12 transposes the oscillator one octave up, -12 transposes one octave down etc.

#### Pitch Fine

The overall fine tuning in cents. Cents are a fraction of a semitone (+50 equals half a semitone, +100 a full semitone). Modulate this parameter using a LFO as a source in order to obtain vibrato effects. For strong vibratos spanning a larger pitch range, use Pitch Semi as a destination instead. The modulation can be restricted to specific voices if desired. For example, you could have a static, low-pitched sine wave on the first voice and a vibrating sawtooth on the second.

#### Osc 1+2 Init Phase

This destination provides a way to set the initial phase for oscillator 1+2 simultaneously. See the Oscillator section below.

#### Arp 1 Rate

Changes the rate of arpeggiator 1. Note that the arpeggiator Sync should be set to Off when using this destination.

## **Arp 2 Rate**

Changes the rate of arpeggiator 2. Note that the arpeggiator Sync should be set to Off when using this destination.

## **Arp Hold**

Turns on Arp Hold if the source times the amount is greater or equal to 50. The intended usage is to assign a pedal to this destination, e.g. "Foot" "+50" "Arp Hold".

## **Mixer Volume**

Adjusts the volume of all oscillators, ring mod and noise. This destination is useful whenever the pre-filter volume needs to be automated (otherwise, Amp Level can be used as a destination instead).

## **Master Volume**

Sets the master volume.

### **8.3.2. Oscillator 1, 2, 3**

This section describes all oscillator destinations. Note that oscillator 3 has fewer parameters, only the first six parameters are available here.

## **Osc Semi**

The tuning of the oscillator in semitones. An amount value of +1 corresponds to one semitone, +12 transposes the oscillator one octave up, -12 transposes one octave down etc.

In the context of the differential unison engine, choosing Osc Semi as a destination allows you to specify arbitrary tunings for each voice.



For example, if you long for a fourth or fifth oscillator having a different pitch than the first three oscillators, you could simply increase the number of unison voices and set the oscillator pitch of that voice only.

### **Osc Fine**

The fine tuning of the oscillator in cents. Cents are a fraction of a semitone (+50 equals half a semitone, +100 a full semitone). Modulate this parameter using a LFO as a source in order to obtain vibrato effects. For strong vibratos spanning a larger pitch range, use Osc 1/2 Semi as a destination instead.

The modulation can be restricted to specific voices if desired. For example, you could have a static, low-pitched sine wave on the first voice and a vibrating sawtooth on the second.

### **Osc Volume**

The mixer volume of the oscillator.

### **Osc Pan**

The mixer panorama of the oscillator.

### **Osc Init Phase**

This parameter sets the absolute starting phase of the oscillator whenever a note is triggered. As a consequence, it will no longer be in free-run mode. Set to zero, the specified oscillators starts at zero phase.

An amount of +50 corresponds to a +180 degree phase shift, an amount of +100 to +360 degrees, etc. While the human hearing is largely insensitive to the starting phase of a single oscillator, the relative phase difference between both oscillators can matter in some cases, for example when both oscillators share the exact same frequency.

### **Osc Keytrack**

This parameter sets the key tracking of the oscillator and is only available in the modulation matrix. An amount of +100 corresponds to the regular key tracking, where each MIDI key number corresponds to one semitone. By changing this parameter to values other than +100, you can either realize strange tunings or turn off key tracking entirely by setting KeyTrk to 0. This is often done for drum sounds, which have no defined pitch.

### **Osc Detune**

Modifies the detune parameter of the oscillator stack. Oscillator density must be higher than 1 for this to work.

### **Osc Amount**

Modifies the amount parameter (located next to "Density" on the top left). Oscillator density must be higher than 1 for this to work.

### **Osc Spread**

Modifies the spread parameter. Oscillator density must be higher than 1 for this to work.

### **Osc VA Pulse Width**

When the oscillator is set to "Pulse" in VA mode, this parameter changes the pulse width.

### **Osc VA Sync Freq**

When the oscillator is set to VA mode with Sync enabled, this parameter changes the synchronization frequency.

### **Osc WT Position**

When the oscillator is set to WT mode, this parameter sets the Wavetable position. This allows to sweep through the wavetable by applying a LFO or MSEG.

### **Osc FM Amt A**

When the oscillator type is set to FM mode, this changes the amount of operator A.

### **Osc FM Amt B**

Changes the amount of operator B.

### **Osc FM Amt C**

Changes the amount of operator C.

### **Osc FM Feedbk**

Changes the amount of operator feedback.

### **Osc FM Ratio A**

Changes the operator A ratio. Note this destination is quantized, just like the corresponding ratio knob. For smooth changes, use the Amount destination instead.

### **Osc FM Ratio B**

Changes the operator B ratio. Note this destination is quantized, just like the corresponding ratio knob. For smooth changes, use the Amount destination instead.

### **Osc 3 Sample Pos**

Allows to set the current sample position for WAV playback, between the first and last sample of the wave file. Note that in order to specify a sample offset when starting a WAV (e.g. a random starting offset to make the sound less static), the Osc 3 Init Phase destination must be used instead.

### **8.3.3. Ring Mod / Noise**

#### **Ring Mod Volume**

Adjusts the mixer volume of the ring modulator.

#### **Ring Mod Pan**

Adjusts the mixer panorama position of the ring modulator.

#### **Noise Volume**

Adjusts the mixer volume of the noise generator.

#### **Noise Pan**

Adjusts the mixer panorama position of the noise generator.

#### **Noise Lowpass**

Changes the low-pass filter of the noise generator.

#### **Noise Highpass**

Changes the high-pass filter of the noise generator.

## **Noise Width**

Changes the stereo width of the noise generator.

### **8.3.4. Filter**

This section describes the filter destinations.

#### **Filter 1 Cutoff**

Adjusts the cutoff frequency of the first filter relative to the front panel knob and using the same value range.

#### **Filter 1 Reso**

Adjusts the resonance for the first filter relative to the front panel knob, using the same value range.

#### **Filter 1 Keytrack**

Sets the key tracking for the first filter, which specifies how much the cutoff frequency changes according to which MIDI note number a voice is playing. The parameter works in the same manner as the front panel knob, however you may specify negative values, too. Negative values cause low keys to have a higher cutoff frequency than high keys.

#### **Filter 1 Drive**

Adjusts the drive parameter for the first filter.

#### **Filter 1 Env Amt**

Adjusts the envelope amount for the first filter relative to the front panel knob, using the same value range.

**Filter 2 Cutoff**

Adjusts the cutoff frequency of the second filter.

**Filter 2 Reso**

Adjusts the resonance for the second filter.

**Filter 2 Keytrack**

Sets the key tracking for the second filter.

**Filter Balance**

Changes the filter 1/2 balance.

**Filter 2 Drive**

Adjusts the drive parameter for the second filter.

**Filter 2 Env Amt**

Adjusts the envelope amount for the second filter.

**Filter FX Amount**

Changes the filter effect's "Amount" knob.

**Filter FX Lowpass**

Most filter effects (esp. Distortion), are followed by a low-pass filter to roll off unwanted high frequencies. This matrix destination allows you to modulate this parameter.

### **Filter Attack, Decay, Sustain, Release**

Adjusts the filter envelope parameters relative to whatever is set on the front panel. This allows you to specify different filter envelopes for different voices or to dynamically vary those parameters for each note.

### **8.3.5. Amplifier**

This section describes the amplifier destinations.

### **Amp Attack, Decay, Sustain, Release**

Adjusts the amplitude envelope parameters relative to whatever is set on the front panel. This allows you to specify different amplitude envelopes for different voices or to dynamically vary those parameters for each note.

### **Amp Level**

Adjusts the amplitude envelope level.

### **Amp Pan**

Adjusts the panorama position of the amplitude envelope. This parameter has no corresponding knob on the front panel; it allows you to control the panorama of the entire voice.

### **8.3.6. MSEG**

This section describes the available destinations for the graphical envelopes, MSEG 1-4.

### **MSEG Rate**

Adjusts the rate of the MSEG. Note that MSEG Sync must be turned off for rate modulation to work.

### 8.3.7. LFO

This section describes the available destinations for low frequency oscillators 1-3.

#### **LFO Amount**

Adjusts the amount of the specified LFO.

#### **LFO Offset**

Adjusts the offset of the specified LFO.

#### **LFO Rate**

Adjusts the rate of the specified LFO.

#### **LFO Init Phase**

Adjusts the initial phase of the specified LFO when a new voice is started. This parameters works just like oscillator init phase.

### 8.3.8. Mod Matrix (MM Amount 1-32)

The MM Amount destinations allow you to change the modulation amount of any modulation slot. This feature can be useful to build complex modulations. For instance, if LFO 1 modulates the noise level using MM slot 1, then you could have LFO 2 modulate that slot by choosing LFO 2 as a source, and MM Amount 1 as a destination.

Another useful application is to make modulations dependent on a MIDI controller, e.g. you program a modulation envelope and wish to have the modulation depth controllable by the modulation wheel. In that case you would choose Mod Wheel as a source and the MM slot(s) containing the modulation envelope as a destination.



You may also have one MM slot modulate another MM slot, which in turn modulates another MM slot etc. For such modulations to work properly, you must ensure that the MM Amount numbers occur in ascending order.

### **8.3.9. FX Bus 1 and Bus 2**

This section describes the effect destinations. Almost all destinations correspond to their front panel knobs (refer to chapter 4. *Sound Parameters* for a detailed description of the parameters), only differences will be explained here.

#### **Dist**

Adjusts the corresponding distortion parameters.

#### **EQ 1/2**

Adjusts the equalizer parameters. All parameters can be modulated, except switches and the low- and high shelving filters.

#### **Phas**

Adjusts the corresponding phaser parameters.

#### **Chrs**

Adjusts the corresponding chorus parameters.

#### **Del 1/2**

Adjusts the corresponding delay parameters. When delay sync is turned off, the delay time may be modulated, relative to the L/R times specified with the front panel knobs.

**Rev**

Adjusts the corresponding reverb parameters. All parameters except pre-delay can be modulated.

**Comp**

Adjusts the corresponding compressor parameters.

**FX Chain Pan**

This destination sets the stereo panorama of an entire effect bus. As an example, a reverberated sound could be sent to just the left channel. When the FX Chain Pan destination is in use, typically both FX busses should be enabled on the front panel.

## 9. MIDI Reference

Most knobs and buttons on the front panel can be remote controlled via MIDI. DUNE's default controller assignments follow common conventions and the MIDI standard as much as possible. The number of sound parameters DUNE 3 offers, however, is higher than the amount of available MIDI controllers. Most effect parameters and large parts of the modulation matrix are thus not assigned to any MIDI controller.

To override the default MIDI assignment, right-click on a knob and select "MIDI Learn", then move your hardware controller knob. To remove an assignment, choose "MIDI Forget". Once you close DUNE, the changes will be made persistent and apply to any future instance of DUNE.

Parameter	CC#
<b>Common</b>	
Modulation Wheel	1
Glide	5
Sustain Pedal	64
Volume	111
<b>Unison</b>	
Detune	53
Pan Spread	54
<b>Oscillators</b>	
Osc 1 Detune	102
Osc 1 Spread	30

Parameter	CC#
Osc 1 Semi	77
Osc 1 Fine	24
Osc 1 PW	70
Osc 1 Level	75
Osc 2 Detune	103
Osc 2 Spread	31
Osc 2 Semi	78
Osc 2 Fine	34
Osc 2 PW	71
Osc 2 Level	76
Osc 3 Semi	105

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Parameter	CC#
Osc 3 Level	106
Noise LP	107
Noise HP	108
Noise Width	109
Noise Level	110
Ring Mod Level	79
<b>Arpeggiator</b>	
Arp 1 On/Off	47
Arp 1 Rate	50
Arp 1 Length	52
<b>LFOs</b>	
LFO 1 Rate	14
LFO 1 Waveform	15
LFO 1 Sync	16
LFO 1 Fade	18
LFO 2 Rate	19
LFO 2 Waveform	20
LFO 2 Sync	21

Parameter	CC#
LFO 2 Fade In	23
LFO 3 Rate	80
LFO 3 Waveform	81
LFO 3 Sync	82
LFO 3 Fade	84
<b>Filter</b>	
Filter Attack	38
Filter Decay	39
Filter Sustain	40
Filter Release	41
Filter 1 Cutoff	74
Filter 1 Resonance	42
Filter 1 Env Amt	43
Filter 1 Key Track	46
<b>Amp Envelope</b>	
Amp Attack	73
Amp Decay	36
Amp Sustain	37

## MIDI Reference cdn.

Parameter	CC#
Amp Release	72
<b>Effects (Bus 1)</b>	
EQ 1 Band 1 Gain	89
EQ 1 Band 2 Gain	90
Reverb Dry/Wet	91
Delay 1 Dry/Wet	92
Chorus Dry/Wet	93
Distortion Dry/Wet	94
Phaser Dry/Wet	95
<b>Mod Matrix</b>	
MM Amount 1	112
MM Amount 2	113
MM Amount 3	114
MM Amount 4	115
MM Amount 5	116
MM Amount 6	117
MM Amount 7	118
MM Amount 8	119

## 10. Optimizing performance

The most important switch to boost performance is to enable the multi-threading switch (located in the MAIN section of the center screen). Additional performance gains can be achieved by optimizing your system or by adjusting the patch settings.

### Optimizing your system

If you use a Windows-based system, often many services run in the background which can degrade performance – check if you really need all of them. If you run DUNE 3 on a modern quad-core or better CPU with multi-threading enabled, you should see a substantial reduction in CPU usage, especially when playing pad/string-type sounds with a high polyphony and voice count. If you do not experience a significant performance boost with multi-threading, then either your host's audio buffer size is too small (the minimum is 128 samples), or your system is not optimized enough for realtime audio playback.

### Optimizing patch settings

The two most important patch settings affecting CPU usage are polyphony (how many notes can be active simultaneously), and the modulation rate.

### Polyphony

The polyphony can be adjusted on the top right of the user interface, inside the MASTER section. Set the polyphony as low as possible. If you start to hear clicks, you have two options:

- Shorten the amplitude envelope release time. This will reduce the overlap of successively played notes, and thus lower the required polyphony.
- Increase the polyphony again, until the clicks disappear.

## Modulation Rate

The modulation rate can be adjusted in the SETTINGS tab, inside the center screen. Set the modulation rate to "Normal" for best performance. Use "Fast" or "Audio-rate" only if you are certain the patch requires this rate, and if you can really hear a difference. Audio-rate modulation needs a lot of CPU, as the entire synth engine, plus all modulation sources (e.g. LFOs, MSEGs, ...) are processed sample by sample, rather than in blocks.

## Patch structure

If you design sounds from scratch, here is a few tips that will help you reduce CPU usage:

- Oscillator stacks 1+2 can be turned off by setting the oscillator count to zero. This is strongly recommended if you do not need them.
- Avoid using more unison voices than necessary. For example, using 8 unison voices with a single oscillator per voice is much more CPU intensive than using a single unison voice with a stack of 8 oscillators in it. DUNE 3 has stereo oscillators with pan + pan spread, it is thus not necessary to use multiple unison voices to achieve a stereo sound.
- Set the filter to "Off" for patches that do not require a filter.
- Use the more CPU-demanding filters like the analog-modeled filters only if you really need them. They use much more CPU than the corresponding multi-mode filters.

## 11. Troubleshooting

If DUNE 3 does not work as expected, check the following points. If your problem is not listed here, do not hesitate to contact us at [service@synapse-audio.com](mailto:service@synapse-audio.com), we are happy to assist you.

### 11.1. Notes click when pressing or releasing a key

Check the amplitude attack and amplitude release time, and increase both if necessary. Furthermore, check the polyphony setting (top right of the user interface) and increase the polyphony if necessary.

### 11.2. Patches do not recall

The most likely reason is that you have the demo version installed. If you have purchased the full version, make sure to uninstall the demo version then install the full version.

### 11.3. Parameters change unexpectedly

Check your MIDI setup. While DUNE 3 uses a controller map designed to not react on controllers such as bank change or volume (which may be sent by some hosts), it is possible that your gear sends other controller messages that DUNE 3 will respond to. The default controller assignments are given in the chapter *9. MIDI Reference*.

### 11.4. The sound is distorted

Lower the master volume, and watch out for the clipping indicator in your host sequencer. If this does not help, check your audio driver settings and the CPU load – if the CPU load is too high, clicks or other artifacts will occur. Refer to chapter *10. Optimizing performance* for tips on how to reduce the CPU usage.



If none of the above helps, check if the active patch intentionally uses distortion, e.g. by switching to other patches and comparing the output. There is many ways to intentionally add distortion to a patch. The most obvious way is the distortion effect in one of the master effect busses. The filter, filter effect as well as the compressor can distort, too, however.

### **11.5. MIDI messages are not received**

Check the MIDI IN indicator, located on the top right of DUNE's user interface. If this indicator never lights up, DUNE 3 does not receive any MIDI messages. Check the MIDI setup of your host sequencer, as well as the setup of your hardware.

### **11.6. The CPU usage is very high**

Check if your system meets the minimum system requirements. Check if Multi-threading is disabled, if yes try enabling it. Also read chapter *10. Optimizing performance*.

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## Appendix. Sound Design Reference

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IW	Ingo Weidner	<a href="mailto:ingo_weidner@web.de">ingo_weidner@web.de</a>
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MK	Michael Kastrup	<a href="http://www.xsynth.com">http://www.xsynth.com</a>
PK	Piet Kaempfer	<a href="http://www.protonica.de">www.protonica.de</a>
RH	Richard Hoffmann	<a href="http://www.synapse-audio.com">www.synapse-audio.com</a>
RL	Rob Lee	-
SDX	Shadx	-
ST	Solidtrax	<a href="http://www.solidtrax.nl">http://www.solidtrax.nl</a>
TK	Teksonik	<a href="mailto:teksonik@outlook.com">teksonik@outlook.com</a>

If you wish to contribute patches or banks to future versions of DUNE, do not hesitate to contact us.