DUNE 3

User's Manual

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VST COMPATIBLE

SYNAPSE AUDIO SOFTWARE
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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 personal computer. It was developed with the highest possible standard of audio quality in mind and to offer exceptional flexibility in sound design tasks. The latter is achieved via its modulation system, the new Differential Unison Engine (DUNE) - hence the name. Despite the complexity of the synthesizer, fast vector processing and support for multiple processor cores both allow 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, as they all run fine in parallel. Furthermore, note that DUNE 3 is patch-compatible with DUNE 2. This means that patches from DUNE 2 can be fully usable in DUNE 3. The best way to do this is to simply move your patches/folders from DUNE 2's Soundbanks folder to DUNE 3's Soundbanks folder.

Installation on Windows

Unzip dune3win.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. If you are unsure about where the host software's VstPlugins directory is located, then refer to your host software's manual.

All content files (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 "dune3mac.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 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 immediately 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 method 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:

https://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 64-bit VST or AU-compatible host. If you encounter any compatibility issues with your host software, do not hesitate to contact us (service@synapse-audio.com). DUNE 3 has been tested using the following hosts:

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

1.4. System Requirements

To maximize sound quality, DUNE 3 employs complex DSP algorithms that can be 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. At a minimum, the following is necessary to run the software optimally:

- MACs require OS X 10.11 or later, and should be equipped with M1, M2, M2 Pro, M2 Max or Intel 2,8 GHz quad core processor or better.
- PCs require Windows 7 SP 1, and should be equipped with a 2,8 GHz quad core CPU or better. Most importantly, Windows must be optimized for realtime audio 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's important to choose a good latency/audio buffer size. We recommend using between 10-20ms, 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 synchronization overhead will become too significant.

The memory requirement of DUNE 3 is approximately 200 mb per instance. If you wish to run many instances, your system should be equipped with 8 GB of memory or more.

1.5. Tutorials

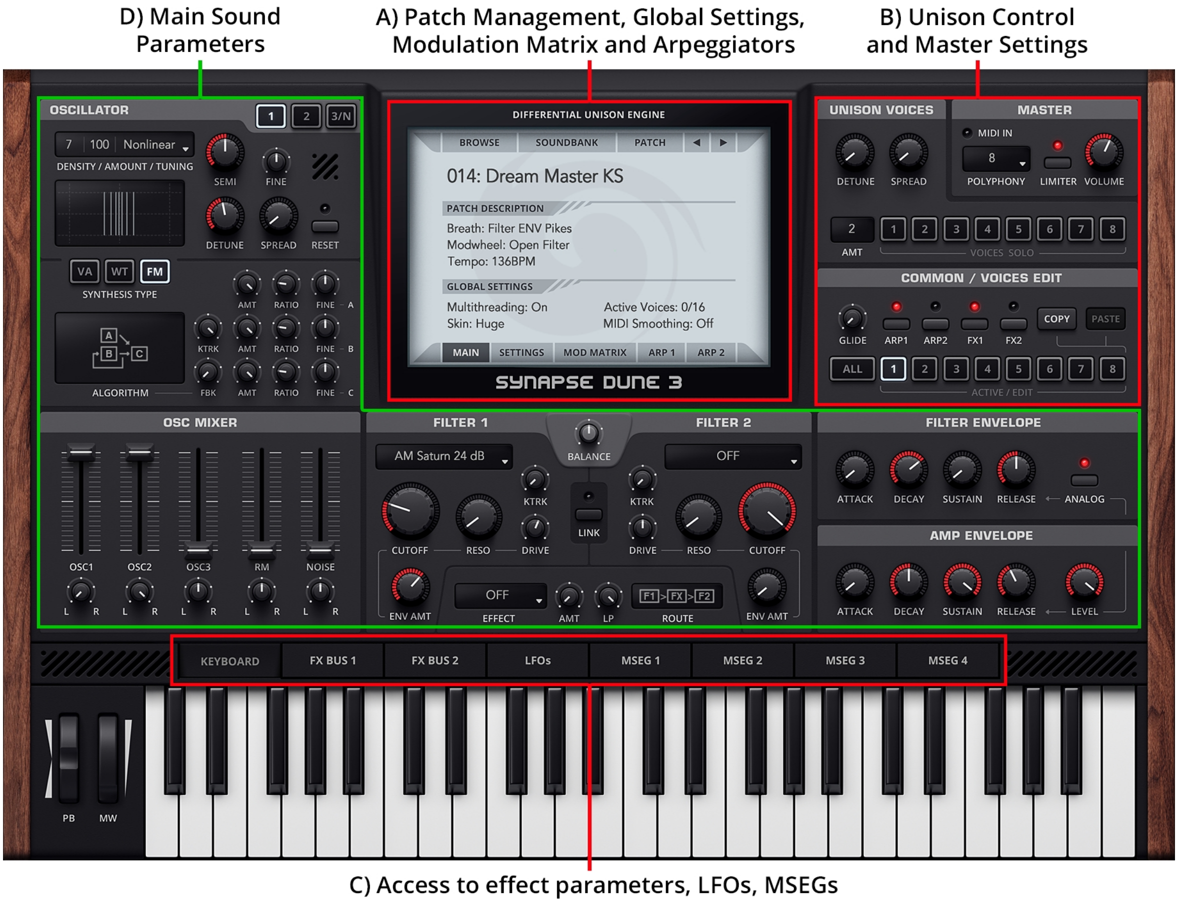
In addition to reading this manual, consider watching our DUNE 3 tutorials and workshops on our Youtube channel:

https://www.youtube.com/@synapseaudio2488

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 arpeggiators. The center screen is covered in chapter *3. The Center Screen*.

The right side (B) of the interface contains the master section with global volume and the polyphony setting, as well as the unison voice controls. DUNE 3 allows the user to directly edit the eight unison voices, either all at once, or individually. This makes DUNE 3 a multi-part synthesizer, which facilitates the creation of 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 a vertical direction (see fig. 2.2).

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

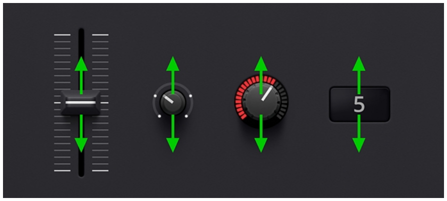


Figure 2.2. Controlling parameters.

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. The mouse wheel can be used in drop-down lists to select the previous or next parameter in the list. Knobs and faders can be fine-tuned using the mouse wheel, and in the graphical envelope editors (MSEG 1-4) it zooms the view in or out.

2.3. MIDI Learn

Right-click on a parameter, then choose "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 and choosing "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

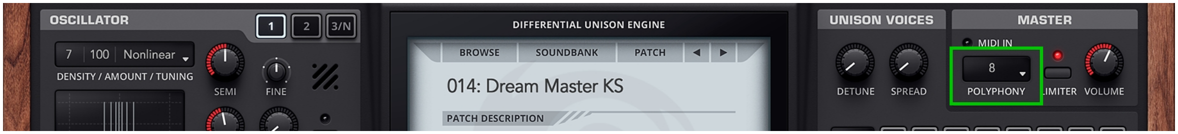
Select a patch by clicking on the patch name in the center screen. A more detailed description of the patch management is given in chapter *3. The Center Screen*.

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. 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.

2.6. Adjusting Polyphony

DUNE 3 allows for up to 24 note polyphony. The "Polyphony" drop-down menu in the master section on the righthand side of the center screen 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 is displayed in the MAIN section of the center screen). Since each voice costs processor time, it's a good idea to limit polyphony as much as possible.

When choosing Mono or Legato, only one note can be played at a time. Legato mode glides notes smoothly from one to the other 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 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 at times require adjusting polyphony.

2.7. Pitch Bend and Modulation Wheel

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

2.8. Microtuning

DUNE 3 supports microtuning using the MTS-ESP standard. With the free MTS-ESP plugin, which can be downloaded from https://oddsound.com/mtsespmini.php, DUNE 3 can automatically use any scale. The microtuning is set in the MTS-ESP plugin. Afterwards, the microtuning is recognized immediately by DUNE 3 and any other plug-in instances in the project, that support the MTS-ESP standard.

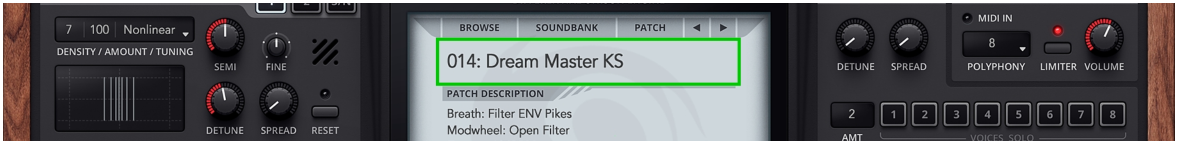
3. The Center Screen

This chapter explains all parameters contained in 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 is played on DUNE 3 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 roughly 1000 patches. The patches are sorted into categories like Bass, Lead, Pads, SFX, etc. The default bank shown is "DUNE 3", which contains the latest patches that have been 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 saved as a single Cubase .FXP file, a common format for storing patches.

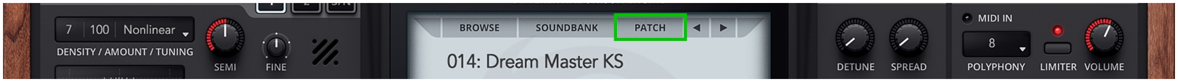
Any directory contained within the Soundbanks directory represents a bank. In this way, the user can easily organize patches and banks within the Soundbanks directory.

Below the patch name, the PATCH DESCRIPTION section provides information about the currently selected patch. For example, the PATCH DESCRIPTION for a chosen patch might describe 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 while editing a patch, don't panic! The Undo/Redo functions are provided so that the user can recall or undo changes made to a patch.

Patch menu

The patch menu is invoked by clicking on "PATCH". The specific functions within the patch menu and their uses are listed below.



- **Load Patch** loads a particular patch from disk. This is useful for browsing patches outside of Dune's Soundbanks folder. Nevertheless, we recommend storing patches in your Soundbanks folder to simplify the process of browsing a whole bank.
- **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 your changes will be lost. 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 asterix (*). After saving the patch, the asterix is removed.
- **Copy FX 1/2** copies all effect bus 1/2 parameters into the clipboard. By copying the effect parameters, the user can apply them to a different patch or other DUNE 3 instances.
- **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's useful to start with a "minimal" patch with an empty modulation matrix, effects turned off, etc. This 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** restores the patch to previous edit states. For instance, if you accidentally switch the current patch during editing, simply click Undo to restore the previous patch you had 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. It's important to set the buffer size/audio latency in the host sequencer to at least 512 samples (at 44.1/48 kHz) or 10 ms, for best results. Depending on the computer and the host sequencer used, multithreading may have no effect, or may 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. If you work on an older PC with a low-resolution display, choose the Small skin. For monitors with a very large resolution (4K, 5K and above), use Large/Huge or download the optional UHD skins from your user account.

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 greatly velocity will affect the volume of a patch. Set to zero, velocity has no effect on volume. Positive values cause higher velocities to increase volume. By contrast, negative values invert the effect, with higher velocities decreasing volume.
- **Pan** adjusts how greatly velocity affects the stereo position. Set to zero, velocity has no effect on panning. Positive values move the sound to the right as velocity increases, negative values move the sound to the left.
- **Filter** adjusts how greatly velocity affects the filter cutoff. Set to zero, velocity has no effect on the filter. Positive values cause higher velocities to increase the filter cutoff frequency, whereas negative values invert the effect.
- **Env Amt** adjusts how greatly 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 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 a vertical direction.

By default, all MIDI velocity values control the destination in a linear manner (1:1).

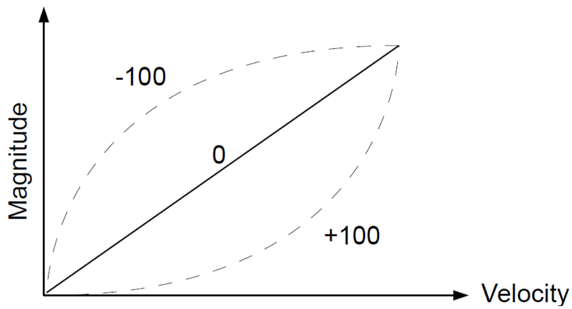


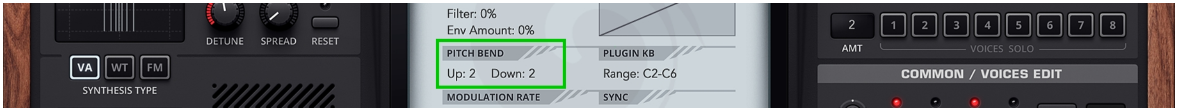
Figure 3.2. Velocity Curve.

For example, an incoming velocity value of 80 has twice the effect compared to a velocity value of 40. By setting the curve parameter, you can change this 1:1 mapping to have a nonlinear behavior. A value of zero corresponds to the default (linear) scale.

The curve parameter is useful for fine-tuning the velocity response to accommodate your personal playing style, as well as your MIDI keyboard.

Pitch Bend

The Bend up/down parameters specify the number of semitones a sound is pitch shifted when the pitch bend wheel is turned all the way up or down.



Click on the numbers next to either "UP" or "DOWN" and drag the mouse up or down to increase or decrease the pitch bend range.

If a MIDI keyboard is connected to your computer, turning the physical pitch bend or modulation wheel should automatically turn the same wheel in DUNE 3. If this is not the case, then the required MIDI information is not transmitted to the plugin. Refer to your host software's manual to address this issue.

Modulation rate

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

- The **Normal** rate is sufficient for most uses and is also the best choice for the majority of sounds.
- The **(Very) Fast** rate is useful when you use rapid LFO modulations (>100 Hz), or when you use MSEGs with fast envelopes.

- **Audio Rate** mode processes the entire synth engine sample by sample. As an example of how this can be useful, oscillators can be used as modulation sources to accurately modulate any (!) sound parameter.

Note that audio rate modulation is highly CPU-intensive and should be used only when necessary. Also note that a higher modulation rate does not correlate to better sound. If you can't hear a noticeable difference, then don't use a higher modulation rate, as this will only 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 the modulation matrix is to connect MIDI controllers, LFOs, and envelopes with DUNE 3's sound parameters. 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 the source will 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%, creating a tremolo effect.

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

For ease of use, each slot contains a crosshair in between the DESTINATION and VOICE sections. These are provided to streamline the process of assigning a modulation destination. Simply drag the crosshair from one of the 32 available slots by holding the left click button on it and drop it on the desired destination. All available modulation destinations on the user interface will light up when holding one of the crosshairs.

Note that only one modulation source and destination can be chosen for each slot. This means that if you want to modulate multiple destinations with the same modulation source, then you must choose the same modulation source in another available slot and assign it to the desired destination again. If a destination has already been chosen for a slot, then dragging the crosshair onto a new destination will cancel the previously chosen destination in favor of the new one.

To get the most out of the modulation matrix, it's important to first understand how LFOs and envelopes work, and what the sound parameters do. Hence, the modulation matrix, with all its source and destination parameters, is described more thoroughly in chapter *8. Modulation Matrix,* after the sound parameter chapter.

3.4. Arpeggiator

An arpeggiator (short: ARP) is a module that generates melodic or rhythmic patterns from one or more pressed keys. This is achieved with the help of a step sequencer that contains 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 COMMON/VOICES EDIT section, individually per unison voice.

Type

The arpeggiator can use either the built-in step sequencer or midi files as a basis of operation. The user can choose from either of these options by clicking on 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 3. Click on buttons 1-8, 9-16, 17-24 or 25-32 to simultaneously edit eight steps.

Each step has three parameters: note, velocity, and tie. A non-zero note value transposes a step up or down by the given number of semitones. For example, +12 corresponds to one octave above the input note pitch. The velocity value is transmitted to the modulation matrix, where it can be used as a modulation source ("Arp Velocity") to change the volume or timbre of each step.

The tie button glues two steps to each other, which provides the option of either sliding from one note to another, or to simply increase the length of the note in that step. A rest can be programmed by turning the note value all the way down, until three dashes ("---") appear.

MIDI mode allows for the use of standard MIDI files as a basis of operation. The files should contain only a single track, and they should be monophonic. One exception is when the "Playback" mode is chosen. In this mode, polyphonic MIDI files can also be played back.

Mode

The arpeggiator module in DUNE 3 comprises 9 different modes. These are 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 as up, 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 operates like Alt Up, but in reverse order.
- **Random** mode cycles through all keys pressed, in random order.
- **Chord** mode chops up a chord into a rhythmic pattern, according to what's programmed in the step sequencer or MIDI file.
- **Playback** mode plays back the sequencer notes (or MIDI file), transposed according to the currently pressed MIDI key. Pressing more than one key simultaneously has no effect in this mode. Playback mode is the only mode that allows the user to play back polyphonic MIDI files.
- **Silent** is a special mode, whereby the arpeggiator doesn't trigger any notes. The purpose of this mode is to use the arpeggiator exclusively for modulation. This can be accomplished by choosing "Arp Note", "Arp Velocity" or both as a source in the modulation matrix.

Octaves

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 play all keys pressed, plus the same keys one octave higher, etc. DUNE 3's arpeggiator allows for a maximum of 4 octaves.

Steps/Bars

When in Step Sequencer mode, Steps sets the pattern length in steps. When the arpeggiator reaches the end of the pattern, it will automatically restart at the beginning.

When MIDI mode is selected, this parameter sets the number of bars to use. It's possible to choose up to a 32-bar sequence. The arpeggiator can also loop MIDI sequences.

Rate

Adjusts the tempo of the arpeggiated sequence. When SYNC is enabled, the tempo is synced to the host sequencer and can be specified in musical intervals such as 8th notes, 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 take on a staccato feel, while the opposite direction yields smoother sounding sequences.

Swing

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

Adjusts the time it takes from the beginning of a slid note to reach its target pitch. To have any audible effect, TIE must be used on a note, and at least two different note numbers must be used during the tie.

Note that Slide only works in Step Sequencer mode, and not in MIDI mode.

4. Sound Parameters

This chapter describes how a patch is constructed. This includes the operation of the knobs and switches on the front panel, the effects 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, with the exception of the Modulation Matrix (which will be covered in detail in chapter *8. Modulation Matrix*).

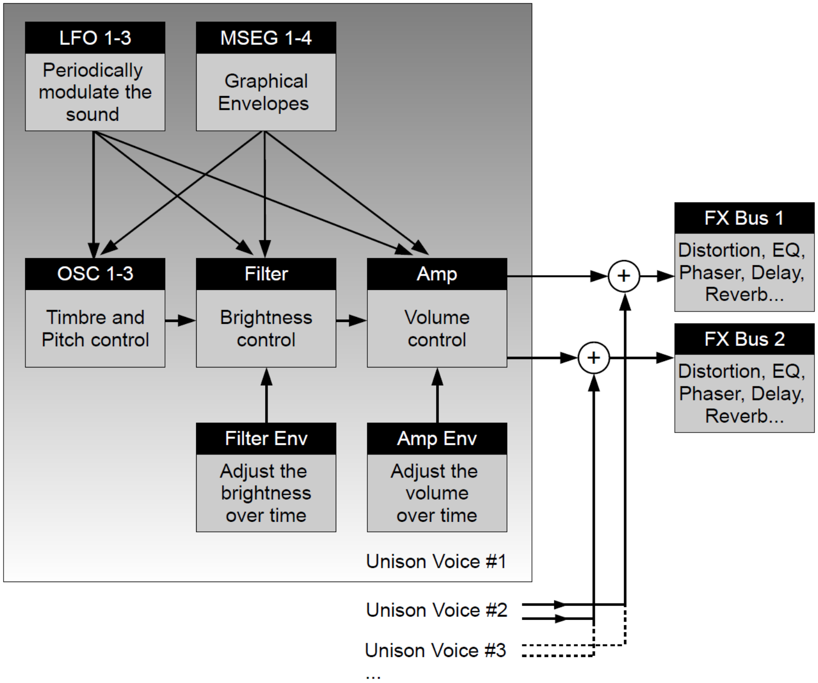


Figure 4.1. Structure of a DUNE 3 Patch.

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 equalization, 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"). These 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. Most often, high frequencies are removed. Hence, this type of synthesis is commonly called "subtractive," as it relies on removing (subtracting) unwanted frequencies. The final volume block controls the volume of the signal.

On their own, the three basic building blocks synthesize a completely static sound. This contrasts with acoustic sounds, where pitch, timbre, and volume change over time. To emulate this effect in a synthesizer, envelopes are used to add dynamic variations to a sound.

The most important envelope in synthesis is the amplitude envelope ("Amp Env"). The Amp Env is essential for controlling the volume of a sound over time. It's what makes a synthesizer playable like an organic instrument. The filter envelope ("Filter Env") is also highly important in sound design. It's what allows us to dynamically vary the brightness, and thus the timbre, of a sound over time. In addition to these two envelopes, DUNE 3 offers four freely programmable graphical envelopes ("MSEGs"), which can be linked to any sound parameter.

While envelopes control the overall progression of a sound, it's sometimes desirable, and often necessary, to add periodic modulations. Such modulations can mimic vibrato or tremolo effects produced by acoustic instruments. These effects can be emulated by using one or more of the low frequency modulation ("LFO") blocks.

4.2. Oscillators 1+2

An oscillator generates a periodic waveform. Oscillators form the basic building blocks of most synthesizers. The most common waveforms are illustrated in fig. 4.2.

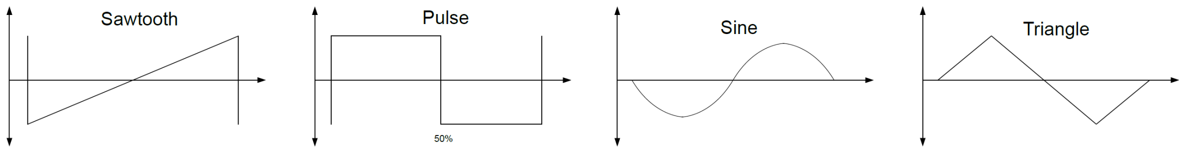


Figure 4.2. Basic oscillator waveforms.

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

DUNE 3 offers three oscillators and a separate noise generator. This corresponds to how traditional synthesizers work.

In DUNE 3, the first two oscillators are stacks of up to 32 oscillators each, with adjustable DETUNE and stereo SPREAD controls. This allows for the creation of thick pad, bass or lead sounds with just one oscillator.

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-clicking/dragging on the DENSITY field in the top-left of the oscillator section. When set to zero, the oscillator stack is turned off, which saves processor time.

AMOUNT

When choosing more than two oscillators with the density control, the amount (AMT) parameter allows the user to modify the level of the oscillators.

The behavior of this parameter depends on the selected tuning mode, but it usually adjusts the level of the oscillators around the center (which always remains at maximum).

TUNING

The chosen tuning mode affects the 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 (a bell-shaped curve) for the volume of the oscillators. The curve 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 modulated individually. This mode adds an extra RATE knob, which controls the rate of modulation.

SEMI

Adjusts the coarse tuning of the oscillator stack in semitones. The range spans +/- 36 semitones, or three octaves above or below the note input pitch. A larger range can be obtained by using the modulation matrix, if required. This will be covered in chapter *8. Modulation Matrix*.

FINE

Adjusts the fine tuning of the oscillator stack in cents. A value of +/- 100% corresponds to half a semitone.

DETUNE

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

SPREAD

When two or more oscillators are active in the stack, the SPRD knob spreads the oscillators in the stereo field around the center. Turned fully to the 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, the initial phase is set to zero. This can be changed in the modulation matrix.

Setting the initial oscillator phase can be useful for obtaining more control of the transient of a sound. When using more than one or two oscillators, note 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's also 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

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 pulse width is 50%, which corresponds to a square wave.

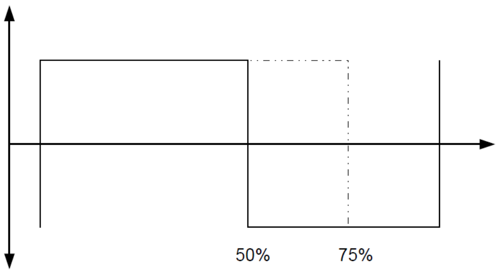


Figure 4.2.2. Pulse Width.

All three waveforms offer oscillator synchronization, which is enabled by pressing the SYNC switch. Oscillator sync causes an oscillator's phase cycle to be periodically reset to zero phase by a second "leader" oscillator. Whenever the leader oscillator has completed a cycle, it resets the "follower" oscillator. If the leader oscillator has a lower frequency than the follower, the result is a new timbre, as the follower oscillator isn't 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 leader oscillator needed to create the synchronization effect. 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

The Wavetable (WT) mode lets the user 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. The waveform can also be set via the POSITION knob. The position knob traverses the entire wavetable and can specify positions between two separate waveforms for even more versatility. The waveform position can 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 INTERPOL (Interpolation) knob.

By clicking on the wavetable popup menu, different wavetables can be selected. Wavetables can have a varying number of waveforms stored in them, ranging from between 3 to 64 waveforms. Custom waveforms can also be created via the EDIT button under the wavetable popup menu.

WAV files can be dragged and dropped directly onto the wavetable display or below the wavetable display next to the available waveforms. This is a fast and easy way to build wavetables from scratch. Note that it's possible to select different wavetables for different unison voices.

FM

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

To create complex and interesting sounds, it's 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 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.

It can also be used to load waveforms by simply dragging and dropping the waveform onto the display containing the WAVEFILE name.

SEMI

Adjusts 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

Adjusts the fine tuning of the oscillator in cents. 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. An exact initial phase angle can be set in the modulation matrix.

4.4. Noise Generator

All three oscillators generate periodic waveforms with certain pitches. Sometimes a sound with a random element that has no fixed pitch can be useful. These types of sounds can be useful for synthesizing percussive sounds, recreating the behavior of wind or plucked string instruments during transients, or for synthesizing natural sounds like fire, water, or wind.

To facilitate the creation of such sounds, DUNE 3 offers a NOISE generator.

LOW PASS

The white noise emitted by the noise generator passes through a low-pass and high-pass filter to shape the timbre of the noise. The LOWPASS (LP) knob controls the cutoff frequency of the low-pass filter. Lower values correlate to darker timbres.

HIGH PASS

The HIGH PASS (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 its minimum value (0%), the noise sounds monophonic. Set to its maximum value (100%), both channels are provided with independent noise sources, resulting in a wide stereo image.

WHITE/PINK NOISE

The most common noise source is White Noise, which is the default option in DUNE 3. White Noise has a flat frequency spectrum. In other words, all frequencies are contained equally.

In some situations, Pink Noise may be preferable. Unlike White Noise, Pink Noise rolls off at 3 dB/octave, so higher frequencies are somewhat attenuated. This why Pink Noise generates a warmer sound than White Noise.

4.5. Ring Modulator

An interesting effect can be achieved by multiplying oscillator stacks 1 and 2 by each other. This can be described as one oscillator modulating the amplitude of another (see fig. 4.5).

The depth of this effect can be controlled via the RM fader in the OSC Mixer section. Set to zero, the ring modulator output is disabled.

Mathematically, the result of ring modulation is that the sums and differences of each signal's frequencies are generated. When the oscillators are detuned, this will generate inharmonic, metallic sounds.

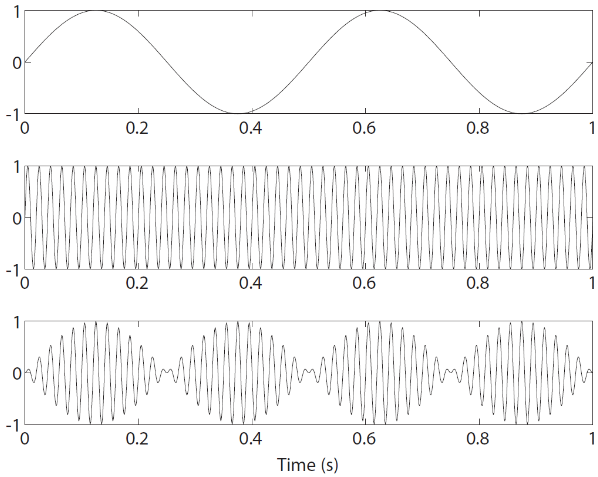


Figure 4.5. Ring modulation

4.6. OSC Mixer

The OSC Mixer section provides control over the level and panorama of oscillators 1-3, the ring modulator and the noise generator, before they enter the filter.

All mixer parameters can also be controlled via the Modulation Matrix. This creates the ability to fade an oscillator in or out, cross-fade between oscillators, modulate the panorama position, etc.

4.7. Filter Section

The raw sound produced by oscillators 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 behavior can be modelled by using a time-varying filter.

DUNE 3 features two identical filters and an insert effect. For maximum flexibility, FILTER 1 and FILTER 2 can be set individually to an arbitrary mode or linked to one another via the central LINK button. Both filters additionally have their own sets of parameters.

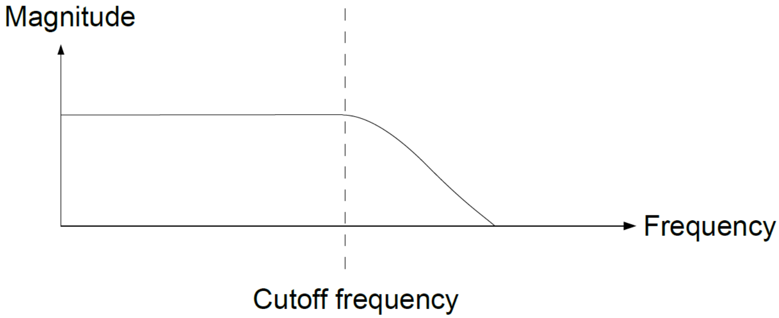
The filters and the insert effect can be routed in six different ways. The routing options will be discussed below in the ROUTE section.

4.7.1. Filter Parameters

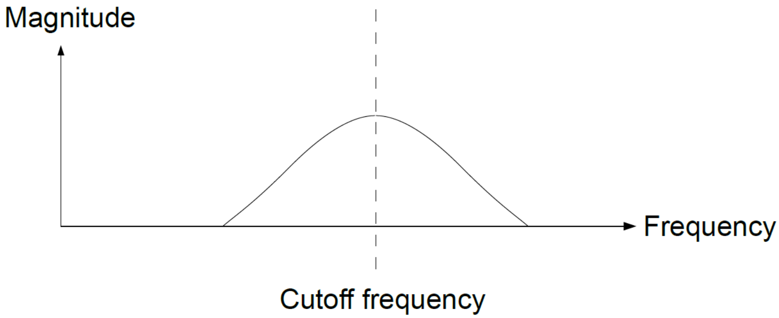
CUTOFF

The CUTOFF knob is perhaps the most important filter parameter. It controls the brilliance, or brightness, of a sound. Its function and behavior depend on the filter type chosen:

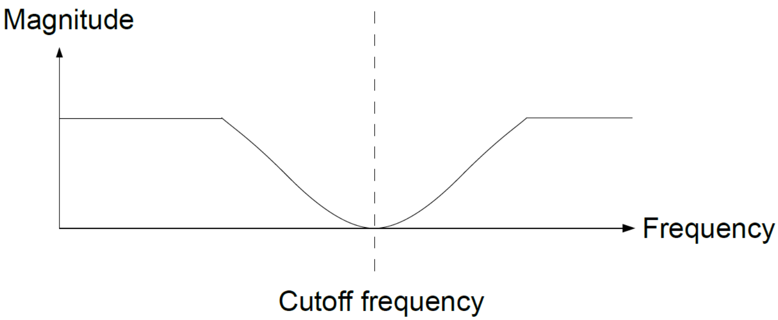
- For low-pass (LP) filter types, frequencies above the cutoff frequency are damped.



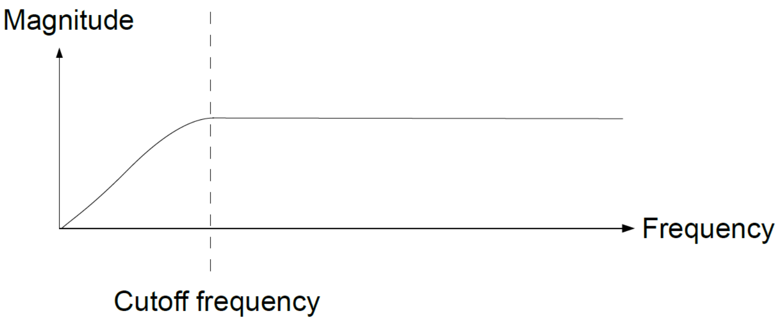
- A band-pass (BP) filter damps frequencies around the cutoff frequency. As a result, bass and treble frequencies are attenuated.



- A notch filter rejects frequencies around the cutoff frequency, but allows all other frequencies to pass.



- A high-pass (HP) filter attenuates all frequencies below the cutoff frequency and allows higher frequencies to pass undamped.



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

The MSEGs, LFOs, and the filter envelope can be used to modulate the CUTOFF frequency and produce a dynamically changing timbre. The most common method of controlling the 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. This is a sinusoidal oscillation near the cutoff frequency (see fig. 4.7.1). The RESO knob controls the depth of this effect.

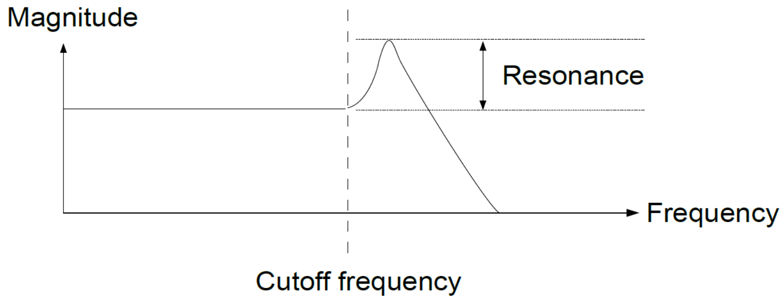


Figure 4.7.1. Sinusoidal oscillation near the cutoff frequency.

At lower settings, resonance can be used to add presence to a sound. Using higher settings, sinusoidal oscillation becomes strong enough to use the filter in a similar fashion as an oscillator.

This effect, known as self-oscillation, can be useful to create special effects sounds like 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.

KTRK

The KTRK knob determines the extent of the MIDI key number's effect on the cutoff frequency. Set to 0%, all notes pressed will share the same cutoff frequency as specified by the CUTOFF parameter.

Nonzero values change the cutoff according to the key pressed, with higher keys corresponding to higher cutoff frequencies. At lower settings, keytracking is useful for creating subtle timbral variations when different notes are played. At higher settings, keytracking can be used to simulate the properties of acoustic instruments that have varying timbres depending on the note played.

4.7.2. Filter Modes

DUNE 3 features multiple different filter models to choose from. All filters except the legacy DUNE 1 filters are zero-delay feedback filters, which more closely resemble the response of analog filters. Most of them can achieve self-oscillation when turning the resonance to high amounts.

Clean Multi-Mode (CL)

The clean filter models resemble analog filters with largely clean direct paths, such as OTA-based ladder filters. Only the feedback path contains mild saturation, to prevent the resonance from ramping to infinity at high resonant levels. The clean nature of these 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**

  Operates like CL Notch but produces even deeper notches in the frequency spectrum.

Expander (XP)

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 and in 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

Comb filters use 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 can be used to keep the timbre stable by setting 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%, as the sound can become very loud.

Analog-Modeled (AM)

DUNE 3's analog-modeled filters are the most accurate reproductions of typical analog filters. Apart from the Alpha filter, the analog-modeled filters use 3x oversampling and feature multiple non-linearities to improve their accuracy. As a result, the analog-modeled filters are computationally more expensive than the other filters in DUNE 3.

For many patches the CL or XP filter types above are sufficient and less CPU-intensive, so care should be taken when choosing one of the analog-modeled filters.

- **AM Alpha 24 dB**

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

- **AM Polaris 24 dB**

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

- **AM Pro 12 dB**

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

- **AM Pro 24 dB**

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

- **AM 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 (BW)

The Brickwall filters in DUNE 3 are special filters featuring very steep slopes. 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, with 36 dB attenuation per octave above the cutoff frequency.

- **BW Lowpass 60 dB**

  A ten-pole lowpass filter, with 60 dB attenuation per octave above the cutoff frequency.

- **BW Highpass 36 dB**

  A six-pole highpass filter, with 36 dB attenuation per octave below the cutoff frequency.

- **BW Highpass 60 dB**

  A ten-pole highpass filter, with 60 dB attenuation per octave below the cutoff frequency.

Transistor Ladder (TR)

The Transistor Ladder filters are exclusively low-pass types and resemble analog transistor-based designs. The key feature of these filters is that they contain saturation in all stages, thus coloring incoming signals, even without any resonance applied. Note that the Transistor Ladder filters require substantially more CPU than the Clean Multimode 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 (SK)

The Sallen-Key filters are two-pole, multi-mode filter designs. What makes them special is the character of the filter's resonance.

It features strong distortion, making for an 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 frequency.

- **SK Highpass 12 dB**

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

Dune 1

DUNE 3 incorporates two filter types from DUNE 1 ("Lowpass 12dB" and "Lowpass 24dB").

These are provided to facilitate creating patches that sound like 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 (BAL) blends seamlessly between Filter 1 and Filter 2. If the knob is turned all the way to the left, only Filter 1 will be audible. Turned fully clockwise to the right, only Filter 2 will be audible. In the middle position, both filters are equally audible.

4.7.4. Link

When the LINK switch is enabled, the parameters of FILTER 1 and FILTER 2 can be controlled simultaneously using the FILTER 1 knobs. For example, when turning the FILTER 1 CUTOFF knob, the FILTER 2 CUTOFF knob will also move.

Note that the FILTER 2 controls keep their relative distance to FILTER 1 (if there is any) prior to moving the knob.

This method of linking filters allows for a flexible workflow.

4.7.5. Insert Effect

DUNE 3 features an insert effect section, which allows for additional 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, which determines the amount of distortion applied. The output of the distortion stage passes through a low-pass filter with an adjustable cutoff frequency. Both the amount of distortion and the lowpass filter cutoff frequency can be modulated via the MOD MATRIX.

- **Hard Dist**

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

- **Foldbk Soft**

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

- **Foldbk Hard**

  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. The sound of foldback distortion 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 AMT 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 like a square wave and creates strong digital artifacts. To tame the artifacts, the lowpass filter on the output can be used to attenuate high frequencies.

- **Ratecrush**

  Ratecrush is another effect that creates strong digital artifacts. In contrast to Bitcrush, Ratecrush employs a sample-and-hold circuit to reduce a sound's sample rate, as opposed to its dynamic range.

- **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, and the LP knob controls the feedback in +/- 100%. A classic sweeping phaser effect can be obtained by modulating the AMT knob position with one of the sources in the modulation matrix.

- **Highpass 1p**

  A one-pole highpass filter with 6 dB attenuation per octave below the cutoff frequency. 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 effects to the main filter and the extra filter.

- **Highpass 2p**

  A two-pole highpass filter with 12 dB/oct attenuation below the cutoff frequency.

- **Lowpass 1p**

  A basic one-pole lowpass filter with 6 dB attenuation per octave above the cutoff frequency.

- **Lowpass 2p**

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

- **Lowcut+KT**

  A low-cut filter with keytracking. This means that the filter will adjust its cutoff frequency according to the current key pressed.

  This filter type is useful in shaping 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".

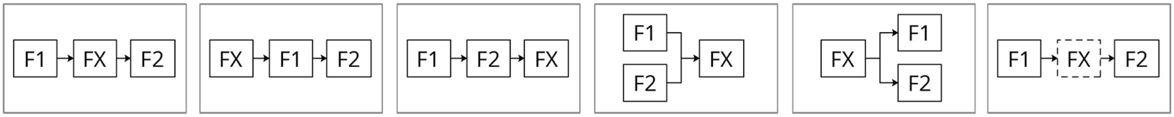
- **Vowel**

  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 with one of the sources in the modulation matrix.

4.7.6. Filter Route

The way the two filters and the insert effect are processed is determined by the ROUTE pop-up. The following setups are possible in DUNE 3:



Route #1 Route #2 Route #3 Route #4 Route #5 Route #6

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.

4.8. Filter Envelope

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

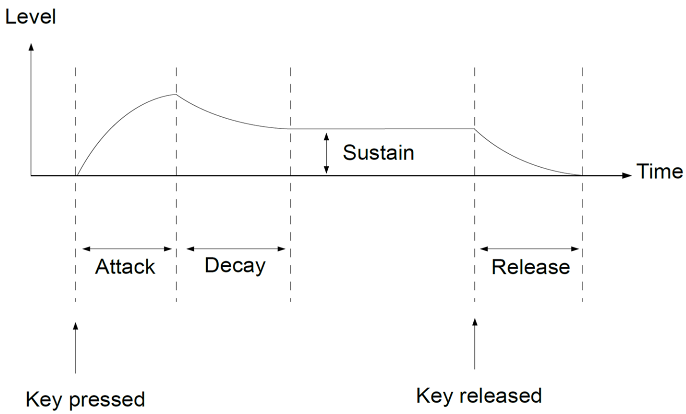


Figure 4.8. The filter envelope.

ENV AMT

The ENV AMT (Envelope Amount) knob controls the extent of the filter envelope's effect on 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 use a low-pass filter with an envelope amount setting in between the two extremes, with 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. These settings mimic the sonic properties of many acoustic instruments.

In rare cases, you may also want to set the envelope amount to a negative value. This is useful for creating 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.

Note that when a dual filter is used, the filter envelope affects both filters simultaneously.

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, which will produce a sound with a bright, snappy start.

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 knob specifies the duration of the decay stage (how long it takes to fall back to the sustain level).

SUSTAIN

Sets the level that is reached after the decay stage ends. The sustain stage lasts as long as a key is held.

RELEASE

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

ANALOG

By default, Dune employs standard envelopes common to most software or hardware synthesizers. The analog switch can be used to create more punchy envelopes, typical of some vintage analog synthesizers. The effect is particularly noticeable with short attack times.

4.9. Amplitude Envelope

The AMP ENVELOPE is located below the FILTER ENVELOPE. The AMP ENVELOPE controls the progression of the volume of a sound (see fig. 4.9). It works in the same way as the FILTER ENVELOPE.

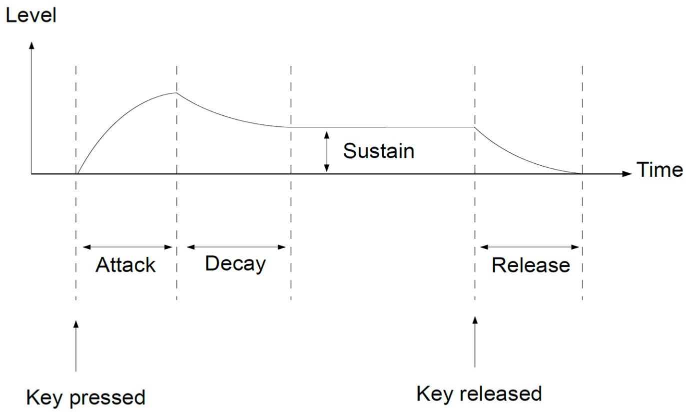


Figure 4.9. The amplitude envelope.

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 (how long it takes the amplitude to fall back to the sustain level).

SUSTAIN

Specifies the level that is reached after the decay stage ends. The sustain stage lasts as long as a key is held.

RELEASE

The final release stage is triggered whenever a key is released. RELEASE specifies the duration it takes the envelope to fall to zero.

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 in the upper righthand corner of the user interface between the central monitor and the MASTER section.

DUNE 3 can utilize 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 with each other.

To use several voices effectively, the individual voices must differ from each other. For a wider or fatter sound, simply increase the number of unison voices, then use the DETUNE or SPRD parameters to widen the sound.

A more in depth way of working with UNISON VOICES is to directly edit specific voices. This can be done by selecting a voice in the COMMON/VOICES EDIT section and making changes to its sound parameters. This way of working with Unison voices lets you stack completely different sounds.

AMT

Controls the number of voices used for each note played. As an example, when set to two voices, each key press will trigger two unison voices.

Note that CPU usage doubles as the number of unison voices is doubled, since two voices per key means twice the processing.

The MAIN monitor displays the current number of active voices, as well as the maximum possible number of voices.

DETUNE

Detunes the unison voices, with higher settings correspond to more harmonic variation. A minimum of two voices is necessary to have an audible change.

The DETUNE amount is centered around the main note pitch. For example, when playing A4 (440 Hz) and detuning two voices by 1 Hz to 339 Hz and 441 Hz respectively, the resulting note of all three voices played together will be perceived as 440 Hz.

SPRD

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 (100%), the voices will be fully spread throughout the stereo field. When using two voices, one voice will be panned hard left, and the other hard right. When using three voices, one voice will be panned hard left, one dead center, and one hard right, etc.

VOICES SOLO

Allows the user to solo individual voices. Note that you can only solo voices that are currently in use. Pressing solo on the 4th voice of a 3-voice patch will thus result in silence.

4.11. Common/Voices Edit

The COMMON/VOICES EDIT section is located just below the UNISON VOICES section. It's where the patch's individual voices can be fine-tuned.

EDIT

From the COMMON/VOICES EDIT section it's possible to edit all unison voices combined (ALL), or to select and edit individual voices. Since each unison voice has its own set of parameters, it's possible to stack up to 8 different sounds.

When a specific voice is selected, its parameters can be copied by pressing the COPY button at the upper righthand side of the COMMON/VOICES EDIT section. To apply the copied parameters to another voice, simply select the destination voice, then press PASTE. The COPY/PASTE function also works with ARP 1 and ARP 2, both of which can be applied from one voice to another and turned on or off for each voice.

MIX

The MIX button of the COMMON/VOICES EDIT section is provided to give full control over the volume of each individual voice within the patch. The 8 knobs provided control the LEVEL of the sound, with the numbers below each knob correlating to the VOICE number.

Turning one of the knobs to the left will lower the volume of that respective voice number in the overall mix of the patch. By contrast, turning the knob to the right will raise the volume of that voice.

GLIDE

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

ARP 1

Enables or disables the first arpeggiator (ARP 1). Note that the arpeggiator can be enabled or disabled for each voice for increased versatility.

ARP 2

Enables or disables the second arpeggiator (ARP 2). Note that the arpeggiator can be enabled or disabled for each voice for increased versatility.

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.

VOICES SOLO

Each of the individual voices can be soloed. Note that only voices that are in use can be soloed. Pressing solo on the 4th voice of a 3-voice patch will thus result in silence.

4.12. Master

The Master section contains basic performance parameters and provides control over the global volume.

POLYPHONY

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

A polyphony of 4 means that a maximum of 4 keys can be held down simultaneously. When pressing a fifth key, one of the existing 4 voices will be cut in favor of the new note.

Note that when sounds have long release times (pads, strings, and choirs, etc.), it's 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. Whereas Mono re-triggers the envelope each time a new note is played, in Legato mode, notes transition smoothly from one to another without re-triggering the envelopes.

Legato mode can be useful for bass and lead sounds, particularly in combination with the GLIDE knob. The result is a unique playing feel and sound which can be better for monophonic lines. Using Mono or Legato mode also demands the least CPU usage.

LIMITER

DUNE 3 features an optional brick-wall limiter at its output, which is 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. It should therefore be used with care and not driven too hard. Otherwise, it may degrade the quality of the output signal.

VOLUME

Sets the overall volume of the synthesizer.

4.13. MSEGs

DUNE 3 features four graphical envelopes. Individually, these are referred to as an MSEG (Multiple Segment Envelope Generator).

Graphical envelopes allow for the precise and customizable adjustment of sound parameters over time, and thus serve as versatile modulation sources. MSEGs can be looped, which allows for the creation of rhythmic gate effects (for instance, the classic "trance" gate sound).

They can also 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 a segment's curve shape by clicking on it with the left mouse button, then drag the mouse up or down.

Six buttons are located on the left side of the MSEG 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). Each voice thus keeps its own envelope position.
- **Note Off** triggers the MSEG when releasing a key. Like Note On, MSEGs are polyphonic in this mode as well.
- **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 the MIDI keys pressed. Key On mode thus allows for the modulation of a whole arpeggiated sequence. For example, Key On can be used to fade in an arpeggiated sequence.
- **Loop** mode periodically loops through the envelope. MSEGs are monophonic in this mode (all MSEG destinations receive the same signal). This is important when using the envelope for trance gate and other rhythmic effects that are synchronized to the song tempo.

SYNC

When the SYNC switch is off, the envelope operates in seconds. The timeline 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 timeline is switched to a musical notation in the format "bars.quarters". For instance, 4.2 correlates to the second quarter note in the fourth bar.

RATE

Adjusts how quickly the envelope is traversed.

When the SYNC switch is off, the RATE knob simply scales the entire envelope from 1/10th to 10 times its duration. Set to the central position, the envelope time is unaffected by the RATE knob. An envelope spanning 1 second would therefore take exactly 1 second to complete.

When SYNC is on, the envelope duration is scaled with precise rhythmic values, which are determined by the host tempo. Setting the RATE knob to half a bar (1/2) will traverse the envelope twice as fast as the default setting 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

Specifies 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. If SYNC is enabled, the maximum duration is given in measures.

VELOCITY (VEL)

With the Velocity (VEL) knob, the envelope amount can be varied according to the velocity of the MIDI key pressed. At the default center position, velocity has no effect.

Set to negative values, low velocities will have a stronger effect than high velocities. Set to positive values, higher velocities increase 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).

KTRK

Makes the envelope rate dependent on the MIDI key number pressed.

Keytracking is normally set to zero, which means the MIDI key number has no effect on the envelope. The higher the keytrack value, the shorter the envelope will be when pressing higher keys. This is useful for imitating the behavior of acoustic instruments such as the guitar, which decays more quickly at higher notes than lower ones).

Keytracking requires the SYNC switch to be turned off, otherwise it will have no effect.

4.14. LFOs

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

Using oscillators, the filter, and envelopes, it's 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 pads and lead sounds, the sustain stage can still sound dull.

This is because the pitch, filter cutoff, and volume are steady and unchanging during the sustain stage.

This is where LFOs (Low Frequency Oscillators) come into play. LFOs function like standard oscillators, and usually generate a periodic signal using similar waveforms (see fig. 4.14). LFOs are inaudible, however. Their only purpose is to continually modulate one or more aspects of the sound.

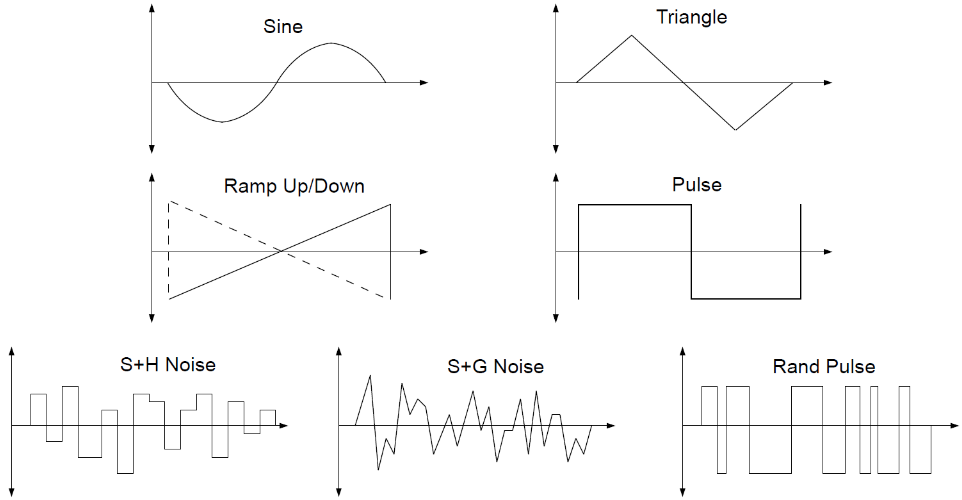


Figure 4.14. LFO waveforms.

The most typical application for an LFO is to modulate volume, filter cutoff frequency, or pitch to create a vibrato or tremolo effect. LFOs can also modulate each other's volume or frequency to obtain even more interesting variations in sound. DUNE 3's three LFOs are much more capable than that, as almost any sound parameter can be used as a modulation destination in the Modulation Matrix.

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

SHAPE

By clicking on the shape popup menu, you may 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 LFO's modulation rate is specified in rhythmic units determined by the project tempo. These units include quarters, eighths, or sixteenth notes, with either standard rhythmic durations, or in triplet (T) and dotted (*) form. Examples:

- 1/4 sets the modulation rate to a quarter note.
- 1/8* sets the modulation rate to a dotted eight note.
- 1/16T sets the modulation rate to a sixteenth note triplet.
- 1/1 sets the modulation rate to span one bar.

POLY

By default, LFOs operate in a global fashion. In other words, all voice destinations receive the same LFO signal. By contrast, when the POLY switch is enabled, each voice uses its own local LFO. This lets the user program modulations that always start at zero phase, for instance.

FADE-IN

Modulations usually start immediately when a key is pressed, and last for the entire duration of the sound.

Sometimes, however, strong modulations can be off-putting in the early attack stage of the sound. To accurately preserve a sound's transients while applying modulation, the fade-in parameter can be used to gradually increase the modulation from zero to its maximum value. The duration of the FADE-IN can be specified in seconds, ranging from 0.01 to 10.00.

OFFSET

Adds or subtracts an offset from the LFO Output, while simultaneously adjusting the LFO amplitude. This can produce interesting effects when modulated via the Modulation Matrix.

AMOUNT

In combination with the Modulation Matrix, this is useful for fine-tuning the overall amount of modulation generated by the LFO. Furthermore, the AMOUNT knob can itself be modulated via the modulation matrix. This is useful for varying the modulation amount over time with a second LFO or with an envelope.

4.15. Effect Bus 1+2

DUNE 3 offers two effects busses. Each contains 9 effects to further enhance the shape and timbre of patches. All effects units may be used simultaneously. Every unit, except for EQUALIZER 1 and 2, offer several sub-types for added versatility.

Note that all effects are global. This means that all voices enabled for a specific effect bus are first summed and subsequently processed by the effect bus.

The effects are processed from left to right in the order they appear. By default, DISTORTION is applied first and REVERB last. You can however modify the order of effects by dragging the individual sections to the left or right.

For example, when you click on an effect label (e.g., "DELAY") a red border will appear. While holding the left-click on the effect label, you can move it into any spot you like.

Changing the effect order is not always audible. The most significant audible change usually occurs when moving an active DISTORTION or COMPRESSOR effect to a different position.

4.15.1. Distortion

Distortion changes the audio signal in a nonlinear manner, which creates new overtones. This results in a harsher sound.

DUNE 3 offers a variety of different distortion types, which are listed below:

- **Crunch** simulates a typical guitar pedal distortion effect.
- **Dirt** is like Crunch, but with a brighter timbre.
- **Fuzz** simulates a fuzz pedal with silicon transistors.
- **Grunge** simulates the vintage DS-1 pedal, popularized by its use in Grunge music.
- **Hard Clip** boosts, then hard-clips the signal at 0 dB.
- **Overdrive** simulates a typical guitar overdrive effect.
- **Saturation** is based on a smooth saturation curve. Ideal for subtle effects.
- **Screamer** is a classic stomp box emulation. There are two variations of this algorithm, one with 4x oversampling and one with 32x oversampling. The latter requires more processing power but can provide higher quality, particularly when distorting high notes.
- **Triode Amp** 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, but with a different timbre that depends on the input signal. Sometimes the results are more subtle than Bitcrush, while at other times they can be harsher.
- **Ratecrush** reduces the sample rate of the signal by using a sample-and-hold circuit.
- **Exciter** distorts only the high frequencies of a signal, leaving other frequencies largely unaffected.
- **Lo-Fi** simulates band-limited signal transmission, filtering out the high frequencies completely.

Note that Hard Clip, Bitcrush and Ratecrush can sound both digital and harsh at high drive settings, so they should be used with care.

DRIVE

Controls the amount of distortion applied to the sound. Higher settings correlate to a more saturated and aggressive tone. When adjusting the DRIVE amount, and in general when working with distortion, it's a good idea to closely monitor your volume. Additionally, avoid abruptly dialing in amounts. Do so gradually instead.

TONE

Tone usually controls the mid frequencies of the output signal, with the central position resulting in a largely neutral sound. In DUNE 3, the implementation and effect of the TONE knob varies depending on which distortion type is chosen.

DRY/WET

Blends between the dry and processed (wet) signal. For guitar-type distortion effects, this parameter should typically be set to 100%. For all other types, it's often a good idea to start with lower values and adjusting to taste.

4.15.2. Equalizer-1 and Equalizer-2

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

The EQ in DUNE 3 features five basic types:

- **B1/B2 (Peaking Bell)** amplifies or attenuates the region around the chosen frequency.
- **LC (Low Cut)** attenuates all frequencies below the chosen frequency using a constant slope given in dB per octave.
- **LS (Low Shelf)** amplifies or attenuates frequencies below the chosen frequency.
- **HS (High Shelf)** amplifies or attenuates frequencies above the chosen frequency.
- **HC (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. This provides instant visual feedback of how the audio signal is being processed.The gain and attenuation of all frequencies are displayed along the vertical axis of the display, with a gain range of ±20 dB.

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-clicking and dragging a circle.

LC (Low Cut) and HC (High Cut) Filters

The Low-Cut filter (LC) and the High-Cut filter (HC), located at the leftmost and rightmost sides of the EQ, are provided in addition to the usual bell and shelf bands.

Both the LC and the HC are turned off by default to keep the EQ response flat and to reduce CPU usage. The filters are enabled by selecting the desired slope from the popup menu within each filter type's assigned section.

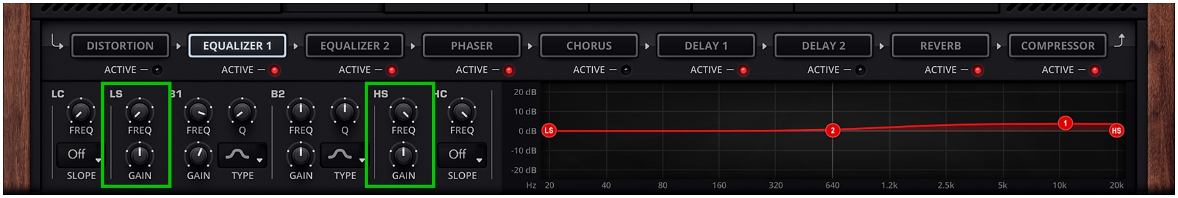
The filter parameters comprise the corner frequency given in Hz, as well as the slope given in dB/octave. The slope can be adjusted from 12 dB to 48 dB, thus allowing a substantial range from a gentle response to a very steep one.

Both the low cut and high cut filters are excellent choices for cleaning up an instrument track, a recording, or even an entire mix. For example, the low cut filter can be used to remove everything below 30 Hz with 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 cut and high cut filter frequencies can be modulated via the Modulation Matrix (see chapter *8. Modulation Matrix*).

LS (Low Shelf) and HS (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.



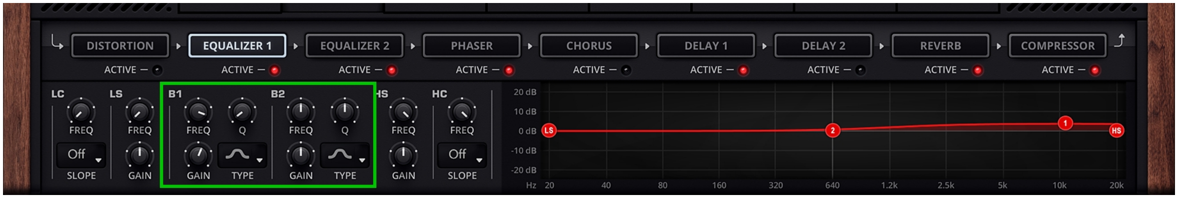
The gain parameter determines the number of decibels (dB) the bass frequencies should be boosted or attenuated. The frequency parameter determines the corner frequency, where the filter starts to work on the signal. The typical range for the low shelf filter is 50-100 Hz.

When attenuating bass frequencies, the low shelf filter only attenuates by the specified dB value. For this reason, it's preferable to use the low shelf filter whenever a subtle change in bass frequencies is desired. For instance, you can use the low shelf filter to lower all bass frequencies below 100 Hz by -3 dB.

The high shelf (HS) filter is a mirrored version of the low shelf filter, but is designed to boost or attenuate high frequencies, as opposed to low ones. The high shelf 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 via the interactive display mentioned above, or via the corresponding knobs on the panel.

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

Bell-shaped EQs are the most common types, due to their versatility. 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 problematic frequencies using a high Q factor and strong attenuation when sweeping through frequencies.

The TYPE button lets the user switch from the default bell-shaped EQ to a shelf type. Due to the extra Q control, this provides a more flexible version of the low/high shelf filters.

4.15.3. Phaser

A phaser modifies the phase of the signal around a 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 an 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. Two stages thus produce one notch. More stages produce more notches, which increases the phasing effect.

"Super Phaser" 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, eighths, or sixteenth notes. These can be as standard rhythmic durations, or in triplet (T) and dotted (*) form. Examples:

- 1/4 sets the modulation rate to a quarter note.
- 1/8+ sets the modulation rate to a dotted eighth note.
- 1/16T sets the modulation rate to a sixteenth note 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 the central position, no feedback will occur.

FREQ

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

DEPTH

Sets the modulation depth of the LFO. Set to 0%, there is no modulation and thus no sweeping effect, resulting in a static sound. When increasing the modulation depth, the phaser operates on a larger frequency region, which expands the sweeping effect.

LR OFFSET

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

DRY/WET

Mixes the dry and processed (wet) signal. Since the phasing effect is achieved by mixing the original and processed signals, the mix ratio should typically be around 50%. Lower values can be useful to make the effect more subtle, 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 produced separately for the left and right channels, 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 that create a warm and spatial stereo image.

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

OFFSET

Adjusts the minimum delay time. Medium to high settings are useful for creating a typical chorus effect, 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 non-zero 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 the central 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

Cuts the bass frequencies of the chorus effect. This only affects the processed signal.

The Low Cut filter is especially useful with low-pitched sounds, as it helps avoid frequency mashes caused by the chorus effect.

HIGH CUT

Cuts treble frequencies of the chorus effect. Again, this only affects the processed signal.

DRY/WET

Mixes the dry and processed (wet) signal. Since the chorus effect is achieved by mixing the original signal with a delayed copy of itself, the mix ratio should typically be around 50%.

Lower values can be useful to make the effect more subtle, 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 twelve delay types are available.

MODE

- **Simple** creates a series of echoes that are centered in the stereo field.

- **Simple+Offset** operates similarly to Simple delay, except that the right channel is offset. When both channels share the same delay time, this sounds like an offset version of the ping-pong delay effect.
- **Ping-Pong** creates echoes that alternate 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 that alternate between the left and right channels.
- **Diffuse** works like the Simple delay, except that each echo becomes increasingly diffused.
- **Tape** works like the Simple delay, except that each echo becomes increasingly saturated.
- **L-C-R** (Left Center Right) is a special delay type where the echoes alternate between the left channel, the center, the right channel and the center again, and so on.
- **Filtered** creates echoes with changing timbres. This delay program features a CUTOFF and Q knob to control the filtering of the delayed sound.
- **Rhythmic** works like the Simple delay but uses a specific pattern.
- **Swing** lets the user create swing-type delay effects by adjusting the timing of every other echo. The swing amount is adjusted via the SWING knob.
- **Vintage** is a special delay program that creates warm and spacious echoes.
- **Reverse** creates a reversed delay signal, then produces echoes of the reversed signal. Note that this effect should be used on a reasonably complex input signal. It's most obvious on a vocal patch: "Hello" becomes "Olleh".

RATE L/R

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

By default, delay times are synced to the host tempo and are specified in quarters, eighths, sixteenths, etc. Triplet (T) or dotted (*) values can be specified as well.

Examples:

- 1/4 specifies an echo duration of a quarter note.
- 1/8+ specifies an echo duration of a dotted eighth note.
- 1/16T specifies an echo duration of a sixteenth note triplet.
- 1/1 specifies an echo duration that spans an entire bar.

Turning SYNC off lets the user specify the delay time in milliseconds (ms) separately for each channel.

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 sound interesting. When SYNC is off, it's also possible to modulate the delay time via the modulation matrix.

This is useful for creating 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 sounds too strong, the width knob can be used to reduce the stereo image down to a mono signal.

FEEDBACK

The feedback knob adjusts the length of time the echoes are repeated for. The percentage specifies the level change from one echo to the next. This means that 100% creates an infinite series of echoes, 50% cuts the level of each subsequent echo in half, and so on.

MOD-RATE

The delay modulation rate in Hertz (Hz).

MOD-AMT

The MOD-AMT knob adjusts the modulation depth of an LFO that modulates the delay time. Set to zero, no modulation takes place, and the echoes will sound 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 pitch modulation, which can be used for special effects.

DRY/WET

Blends between the dry and processed (wet) signal.

4.15.6. Reverb

Reverb is used to create the illusion of a sound being played back in a spatial environment such as a living room, hall, or cathedral. Eighteen different algorithms are offered in DUNE 3.

Note that some of these are treated below individually, while others are treated in pairs.

TYPE

- **Ambience** is useful for adding subtle ambience to a sound.
- **Cathedral** is an algorithm designed for large spaces with a reverb time of 4-5 seconds or more.
- **Chapel** simulates a large chapel and works best for reverb times of 2-3 sec.
- **Big Room** simulates a room, designed for a reverb time around 1 sec.
- **Gated Room** and **Studio Room** are designed for (very) short reverb times.
- **Plate** is a highly diffuse reverb program, as it does not simulate a specific space.
- **RutaVerb** simulates a concert hall. The algorithm produces a dense reverb with exceptional realism and a nice reverb tail.
- **Small Hall / Medium Hall / Large Hall** simulates a hall space of various sizes (small, medium, or large). The algorithm works best with reverb times of 3 secs and more.
- **Slapback Hall** simulates a hall with strong early reflections occurring after a few milliseconds.
- **Shimmer Room** and **Shimmer Hall** 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.
- **Vintage 24** algorithms are based on a timeless vintage reverb from the 1970s. All three types contain an additional DEPTH control which can be used to adjusts the buildup of the reverb.
- **Vintage 24 Room** is designed for small spaces and short reverb times of up to 2 sec.
- **Vintage 24 Small Hall** and **Large Hall** are designed for larger spaces. Whereas the Small Hall tends to sound smooth and bright, the Large Hall tends to sound dark and grainy.

PRE-DELAY

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 listener's perception of the room size.

TIME

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

Changes the simulated room's wall materials. Low settings simulate reflective walls with a bright, uncolored sound. High settings simulate absorbent walls with progressively darker timbres.

LOW CUT

Useful for removing unwanted low frequencies from the processed signal. This is useful for sounds containing strong bass frequencies, such as bass drums, etc. Note that only the reverberated signal, and not the dry signal, is affected by this parameter.

HIGH CUT

This is an additional parameter in the Vintage 24 algorithms. This parameter removes high frequencies from the reverberated signal, and can be used to create darker spaces.

COLOR

Can be used to equalize the mid frequencies of the reverberated signal. At the central position the reverberated signal is unaffected. Turning the knob to the left cuts the mid frequencies, while turning it to the right boosts the mid frequencies.

MOD-AMT

All reverbs in DUNE 3 employ modulation via an LFO to generate a rich timbre. MOD-AMT (Modulation Amount) adjusts how strong this modulation is.

WIDTH

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

DRY/WET

Blends between the dry and processed (wet) signals.

4.15.7. Compressor

Compressors reduce the dynamic range of a signal. This can increase the perceived loudness of a signal, but it can also reduce clipping 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 the form of a threshold level given in decibel (dB) amounts.

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 that use optical attenuators to control the dynamics of the signal. Unlike the Air compressor, Opto yields a more transparent signal if the amount of compression applied is low.
- **Punch** is based on a classic analog solid-state design. It's useful for giving more punch to kick or drum sounds.
- **Vintage** is a classic single-band compressor that 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 at around 0 dB.

RATIO

Specifies how much the compressor lowers (compresses) passages that are louder than the threshold. For example, a ratio of 1:1 means that the sound comes out at 1 to 1 (as it is). The compressor will in this case do nothing.

At a ratio of 2:1, loud passages will be lowered by a factor of 2 (the output produced will be only half as loud as the input). At a ratio of 4:1, they will come out 4 times softer, and so on.

Refer to figure 4.15.7 for a graphical representation of the compression input/output ratio. The image shows ratios 1:1, 2:1 and 100:1.

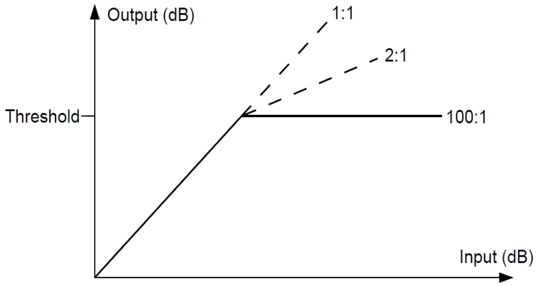


Figure 4.15.7. The compression input/output ratio.

At the highest setting (100:1), the compressor will act as a limiter, and will not allow any signal levels above the threshold to pass.

THRES

The threshold knob sets the level at which a passage is considered "loud" (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 through, while anything above -16 dB will be attenuated according to the ratio setting.

ATTACK

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 they can also cause distortion artifacts on signals when the compressor reacts too quickly. Long attack times usually don't cause distortion, but fast transients pass through uncompressed, which can sometimes sound objectionable.

RELEASE

Adjusts the time it takes for the compressor to recover when the threshold is no longer exceeded.

OUTPUT

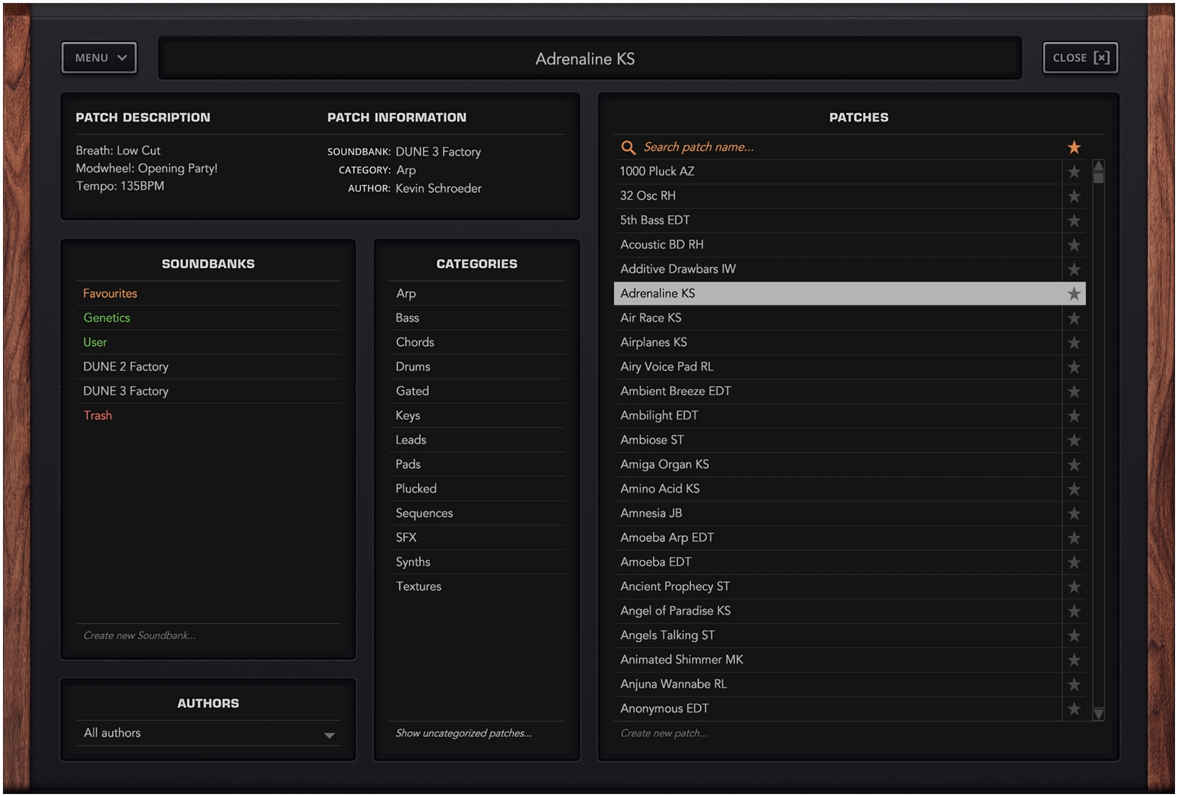
Adjusts the compressor output level in decibels (dBs).

DRY/WET

Blends between the dry and processed (wet) 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 has maintained an extensive Patch Browser which lets the user search for specific sounds, manage soundbanks, mark favorites, and 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:** displays all patches that have been marked as favorites.
- **Genetics:** displays all saved patches made with the GENETICS function.
- **User:** displays the user's own patches, which 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 displayed 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:

- **Open Genetics:** opens the Genetics tab, which can be used to create new sounds (see below).
- **Import Patch / Import Folder:** can be used to add a single DUNE 3 patch or a directory of patches to the soundbanks. This can be useful to quickly integrate patches from external hard drives or USB sticks.
- **Import ZIP Contents:** lets the user directly import a zipped soundbank. If there are additionally required wavetables in the ZIP file besides the patches, these are also automatically copied into the wavetables directory. In this way, third-party soundbanks can be installed quickly and conveniently.
- **Reload Soundbanks:** reloads all patches. This is provided 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 existing patches. In doing so, properties from these patches are pieced together randomly.

GENERATE

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

IMPORTANT*: When pre-listening, the listening volume should be relatively low since the generated patches are random. This means they can become very loud when generating new variations.

AUTO

Functions similarly to the GENERATE function. It differs however by randomly selecting and combining two new patches from the current directory with each click.

SAVE

Stores patches created with the GENERATE and AUTO functions in their own soundbank ("Genetics"). These patches are date and time stamped, which is how they are named in the folder. They can be renamed at any time by right-clicking on the patch in the Genetics folder, choosing Rename Patch, and typing the desired name into the box. Alternatively, the selected patch can be renamed by pressing F2.

AMOUNT

The Genetics function can weight the selected patches differently. This is controlled by the "Amount" knob.

The higher 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, and the first patch remains as is.

KEYBOARD SHORTCUTS

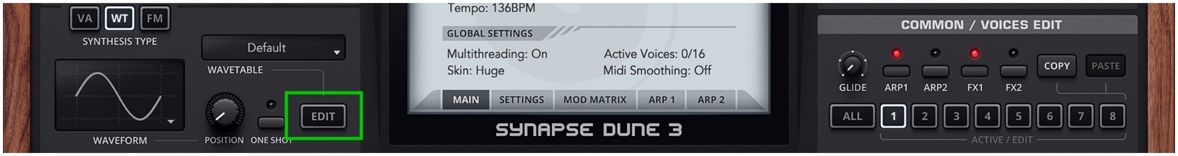
To speed up the workflow, the browser supports the following keyboard shortcuts:

- **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 ability to create your own waveforms and wavetables. To use the Wavetable Editor, one of the oscillator blocks must be set to Wavetable (WT) mode.

With Wavetable (WT) mode chosen, open the Wavetable Editor with the EDIT button:



The Wavetable Editor is divided into three sections.

- In the upper toolbar there are tools for editing and a small window for navigating within the waveform.
- In the middle is the actual editor. Waveforms can be drawn here with the mouse.
- In the lower area the user can select the desired waveform for editing. A wavetable can contain up to 256 waveforms. The number of waveforms used can be adjusted with the +/- buttons. Waveforms or even complete wavetables can be created by entering a mathematical formula.

6.1. Toolbar

MENU

The MENU popup contains multiple useful functions for creating wavetables.

Import WAV (Single-Cycle)

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. 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 reused in other patches or projects. Otherwise, saving the current patch will also save the wavetable settings.

Export Wavetable (WAV)

Saves a wavetable in WAVE format.

Morph Wavetable

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

Removes the DC offset of a waveform.

This ensures that the energy above and below the zero line is balanced. Removing DC offsets avoids clicks during the transient stage.

Normalize Waveform

Normalizes the amplitude of the current waveform to 0dB.

Clean Waveform

Clears 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. When selecting Remove Waveform, the wavetable will become 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. The left mouse button is used to set curve points, which can then be moved by dragging the mouse.
- The **Sine** tool opens the additive editor, which lets the user 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, for example, organ or vocal-like waveforms.

Undo/Redo

The Undo and Redo buttons (left/right arrow) are used to return to previous steps in the editing process.

Navigation Pane

The Navigation Pane at the top of the WT Editor shows the current waveform and the currently selected section of it. By zooming into the waveform, the desired section can be moved using the Navigation Pane.

Zoom

With the zoom buttons -/+ on the upper righthand side the user can zoom the waveform in and out.

Alternatively, you can use the mouse wheel to zoom in and out if the mouse is either in the Navigation Pane or the Edit Pane.

Close

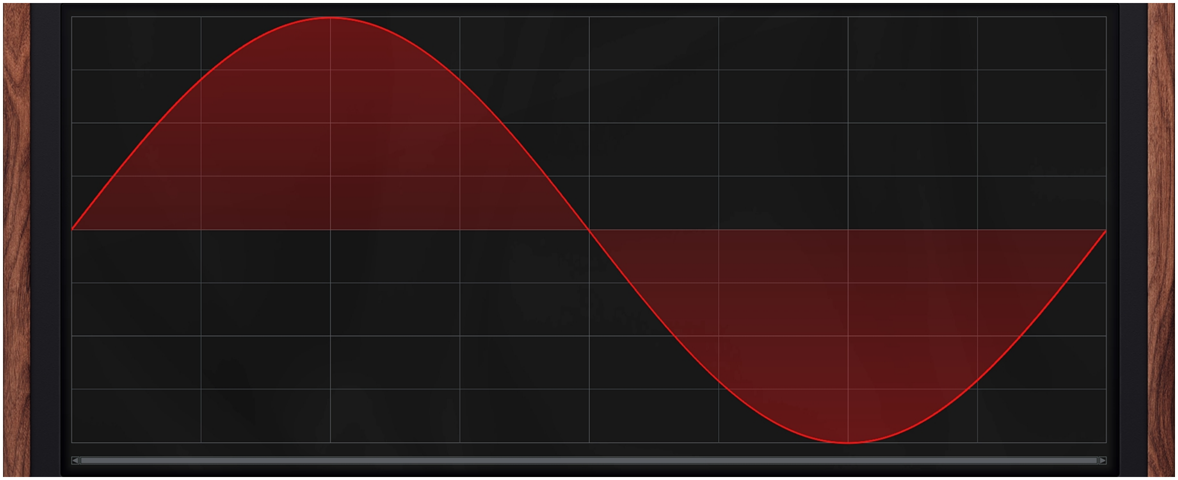
Closes the wavetable editor.

6.2. Editor

The Edit Pane is in the center of the Wavetable editor.

In the Edit Pane, the user can use a 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

Using the selection tool, a section of the wavetable can be created by left-clicking/dragging. Alternatively, the entire waveform can be selected by double-clicking.

With a right click, operations like copy/paste, fade in/out, etc., can be applied to the section.

For real-time editing, Ctrl+Left-click and Alt+Left-click are useful tools for changing the phase or volume of the selection.

Pen tool

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

Line tool

The Line tool allows the user to draw a straight line by left-clicking/dragging.

Shift+left-click is a useful trick that allows the user to draw the line in such a way that it always runs along the curve. By drawing lines this way, there won't be 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.

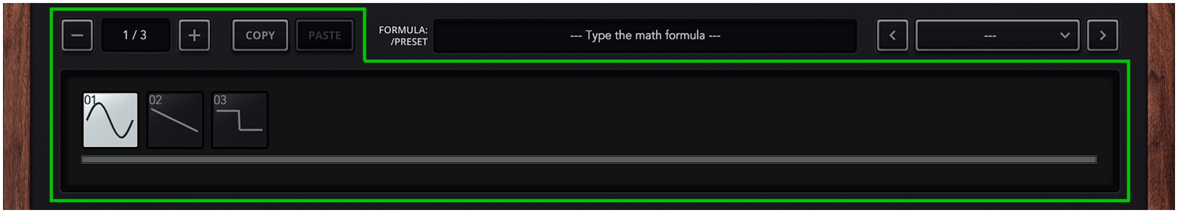
New points can be created or existing points can be moved with the left-click of a mouse. By default, a straight line is drawn between two points, however this can be changed into a curve by left-clicking/dragging.

Additive Editor

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

Use left-click and 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, which are largely responsible for shaping the sound of a waveform. The additive editor is useful for creating create organ, bell, or vocal-type waveforms.

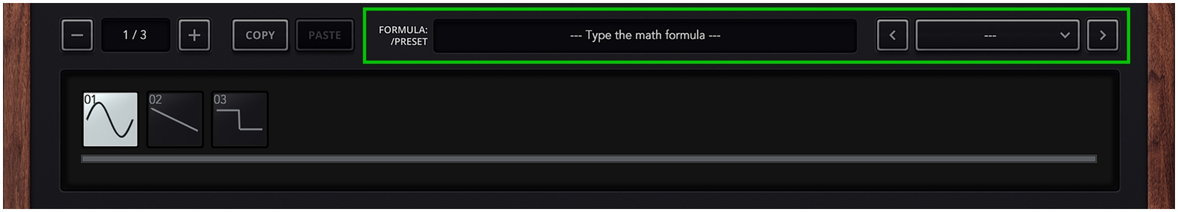
6.4. Waveform selection



All waveforms of the wavetable are displayed in the thumbnail view in the lower part of the wavetable editor. Left-click on one of the waveforms to select it for editing. Use drag and drop to swap individual waveforms with each other.

Drag and drop is particularly important if you are using modulation to pass through the wavetable, as the order of the waveforms has a significant influence on the sound.

6.5. Formula Editor



By entering a formula such as sin(x*PI), a complete waveform can be generated.

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.

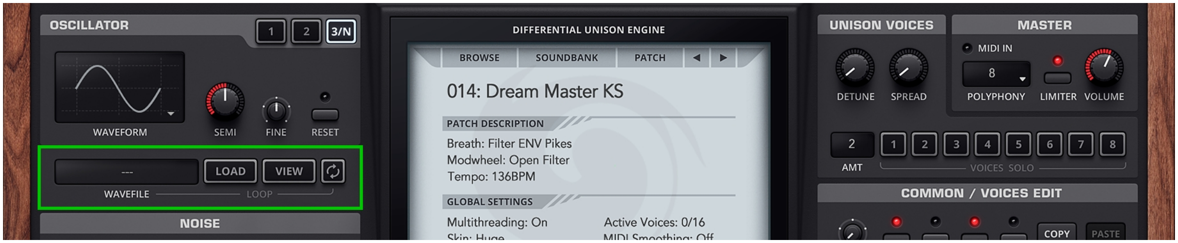
Example: The formula sin(x*PI)*y would slowly fade in a sine wave.

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

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

7. Playing WAV files

Since version 3.4, WAV files can be used in patches. This is done via oscillator 3:



After pressing the LOAD button, a directory window opens where you can select your own WAV files. 16, 24, and 32-bit WAV files between 44.1 and 192 kHz sample rate, in both mono and stereo, are supported. WAV files can be easily dropped from the chosen directory onto the WAVEFILE display.

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 three areas.

- In the upper bar there are tools for editing and a small window for navigation within the waveform.
- In the center 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 several useful functions for editing WAV files.

Loop

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

Draw Ghost

When activated, Draw Ghost will overlay the loop end with the loop starting point 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

Automatically searches for favorable loop points 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

Removes the DC offset of the audio file. This 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 selects a region to 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 is used for selecting and modifying a horizontal section of the audio file.
- The **Pen** tool is used for drawing 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

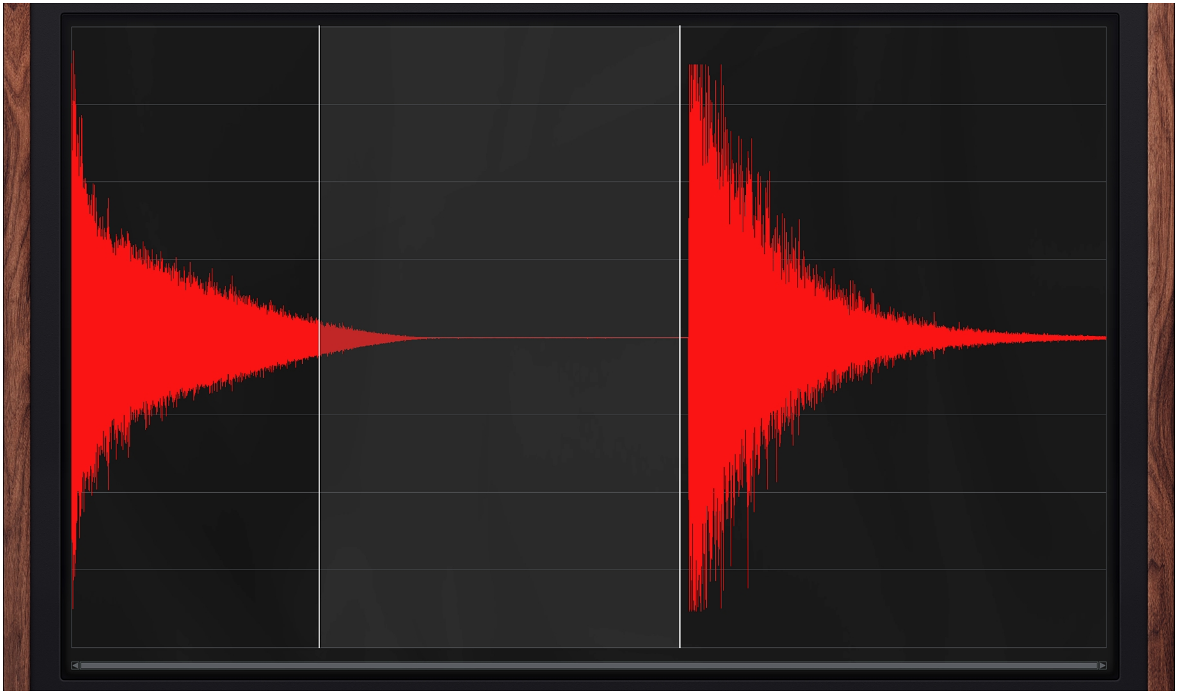
The zoom buttons (-/+) on the righthand side let the user zoom in and out of the audio file. Zooming is also possible using the mouse wheel when the mouse is in either the Navigation Panel or the Edit Area.

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, show/hide, etc. to the selection.

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

Pen tool

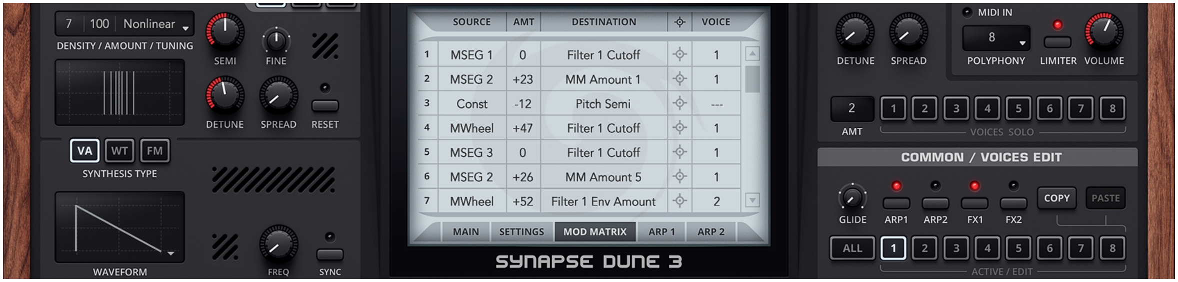
Use the left mouse button to directly modify individual points of the audio file.

8. Modulation Matrix

One of the biggest strengths 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 drawback to subtractive synthesis is that it is achieved by employing fixed structures with a limited set of parameters. This means that it can be limiting in flexibility and creativity in sound design tasks.

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 (destinations):



The Modulation Matrix in DUNE 3 can be found in the center of the interface. It can be 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), which controls the amount of modulation applied to the destination parameter, is moved like a knob. The destinations can also be dragged directly to knobs or switches using the Drag+Drop crosshair icon next to the "DESTINATION" column.

The modulation sources comprise the LFOs. The ability to link an 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), and Filter Cutoff.

Further sources include the graphical envelopes (MSEG 1-4), which must also be assigned to a destination in the modulation matrix to work. Typical destinations for the graphical envelope generators include pitch, volume, or frequency modulation (FM) amount when using FM synthesis.

The modulation matrix is also used for assigning MIDI controllers to sound parameters. For instance, the modulation wheel or expression pedal 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 3 much more flexible and powerful than the Unison function found in traditional synthesizers, which often come with either no (or very few) modulation parameters for unison voices.

This concept is referred to as the Differential Unison Engine, hence the name "DUNE".

To modulate unison voices, the modulation matrix contains 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 the modulation is applied to all unison voices in use. If the number of unison voices is set to five voices, for example, then the filter cutoff will be modulated in the same manner for all five voices.

The second and third column, which are set to modulate volume and panorama, are restricted to affect the second and third unison voices, respectively. As a result, only those voices will be modulated.

To understand the unison engine, try experimenting with 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 in the Unison panel, which is located above the modulation matrix. Use the SOLO switches in the same panel to toggle between individual unison voices and to see how the modulation affects them.

A further example will demonstrate how to set two unison voices to different settings:

Here we modulate two parameters: Filter 1 Cutoff and Filter 1 Reso (resonance). The modulations are set to affect the second unison voice. As a source, we chose "Const" (constant). The result is that the two filter parameters are adjusted relative to the knob settings.

For example, if the Cutoff knob is set to 50%, then voice 2 will have a cutoff setting corresponding to 100% (50% + 50%).

Note that in DUNE 3, there are two ways to change the parameters of unison voices.

For the first, the modulation matrix can be used as described in the second example above. For the second, voice parameters can be edited directly by using the Voice Edit section.

By editing voice parameters directly, absolute changes are made to the unison voices. This is necessary if different filter types, synthesis models, or wavetables are used in the unison voices. Editing unison voices in this way is also generally a good idea if you want to stack completely different sounds.

By contrast, the modulation matrix is used to perform relative changes to the unison voices. This is useful for keeping common controls. For example, suppose a C-minor chord stab is built with three unison voices using the modulation matrix. In this case, all three voices can be edited together. Tweaking the cutoff knob or changing pitch in this way 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 and provides a brief explanation of each. All sources, whether MIDI data or synth parameters, are converted to the same range: [0,+1] for unipolar and [-0.5,+0.5] for bipolar sources. The LFOs, the Pitch bend wheel, the Random and Alternate modifiers 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 will be.

Vel>100

Emits maximum value when the MIDI key velocity is greater than 100, and zero when 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, to shorten the envelopes when pressing higher keys.

Keytrk (C4)

Same as Keytrack, except that the base note is C4, as opposed to C0.

Freqtrack

Identical to keytrack, 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 don't just transmit velocity, they also send pressure information.

This pressure-based information 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 pressed.

PolyAT

Like Aftertouch, however pressure information is transmitted per key (Polyphonic Aftertouch). Note that there are only a few MIDI keyboards which support sending pressure information individually per key. For most MIDI keyboards, using this source will thus have no effect.

Const

This source 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, a parameter can be offset for a specific unison voice, or to set values for parameters only available in the modulation matrix.

Random

Sends 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

Switches the polarity every time a key is pressed. As one example, this source can be used to MAKE sounds alternate between the left and right channel: with "Alternate" as a SOURCE, "+100" as the AMT, and "Amp Pan" as the DESTINATION.

Arp 1/2 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 1/2 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).

Brightness

Uses the current value of MIDI Brightness (CC#74) as a source. This can be used polyphonically for MPE (MIDI Polyphonic Expression).

Using a suitable MIDI controller that supports the MPE standard, it's possible to send individual brightness values for each key pressed.

Osc 1

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

Osc 2

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

Osc 3

The output of oscillator 3 as the modulation source.

When using this source, make sure to adjust the modulation rate to "Fast" or "Audio-Rate".

Noise Gen

The output of the noise generator as the modulation source.

When using this source, make sure to adjust the modulation rate to "Fast" or "Audio-Rate".

LFO 1

Current value of LFO-1 as the modulation source.

LFO 2

Current value of LFO-2 as the modulation source.

LFO 3

Current value of LFO-3 as the modulation source.

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 (AT) data.

Useful for creating vibrato-type effects when applying pressure on the keyboard.

Voice

The unison voice number counting from zero. In other words, 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 for each unison voice to a different value.

Filter Env

Current value of the filter envelope as the modulation source.

Amp Env

Current value of the amplitude envelope as the modulation source.

MSEG 1

The output of envelope generator 1 as the modulation source.

MSEG 2

The output of envelope generator 2 as the modulation source.

MSEG 3

The output of envelope generator 3 as the modulation source.

MSEG 4

The output of envelope generator 4 as the modulation source.

8.3. List of Destinations

The DESTINATION menu of the MOD MATRIX contains virtually all of the sound parameters DUNE 3 has to offer (as described in the third chapter). It also contains most effects parameters and a few helper functions that are not available on the front panel.

8.3.1. Common

Pitch Semi

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

Pitch Fine

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

Osc 1+2 Init Phase

Provides a way to set/modulate the initial phase for oscillators 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 than or equal to 50. The intended use of this destination is to assign (e.g., "Foot" "+50" "Arp Hold").

Mixer Volume

Adjusts the volume of all oscillators, RING MOD (RM) 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 as the modulation destination.

8.3.2. Oscillator 1, 2, 3

This section describes all oscillator destinations.

Note that oscillator 3 has fewer parameters, as it shares only the first six parameters with oscillators 1 and 2. Nevertheless, since oscillator 3 is used to load WAV files, it also contains one additional parameter which is not available for oscillator 1 or 2. This will be covered in more detail below.

Osc Semi

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

In the context of the Differential Unison Engine, choosing Osc Semi as a destination lets the user specify arbitrary tunings for each voice.

For example, to give a fourth oscillator a different pitch than the first three oscillators, simply increase the number of unison voices and set the oscillator pitch of that voice only via the mod matrix.

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 to obtain vibrato effects. For strong vibratos spanning a larger pitch range, use Osc 1/2 Semi as a destination instead. Modulation can be restricted to specific voices, if desired. For example, it's possible to have a static, low-pitched sine wave modulating the first voice and a vibrating sawtooth modulating the second for varying vibrato results.

Osc Volume

The mixer volume of the oscillator.

Osc Pan

The mixer panorama of the oscillator.

Osc Init Phase

Sets the absolute starting phase of the oscillator whenever a note is triggered. Consequently, the oscillator will no longer be in free-run mode.

Set to zero, the specified oscillators start at zero phase. An amount of +50 corresponds to a +180-degree phase shift, while an amount of +100 corresponds to a +360-degree phase shift, etc. Though the human hearing is largely unable to differentiate the starting phase of a single oscillator, the relative phase difference between both oscillators can matter in some cases.

For example, it becomes more obvious when both oscillators share the exact same frequency.

Osc Keytrack

Sets the keytracking of the oscillator. Note that this is only available in the modulation matrix.

An amount of +100 corresponds to standard keytracking behavior, where each MIDI key number corresponds to one semitone. By changing this parameter to values other than +100, strange tunings may be realized, or keytracking can be turned off 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 PW (Pulse Width) (Uni)

Selects pulse width as the modulation destination.

Note that this only works when the oscillator is set to "Pulse" in VA mode.

The Osc VA PW (Uni) target is used for unipolar modulation from 50% to 100%.

OSC VA PW (Pulse Width) (Bi)

Allows for bipolar modulation of pulse width. Bipolar modulation includes widths ranging from as high as 50% all the way down to 0%.

Osc VA Sync Freq

Changes the synchronization frequency when the oscillator is set to VA mode with Sync enabled.

Osc WT Position

Sets the Wavetable position when the oscillator is set to WT mode. This lets the user sweep through the wavetable by applying an LFO or MSEG.

Osc FM Amt A

Adjusts the operator A amount when the oscillator type is set to FM mode.

Osc FM Amt B

Adjusts the operator B amount.

Osc FM Amt C

Adjusts the operator C amount.

Osc FM Feedbk

Adjusts the amount of operator feedback.

Osc FM Ratio A

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

Osc FM Ratio B

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

Osc 3 SamplePos

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 SampleStart destination should be used instead.

Osc 3 SampleStart

Allows to change the sample start position for WAV playback. A value of 0 is equivalent to the start of the WAV file, while 100 specifies the end.

8.3.3. Ring Mod / Noise

Ring Mod Volume

Adjusts the mixer volume of the ring modulator (RM).

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

Adjusts the low-pass filter of the noise generator.

Noise Highpass

Adjusts the high-pass filter of the noise generator.

Noise Width

Adjusts the stereo width of the noise generator.

8.3.4. Filter

This section describes the available filter destinations in the mod matrix.

Filter 1 Cutoff

Adjusts the cutoff frequency of FILTER 1 relative to the front panel knob, using the same value range.

Filter 1 Reso

Adjusts the resonance for FILTER 1 relative to the front panel knob, using the same value range.

Filter 1 Keytrack

Sets the key tracking for FILTER 1. This specifies how much the cutoff frequency changes according to which MIDI note number a voice is playing.

The parameter works the same as the KTRK knob on the front panel, however negative values may also be specified in the Mod Matrix. Negative values will reverse the standard effect of keytracking by causing low keys to have higher cutoff frequencies than high keys.

Filter 1 Drive

Adjusts the drive parameter for the FILTER 1.

Filter 1 Env Amount

Adjusts the envelope amount for FILTER 1 relative to the front panel knob, using the same value range.

Filter 2 Cutoff

Adjusts the cutoff frequency for FILTER 2.

Filter 2 Reso

Adjusts the resonance for FILTER 2.

Filter 2 Keytrack

Sets the key tracking for FILTER 2.

Filter 2 Drive

Adjusts the envelope amount for FILTER 2.

Filter 2 Env Amount

Adjusts the envelope amount for FILTER 2 relative to the front panel knob, using the same value range.

Filter Balance

Adjusts the balance between FILTER 1 and FILTER 2.

Filter FX Amount

Adjusts the filter effect's "Amount" knob.

Filter FX Lowpass

Most filter effects (esp. Distortion), are followed by a low-pass filter that can be used to roll off unwanted high frequencies. This 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 that are available in the Mod Matrix.

Filter Attack, Decay, Sustain, Release

Adjusts the amplitude envelope parameters relative to the settings on the front panel. This lets the user 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, giving the user control over the panorama of the entire voice.

Note that this parameter has no corresponding knob on the front panel and is only available in the Mod Matrix.

8.3.6. MSEG

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

MSEG 1-4 Rate

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

8.3.7. LFO

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

LFO 1-3 Amount

Adjusts the amount of the specified LFO.

LFO 1-3 Offset

Adjusts the offset of the specified LFO.

LFO 1-3 Rate

Adjusts the rate of the specified LFO.

LFO 1-3 Init Phase

Adjusts the initial phase of the specified LFO when a new voice is started.

This parameter works like oscillator Init Phase.

8.3.8. Mod Matrix

MM Amount 1-32

The MM Amount destinations allow the user to adjust the modulation amount of any modulation slot.

This feature can be useful for building complex modulations. As an example, let's say LFO 1 is used as a SOURCE in MM slot 1 to modulate Noise Volume as a DESTINATION. LFO 2 could then be used to modulate/control the modulation amount of LFO 1 by choosing LFO 2 as a source in any other open slot, with MM Amount 1 as the DESTINATION.

Another useful application for MM Amount is to make modulations dependent on a MIDI controller. For example, by choosing Mod Wheel as a SOURCE and an MM slot(s) containing a modulation envelope as a destination, the modulation depth of that slot can be controlled by the modulation wheel.

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

8.3.9. FX Bus 1 and 2

This section describes the effects destinations available in the MOD MATRIX.

Almost all destinations correspond to their front panel knobs (refer to chapter *4. Sound Parameters* for a detailed description of the parameters).

Dist

Adjusts the corresponding DISTORTION parameters.

EQ-1 / EQ-2

Adjusts the EQUALIZER parameters. All parameters can be modulated, apart from any switches and the low-shelf and high-shelf filters.

Phas

Adjusts the corresponding PHASER parameters.

Chrs

Adjusts the corresponding CHORUS parameters.

Del 1 / Del 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

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

9. MIDI Reference

Most knobs and buttons on the front panel can be controlled remotely via MIDI. DUNE 3's default controller assignments follow common conventions and the MIDI standard as much as possible.

The number of sound parameters DUNE 3 offers is however higher than the amount of available MIDI controllers. Most effects parameters and large parts of the MOD MATRIX are thus not assigned to any MIDI controller number. To override the default MIDI assignment, right-click on a knob and select "MIDI Learn", then move your hardware's desired controller knob. To remove an assignment, choose "MIDI Forget". Once you close DUNE 3, the changes will be made permanent and will apply to any future instance of DUNE 3 until you reassign them.

Parameter	CC#
Common	
Modulation Wheel	1
Glide	5
Sustain Pedal	64
Volume	111
Unison	
Detune	53
Pan Spread	54
Oscillators	
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
Arpeggiator	
Arp 1 On/Off	47
Arp 1 Rate	50
Arp 1 Length	52
LFOs	
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
Filter	
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
Amp Envelope	
Amp Attack	73
Amp Decay	36
Amp Sustain	37

MIDI Reference cdn.

Parameter	CC#
Amp Release	72
Effects (Bus 1)	
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
Mod Matrix	
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, many services often 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 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

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 it, and if you can really hear a difference. Audio-rate modulation needs a substantial amount 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 are a few tips that will help reduce CPU usage:

- Oscillator stacks 1+2 can be turned off by setting the oscillator count to zero. This is strongly recommended if they aren't needed.
- 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. DUNE 3 has stereo oscillators with pan + pan SPREAD, so it's 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. We're happy to assist you!

11.1. Notes click when pressing or releasing a key

Check the amplitude attack and amplitude release time. Increase 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 the demo version is 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 *9. MIDI Reference* chapter.

Note that you can right-click on any knob and choose "Clear MIDI Map", to ensure that no MIDI controller messages can change sound parameters unexpectedly.

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.

There are 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, and COMPRESSOR can also distort.

First check to see if DISTORTION is used in one of the effects sections or as a Mod Matrix parameter. Otherwise, switch to other patches and compare the output.

11.5. MIDI messages are not received

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

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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You may install the Software on up to three computers simultaneously, as long as only one installation is used at any given time.

For the products DUNE 3 and The Legend, a one-time online activation of the software is required after installation. The software activation is only used for license control, to protect the software against unlawful copying. No personal data is transmitted in this process.

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Appendix. Sound Design Reference

The last two characters of every patch name are the author's initials. The following table lists all sound designers who contributed patches to DUNE 3, and a website where you can learn more about their work.

Ab.	Author name	Email or website
ARK	Arksun	http://www.arksun-sound.com/
DG	David Goodwin	kriminal@ntlworld.com
EDT	Ed Ten Eyck	http://www.edtaudio.com
KS	Kevin Schroeder	http://www.dejavu-sound.de/
IW	Ingo Weidner	ingo_weidner@web.de
LE	Lance Emmerich	lance.emmerich@gmail.com
MC	Michael Cavallo	michael@monomosound.com
MH	Mark Holt	http://flavors.me/markholtuk
MK	Michael Kastrup	http://www.xsynth.com
PK	Piet Kaempfer	www.protonica.de
RH	Richard Hoffmann	www.synapse-audio.com
RL	Rob Lee	-
SDX	Shadx	-

Sound Design Reference cdn.

Ab.	Author name	Email or website
ST	Solidtrax	http://www.solidtrax.nl
TK	Teksonik	teksonik@outlook.com
XS	Xenos Soundworks	-

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