Zynaddsubfx

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This documentation is a work in progress

Getting Started

ZynAddSubFX is a fairly complex software synthesizer with a very large number of controls. As such, it is not always obvious how to use ZynAddSubFX.

Many applications under Linux transport MIDI over ALSA and transmit audio over JACK. ZynAddSubFX can be run in this configuration by running:

```
zynaddsubfx -I alsa -O jack -a
```

This sets the input driver to be alsa and the output driver to be jack, which should attempt to autoconnect to your soundcard as per the *-a* flag. If this is your first time running ZynAddSubFX, you will see a screen that lets you choose between the advanced and beginner interface. Currently the beginner interface is deprecated, so the advanced one is recommended.

Now you should be able to see ZynAddSubFX's main window, from which you can setup patches, effects, and general configurations, but more importantly it provides links into the parameters of the patches. ZynAddSubFX is a powerful tool with a number of base patches, but its true power lies in the ability to make your own patches.

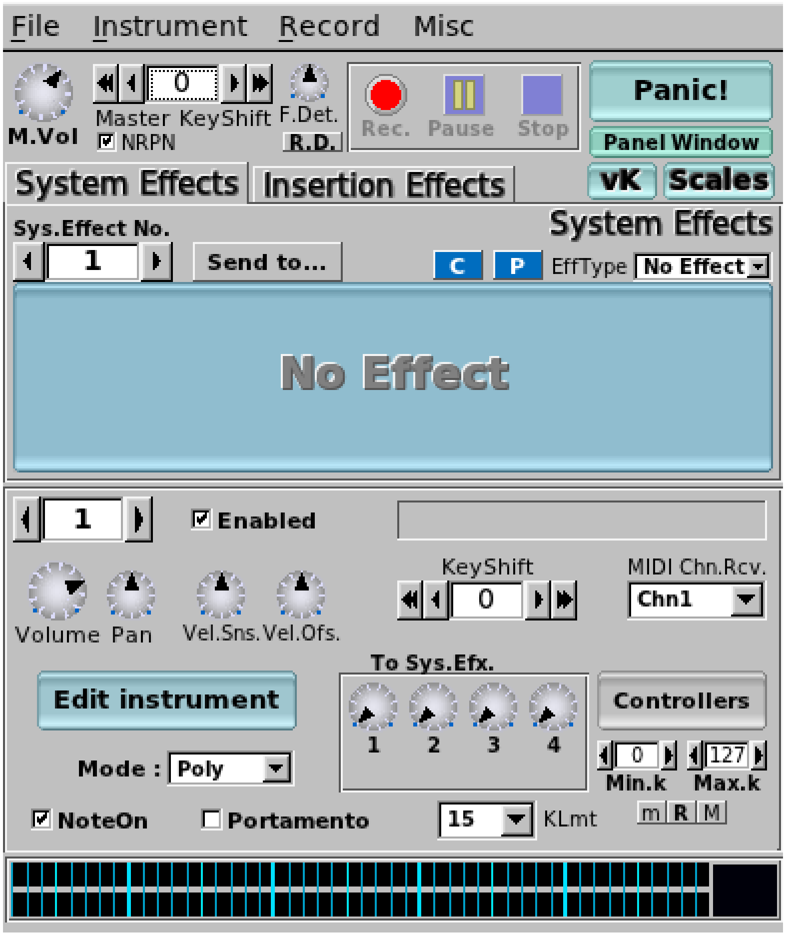


Figure 1. Main Window

For basic usage, you will want to use the button to the right of the enabled label. This button will allow for one to select the desired instrument from the banks that ZynAddSubFX has available. To play notes in ZynAddSubFX, either utilize the builtin virtual keyboard (accessible via the vK button) or connect your keyboard to the system and use **aconnect** to connect it to ZynAddSubFX (assuming that ALSA was used).

This main window provides access to a number of more advanced features. Some of these features are:

- System Effects
- Insertion Effects
- Recording
- Part Settings (instrument level settings)
- Master Settings
- Microtonal Settings

For instance to use the recording feature, a wave file must be selected from the recording menu and then the recording can be started with the record button and stopped with the stop button. This is a simple and quick way of recording some samples from ZynAddSubFX, though there are more full featured options available via JACK recording tools.

NOTE	After hitting record, the wave file will not start recording until a new key has been pressed via either an external midi source or the virtual keyboard Both system and insertion effects can be accessed, the properties are available as well as properties of each instrument.

Filters

ZynAddSubFX offers several different types of filters, which can be used to shape the spectrum of a signal. The primary parameters that affect the characteristics of the filter are the cutoff, resonance, filter stages, and the filter type.

- **Cutoff**: This value determines which frequency marks the changing point for the filter. In a low pass filter, this value marks the point where higher frequencies are attenuated.
- **Resonance**: The resonance of a filter determines how much excess energy is present at the cutoff frequency. In ZynAddSubFX, this is represented by the Q-factor, which is defined to be the cutoff frequency divided by the bandwidth. In other words higher Q values result in a much more narrow resonant spike.
- **Stages**: The number of stages in a given filter describes how sharply it is able to make changes in the frequency response.

The basic *analog* filters that ZynAddSubFX offers are shown below, with the center frequency being marked by the red line. The *state variable* filters should look quite similar.

[filter0]

As previously mentioned, the Q value of a filter affects how concentrated the signal's energy is at the cutoff frequency; The result of differing Q values are below.

TIP For many classical analog sounds, high Q values were used on sweeping filters. A simple high Q low pass filter modulated by a strong envelope is usually sufficient to get a good sound.

[filter1]

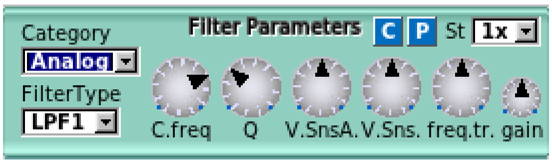
Lastly, the affect of the order of the filter can be seen below. This is roughly synonymous with the number of stages of the filter. For more complex patches it is important to realize that the extra sharpness in the filter does not come for free as it requires many more calculations being performed; This phenomena is the most visible in subsynth, where it is easy to need several hundred filter stages to produce a given note.

[filter2]

There are different types of filters. The number of poles define what will happen at a given frequency. Mathematically, the filters are functions which have poles that correspond to that frequency. Usually, two poles mean that the function has more "steepness", and that you can set the exact value of the function at the poles by defining the "resonance value". Filters with two poles are also often referenced as Butterworth Filters.

> For the interested, functions having poles means that we are given a quotient of polynomials. The denominator has degree 1 or 2, depending on the filter having one or two poles. In the file *DSP/AnalogFilter.cpp*, *AnalogFilter::computefiltercoefs()* sets the coefficients (depending on the filter type), and *AnalogFilter::singlefilterout()* shows the whole polynomial (in a formula where no quotient is needed).

User Interface



- **C.freq**: Cutoff frequency
- **Q**: Level of resonance for the filter
- **V.SnsA.**: Velocity sensing amount for filter cutoff
- **V.Sns.**: Velocity sensing function
- **freq.tr**: Frequency tracking amount. When this parameter is positive, higher note frequencies shift the filter's cutoff frequency higher.
- **gain**: Additional gain/attenuation for filter
- **St**: Filter stages

NOTE TODO add a lengthy section on the formant filter setup

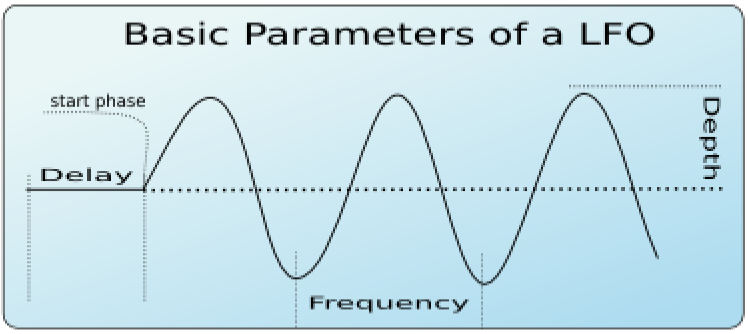
LFO

Introduction

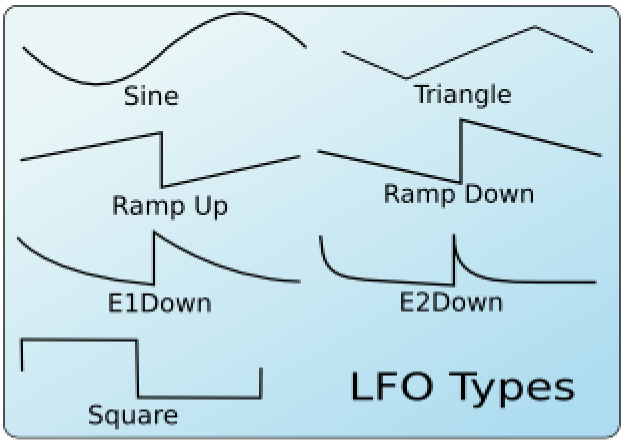
"LFO" means Low Frequency Oscillator. These oscillators are not used to make sounds by themselves, but they changes some parameters (like the frequencies, the amplitudes or the filters).

The LFOs have some basic parameters:

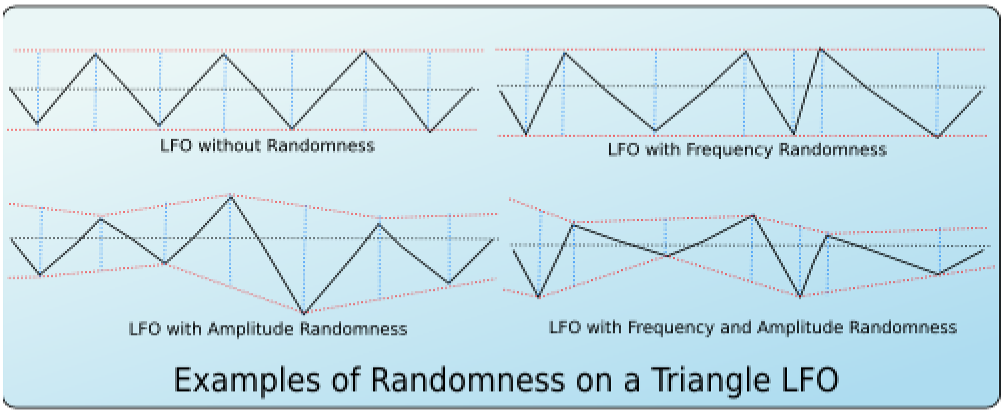
- **Delay**: This parameter sets how much time takes since the start of the note to the start of the LFO
- **Start Phase**: The position that a LFO will start at
- **Frequency**: How fast the LFO is (i.e. how fast the parameter's controlled by the LFO changes)
- **Depth**: The amplitude of the LFO (i.e. how much the parameter's controlled by the LFO changes)



Another important LFO parameter is the shape. There are many LFO Types according to the shape. ZynAddSubFX supports the following LFO shapes:



Another parameter is the LFO Randomness. It modifies the LFO amplitude or the LFO frequency at random. In ZynAddSubFX you can choose how much the LFO frequency or LFO amplitude changes by this parameter. In the following images are shown some examples of randomness and how changes the shape of a triangle LFO.



Other parameters are:

- **Continuous mode**: If this mode is used, the LFO will not start from "zero" on each new note, but it will be continuous. This is very useful if you apply on filters to make interesting sweeps.
- **Stretch**: It controls how much the LFO frequency changes according to the note’s frequency. It can vary from negative stretch (the LFO frequency is decreased on higher notes) to zero (the LFO frequency will be the same on all notes) to positive stretch (the LFO frequency will be increased on higher notes).

User Interface

In ZynAddSubFX, LFO parameters are shown as:

These parameters are:

- **Freq**: LFO Frequency
- **Depth**: LFO Depth
- **Start**: LFO Start Phase - If this knob is at the lowest value, the LFO Start Phase will be random.
- **Delay**: LFO Delay
- **A.R.**: LFO Amplitude Randomness
- **F.R.**: LFO Frequency Randomness
- **C.**: LFO Continuous Mode
- **Str.**: LFO Stretch - in the image above the LFO stretch is set to zero

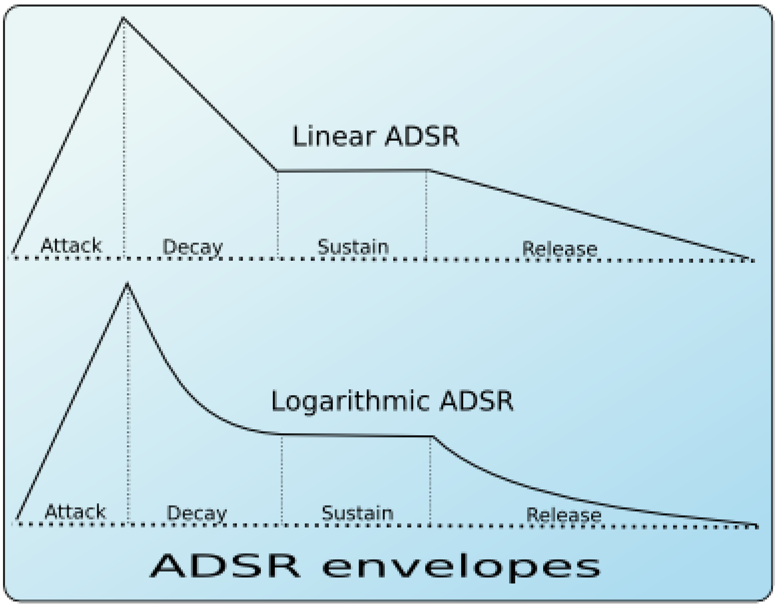
Envelopes

Introduction

Envelopes control how the amplitude, the frequency, or the filter changes over time.

Amplitude Envelopes

These envelopes controls the amplitude of the sound. In ZynAddSubFX, amplitude envelopes can be linear or logarithmic. In the next image, it is shown the differences between these envelopes.

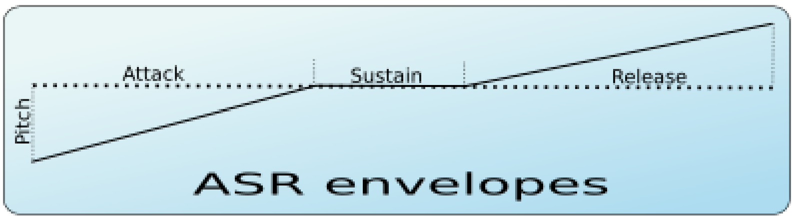


The amplitude envelope is divided into:

- **Attack**: Begins at the Note On. The volume starts from 0 to the maximum. In ZynAddSubFX, the attack is always linear.
- **Decay**: The volume drops from the maximum value to a level called "Sustain level"
- **Sustain**: The volume remains constant until the key is depressed (Note Off). After this, the last stage take place.
- **Release**: The volume drops to zero

Frequency Envelopes

These envelopes controls the frequency (more exactly, the pitch) of the oscillators. The following picture draws the stages of these envelopes.



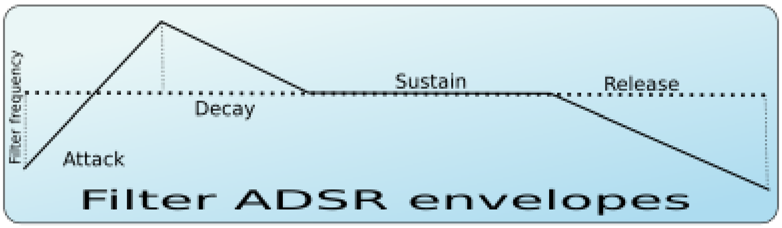
The dotted line represents the real pitch of the sound without the envelope.

The frequency envelopes are divided into 3 stages:

- **Attack**: Begins at the Note On. The frequency starts from a certain value and glides to the real frequency of the note.
- **Sustain**: The frequency is the same on over the sustain period
- **Release**: This stage begins on Note Off and glides the frequency of the note to a certain value

Filter Envelopes

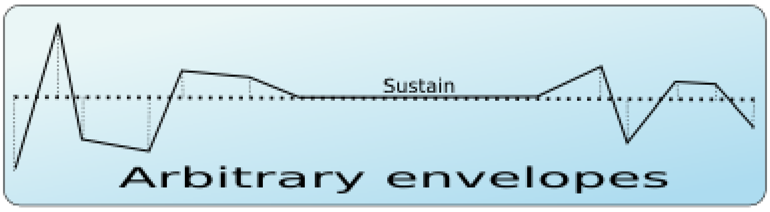
These envelopes controls the cutoff frequency of the filters and are divided into



- **Attack**: Begins at the Note On. The cutoff frequency starts from a certain value and glides to another value
- **Decay**: The cutoff frequency continues to glide to the real cutoff frequency value of the filter (dotted line)
- **Sustain**: the cutoff frequency is the same on over the sustain period (dotted line)
- **Release**: this stage begins on Note Off and glides the filter cutoff frequency of the note to a certain value

Freemode Envelopes

For all envelope there is a mode that allows the user to set an arbitrary number of stages and control points. This mode is called Freemode.



Only stage that always remains defined is the Sustain, where the envelopes freezes until a Note Off event.

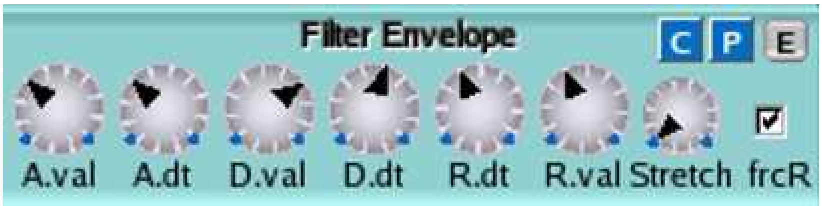
User Interface

All the envelope types has some common controls:

- **E**: Shows a window that you can view the real envelope shape or convert to free mode to edit it
- **Stretch**: How the envelope is stretched according the note. On the higher notes the envelopes are shorter than lower notes. In the leftmost value, the stretch is zero. The rightmost use a stretch of 200%; this means that the envelope is stretched about 4 times/octave.

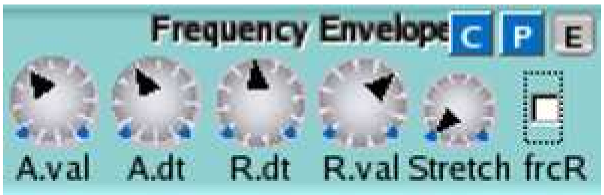
- **frcR**: Forced release. This means that if this option is turned on, the release will go to the final value, even if the sustain stage is not reached. Usually, this must be set.

The parameters for Amplitude Envelopes for ZynAddSubFX are:



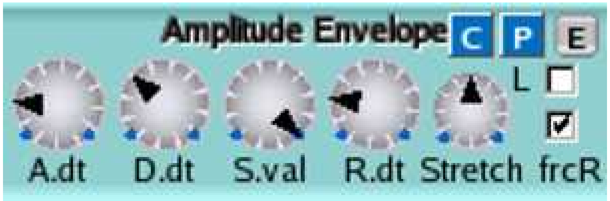
- **A.dt**: Attack duration
- **D.dt**: Decay duration
- **S.Val**: Sustain value
- **R.dt**: Release time
- **L**: If this option is set, the envelope is linear, otherwise, it will be logarithmic

For Frequency Envelopes the interface has the following parameters:



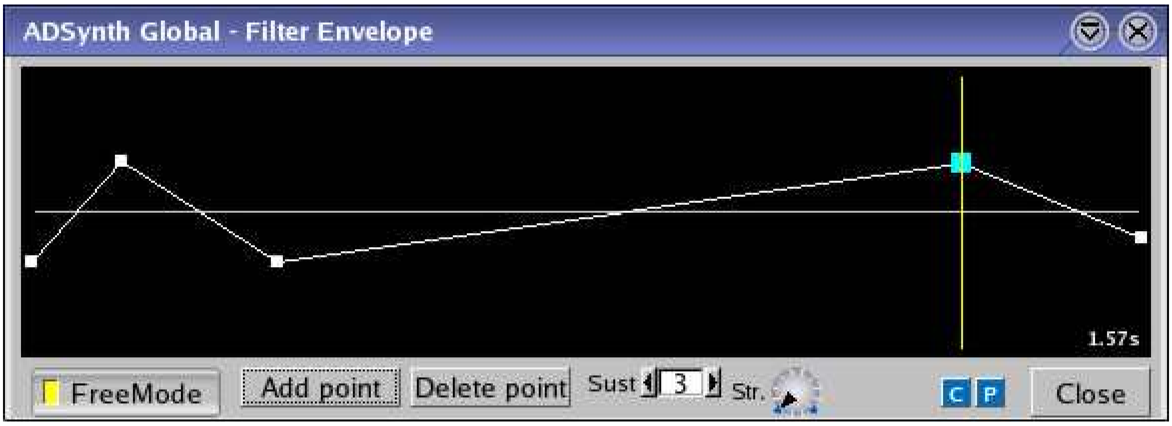
- **A.val**: Attack value
- **A.dt**: Attack duration
- **R.dt**: Release time
- **R.val**: Release value

Filter Envelopes has the parameters:



- **A.val**: Attack value
- **A.dt**: Attack duration
- **D.val**: Decay value
- **D.dt**: Decay time
- **R.dt**: Release time
- **R.val**: Release value

The Freemode envelopes has a separate window to set the parameters and controls:



- **Control points**: You can move the points using the mouse. In the right on the windows, it is shown the total duration of the envelope. If the mouse button will be pressed on a control point, it will be shown the duration of the stage where the point is.
- **FreeMode**: this button activates or deactivates the freemode mode.
- **Add Point**: Adds the point next to the current selected point. You can select a point by clicking on it.
- **Delete point**: Removes the point from the envelope.
- **Sust.**: Set the sustain point. It is shown using the yellow line.
- **Str.**: Envelope stretch

AdSynth

AdSynth, a primarily additive synthesis engine, is one of the three major synthesis engines available in ZynAddSubFX. The basic concept of this engine is the summation of a collection of voices, each of which consist of oscillators.

High Level (Global)

AdSynth's global level consists of the elements shown in the below figure:

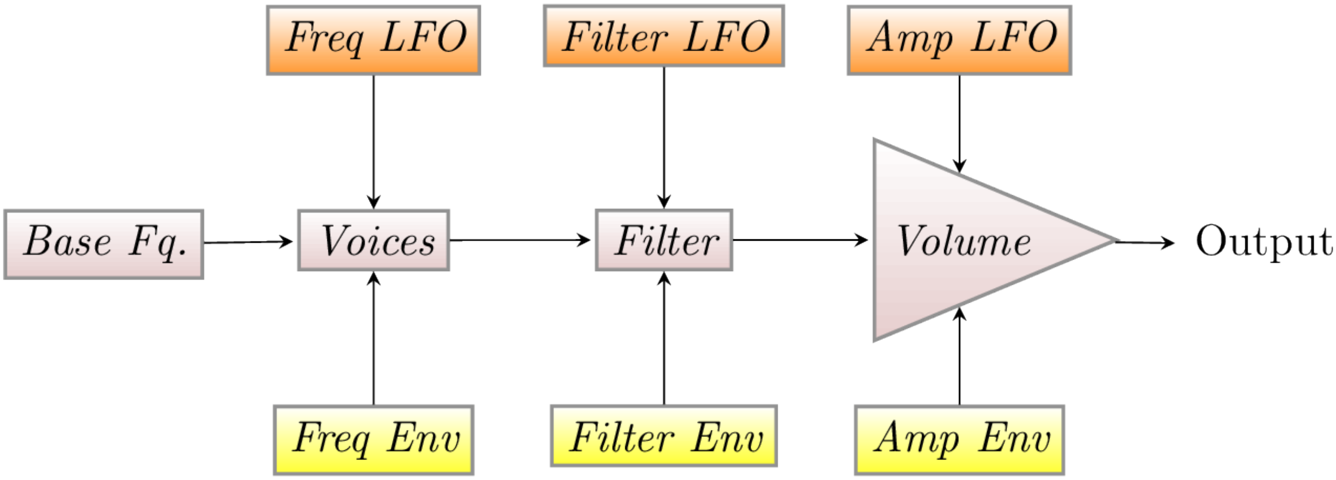


Figure 2. AdSynth Global Elements

The global level of adsynth is almost entirely composed of previously discussed elements. However a few new features appear here, this includes velocity sensing, punch, detune options and relative bandwidth , and resonance.

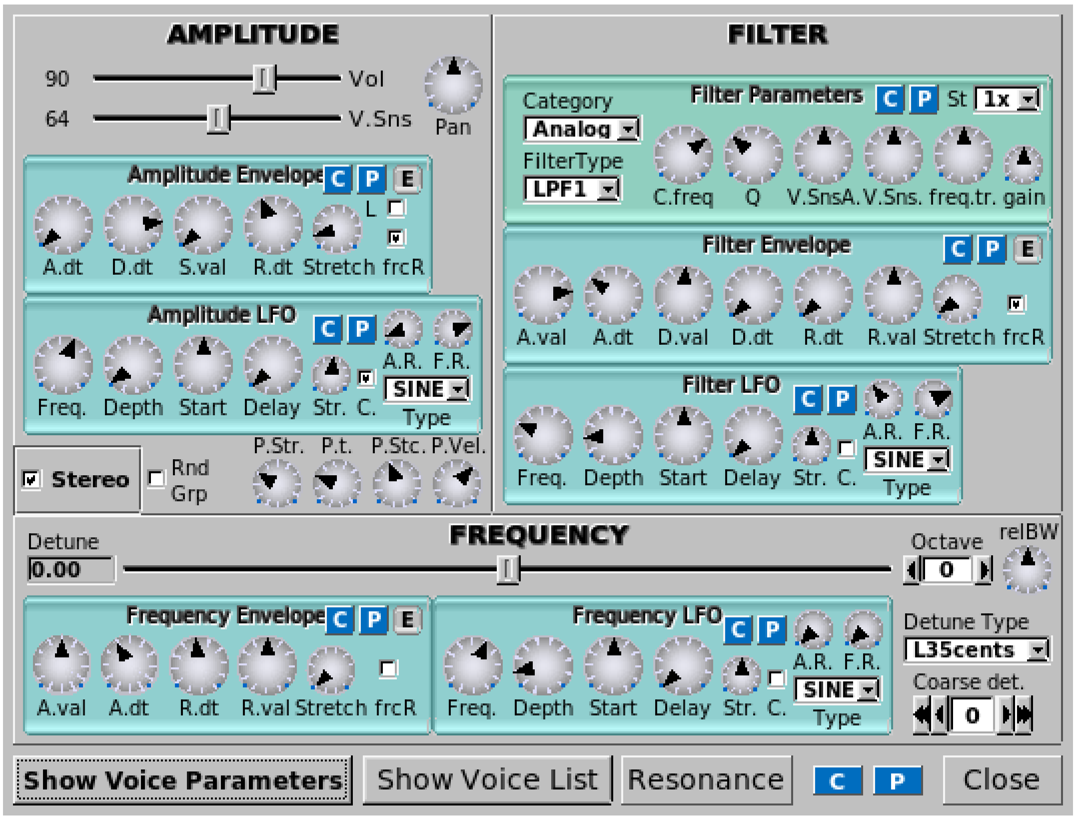


Figure 3. AdSynth Global Window

Velocity sensing is simply an exponential transformation from the note's velocity to some parameter change. The below diagram shows how the velocity sensing controls affects this translation over the whole range of possible note velocities.

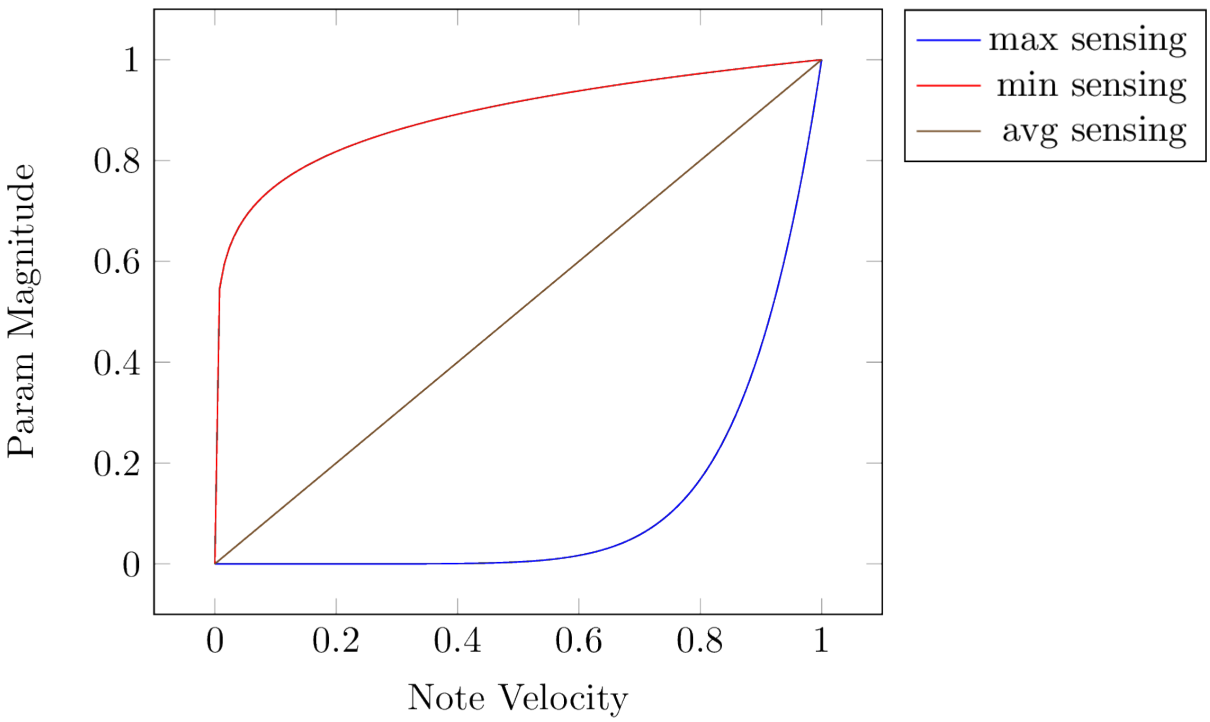


Figure 4. Velocity Sensing Chart

The punch of a note in AdSynth is a constant amplification to the output at the start of the note, with its length determined by the punch time and stretch and the amplitude being determined by the punch strength and velocity sensing. The relBW control in the frequency pane is effectively a multiplier for detuning all voices within an adnote.

NOTE	TODO Talk about resonance

The sum of the voices are passed through filters and amplification to produce the final sound. This could lead one to think that ad-note is just a bunch of minor postprocessing and at this level much of the sound generation is hidden.

Voices

The voice gives access to a similar setup to the global parameters and then some more, such as the modulator, oscillator, and unison features.

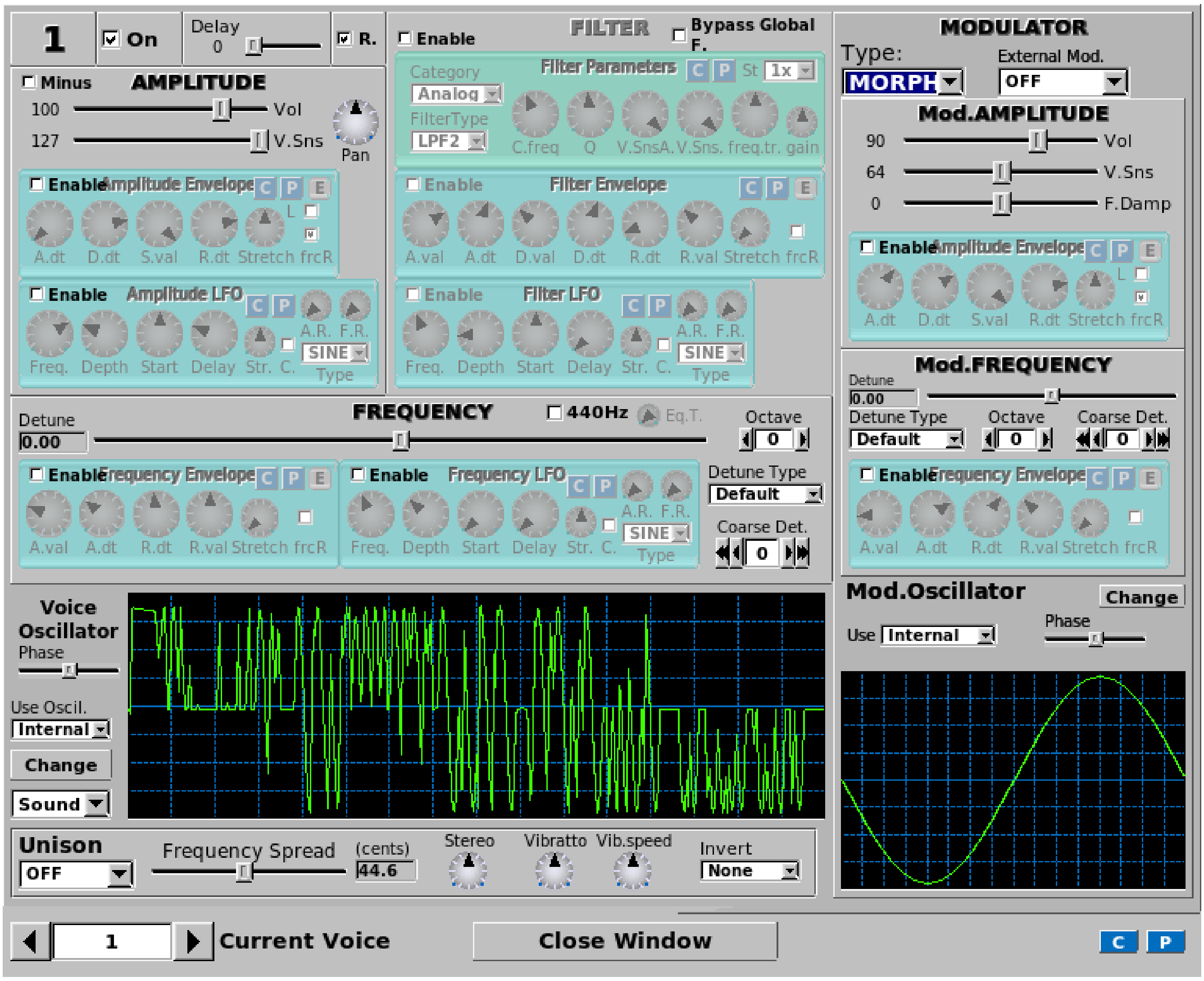


Figure 5. AdSynth Voice Window

Modulation

Within the options for modulation, one can select:

- Morph

- Ring Modulation
- Phase Modulation
- Frequency Modulation
- Disabled

Unison

Unison is useful in creating the chorus like sound of many simultaneous oscillators

Oscillator

The oscillator is lets you choose the basic waveform, which oscillates while the sound is playing and is then further modified.

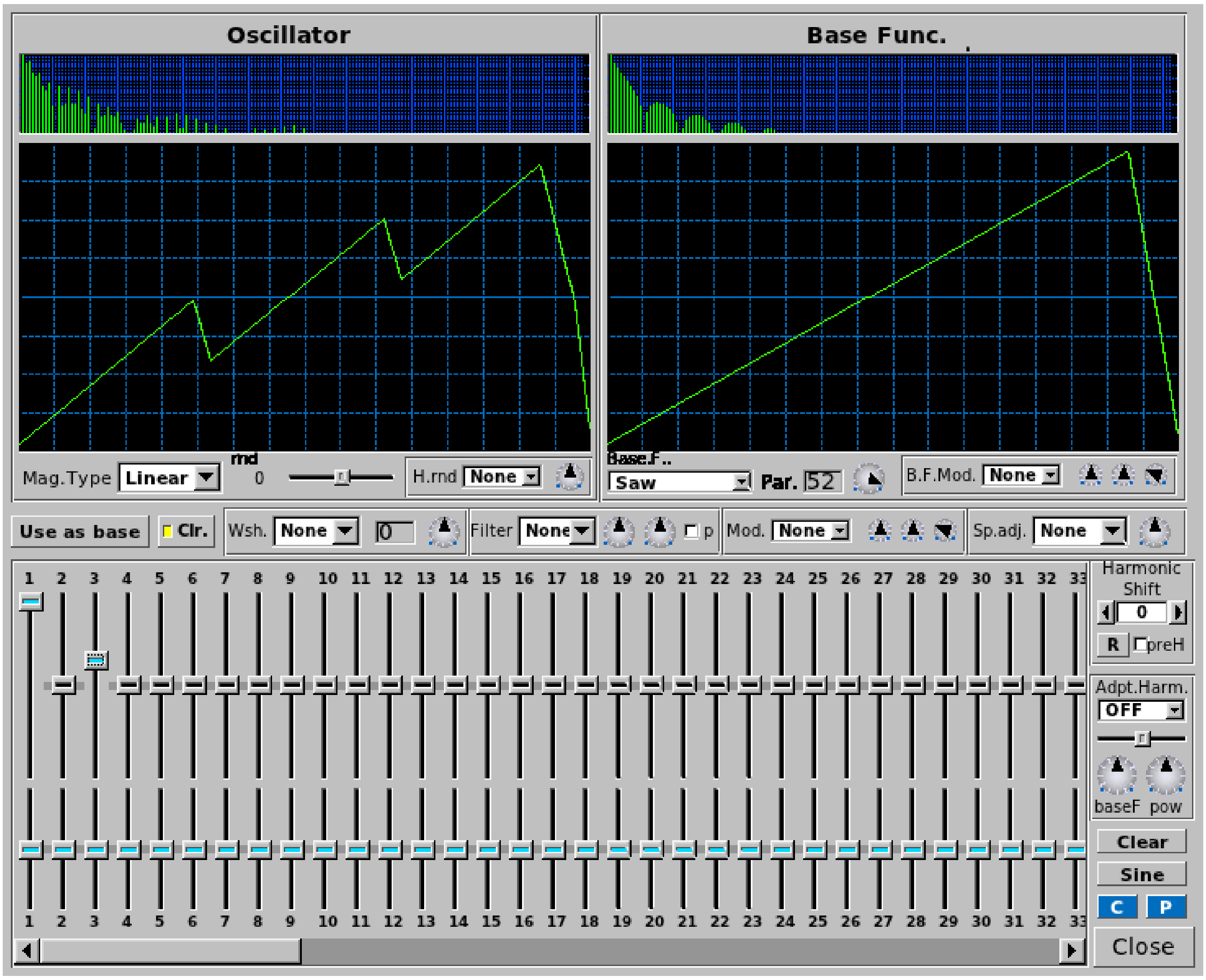


Figure 6. Oscillator Window

Types of Waveshaping

Waveshaping can be done using the **Wsh** area in the Oscillator editor.

The type of distortion has much influence on how the overtones are being placed. Sometimes, you get a "fat" bass, and sometimes, high frequencies are added, making the sound "crystal clear".

Atan & Sigmoid

This is the default setting. It is an easy way to apply loudness to a wave without getting undesired high overtones. Thus, it can be used both for making instruments that sound like "real" ones, but also for electronic music. The transformation turns, roughly said, every amplitude into a square amplitude. Thus, sine, power, pulse and triangle turn into a usual square wave, while a saw turns into a phased square wave. A chirp wave turns into a kind of phase modulated square wave.

Quants

Quantization adds high overtones early. It can be seen as an unnatural effect, which is often used for electronic music.

The transformation is a bit similar to building the lower sum of a wave, mathematically said. This means that the transformation effect turns your "endless high" sampled wave into only a few samples. The more distortion you will apply, the less samples will be used. Indeed, this is equivalent to say that more input amplification is used. To see this, here is a small sample of code, where "ws" is the (correctly scaled amount of input amplification, and "n" the number of original samples.

```
for(i = 0; i < n; ++i)
  smps[i] = floor(smps[i] / ws + 0.5f) * ws;
```

NOTE	If you turn on quantisation very high, you might be confused that, especially high notes, make no sound. The reason: High frequencies are "forgotten" if you sample with only few samples. Also, the sign of an amplitude can be forgotten. This behaviour might make some quantisations a bit unexpected.

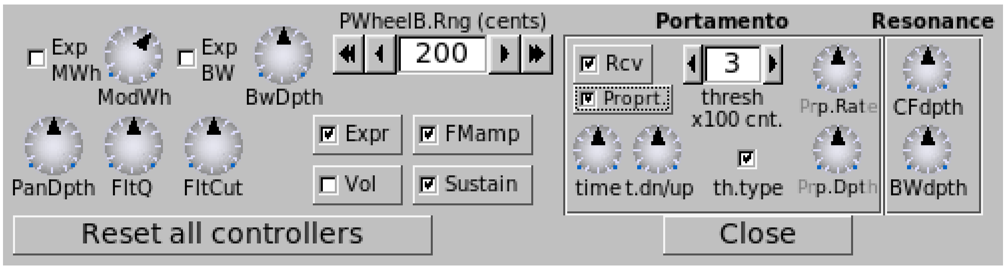
Limiting & Clipping

Limiting usually means that for a signal, the amplitude is modified because it exceeds its maximum value. Overdrive, as often used for guitars, is often achieved by limiting: It happens because an amplifier "overdrives" the maximum amplitude it can deliver.

ZynAddSubFX has two types of limiting. Soft limiting, here as **Lmt**, means that the sound may not exceed a certain value. If the amplitude does so, it will simply be reduced to the limiting value. The overtones are generated in the lower frequencies first.

Hard limiting, is also called clipping and abbreviated **Clip**. This means that if the maximum is exceeded, instead of being constant at the limiting value, the original signal still has some influence on the output signal. Still, it does not exceed the limiting value. For ZynAddSubFX, a signal exceeding the limiting value will continue to grow "in the negative". This leads to overtones being generated on the full frequency band.

Controller



General

- **ModWh**: Modulation Wheel depth
- **Exp MWh**: Exponential Modulation Wheel (changes modulation scale to exponential)
- **BwDpth**: Bandwidth Depth
- **Exp BW**: Exponential Bandwidth (changes bandwidth scale to exponential)
- **PanDpth**: Panning Depth
- **FltQ**: Filter Q (resonance) depth
- **FltCut** Filter Cutoff frequency depth
- **Expr**: enable/disable expression
- **Vol**: enable/disable receiving volume controller
- **FMamp**: enable/disable receiving Modulation Amplitude controller (76)
- **Sustain**: enable/disable sustain pedal
- **PWheelB.Rng (cents)**: Pitch Wheel Bend Range (cents; 100 cents = 1 halftone)

Portamento

- **Rcv.**: If the part receives portamento On/Off (65) controller
- **time**: The duration of the portamento
- **thresh**: The threshold of the portamento. It represents the minimum or the maximum number of halftones (or hundred cents) required to start the portamento. The difference is computed between the last note and current note.
- **th.type**: The threshold type. Checked means that the portamento activates when the difference of frequencies is above the threshold ("thresh"); not checked is for below the threshold.

NOTE	The threshold refers to the frequencies and not to MIDI notes (you should consider this if you use microtonal scales).

Proportional Portamento

- **Propt.**: If the portamento is proportional to ratio of frequencies

- **Prp. Rate**: Ratio needed to double the time of portamento
- **Prp. Dpth**: The divergence from

Resonance

- **CFdpth**: resonance center controller depth
- **BWdpth**: resonance bandwidth controller depth

Effects

Effects are, generally, black boxes that transform audio signals in a specified way. More exactly, the only input data for an effect in ZynAddSubFX is:

- an array of samples, which is read **on line**
- the current system time (used for LFOs)

The output is the transformed array of samples.

NOTE	As described, effects have no information about anything else. For example, key presses are not recognized. Therefore, pressing a key does not initiate the LFO. Phase knobs will always be relative to a **global** LFO, which is only dependent on the system time.

ZynAddSubFX has 3 types of effects:

- System Effects
- Insertion Effects
- Instrument Effects

TODO: Describe these 3 types (their differences).

General topics

- **Wetness** determines the mix of the results of the effect and its input. This mix is made the effects output. If an effect is wet, it means that nothing of the input signal is bypassing the effect. If it is dry, then the effect has no effect. TODO: Difference between Volume and D/W?
- **Pan** lets you apply panning, which means that the sound source can move to the right or left. Set it to 0.0 to only hear output on the right side, or to the maximum value to only hear output on the left side.
- **LRc.** or **L/R** let you apply crossover.
- **Filter stages** are the number of times that this filter is applied in series. So, if this number is 1, you simply have this one filter. If it is two, the sound first passes the filter, and the results then pass the same filter again. In ZynAddSubFX, the wetness is applied after all stages were passed.
- **LFOs** are, as the name says, oscillators with, compared to the frequency of the sound, low

frequency. They often appear in order to control the effect. They can have some of the following controls:

- **LFO Type** determines the shape of the LFO. If not present, the LFO is a sine wave.
- **Freq** determines the LFO’s frequency.
- **Dpth** is a multiplier to the LFO. Thus, it determines the LFOs amplitude and its influence.
- **Rnd** is the LFO amplitude’s randomness
- **St.df** lets you determine how much left and right LFO are phase shifted. 64.0 means stereo, higher values increase the right LFO relatively to the left one.

> Hint: Keep in mind that Effects that can be controlled by LFO can also be controlled arbitrary: Set the LFO depth to zero and manipulate the phase knob (e.g. with NRPNs or maybe via OSC in the future).

Equalizer

Introduction

An equalizer is a filter effect that applies different volume to different frequencies of the input signal. This can, for example, be used to "filter out" unwanted frequencies. ZynAddSubFX’s implementations follow the "Cookbook formulae for audio EQ" by Robert Bristow-Johnson.

Filter Types

This topic is completely discussed in the Filters section.

Usage

We describe all parts of the GUI here. The term passband (or often just "band") refers to the amount of frequencies which are not significantly attenuated by the filter.

- **Gain** (on the left) defines an offset which is added to the complete filter.
- **B.** lets you choose the passband number. Multiple passbands define one filter. This is important if you want multiple filters to be called after each other. Note that filters are commutative.
- **T.** lets you choose the current filter’s type, as described above.
- **Freq** describes the frequencies where the filter has its poles. For some filters, this is called the "cutoff" frequency. Note, however, that a bandpass filter has two cutoff frequencies.
- **Gain** is only active for some filters and sets the amount of a special peak these filters have. Note that for those filters, using the predefined gain makes them effectless.
- **Resonance** lets you describe a peak at the given frequency for filters with 2 poles. This can be compared to real physical objects that have more gain at their resonance frequency.
- **St.** lets you define multiple filter stages. This is equivalent to having multiple copies of the same filter in sequence.

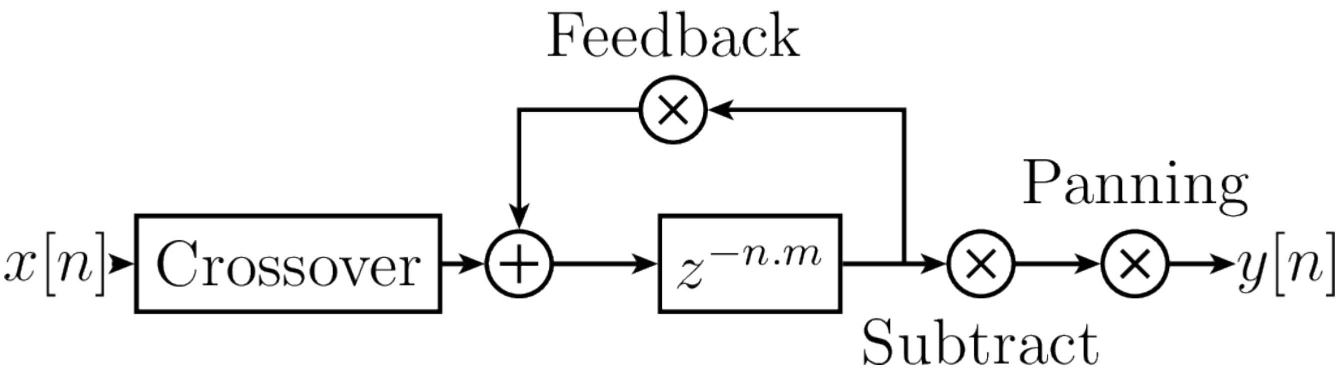
Chorus

Introduction

In a chorus, many people sing together. Even if each of them sings at exactly the same frequency, all their voices usually sound different. We say they have a different timbre. Timbre is the way we perceive sound and makes us differ between different music instruments. This is, physically, achieved by varying both the amplitude envelope and the frequency spectrum. Multiple sounds with slightly different timbres make a sound more shimmering, or powerful. This is called the chorus effect.

Function

The chorus effect can be achieved by multiple people singing together. In a concert, there are many instruments, resulting in the same effect. When making electronic music, we only have an input wave and need to generate these different timbres by ourselves. ZynAddSubFX therefore simply plays the sound, pitch modulated by an LFO, and adds this to the original sound. This explains the diagram below: The multiple pitches are generated by a delayed version of the input. This version is being pitched by an LFO. More detailed, this pitch is generated by varying the reading speed of the delayed sound; the variation amount is controlled by an LFO.



TODO: Add LFO pointing to delay?

Related effects to Chorus are Flangers. Flangers can be described as Chorus with very short LFO delay and little LFO depth. You can imagine a flanger as two copies of a sound playing at almost the same time. This leads to interference, which can be clearly heard. It is popular to apply flangers to guitars, giving them more "character".

Usage

- First, crossover is applied.
- The following 5 knobs (**Freq**, **Rnd**, **LFO Type**, **St.df**, **Depth**) control the LFO for the pitch. If the depth is set to zero, the pitch will not be changed at all.
- **Delay** is the time that the delayed sound is delayed "on average". Note that the delay also depends on the current pitch.
- After the correct element of the sound buffer is found using the LFO, the **Fb** knob lets you set how loud it shall be played. This is mostly redundant to the **D/W** knob, but we have not applied panning and subtraction yet.

- Next, the signal can be negated. If the **Substract** checkbox is activated, the amplitude is multiplied by -1.
- Finally, **Pan** lets you apply panning.

Distortion

Introduction

Distortion means, in general, altering a signal. Natural instruments usually produce sine like waves. A wave is transformed in an unnatural way when distortion is used. The most distorted waves are usually pulse waves. It is typical for distortion to add overtones to a sound. Distortion often increases the power and the loudness of a signal, while the db level is not increased. This is an important topic in the Loudness War.

NOTE	As distortion increases loudness, distorted music can cause ear damage at lower volume levels. Thus, you might want to use it a bit careful.

Distortion can happen in many situations when working with audio. Often, this is not wanted. In classical music, for example, distortion does not occur naturally. However, distortion can also be a wanted effect. It is typical for Rock guitars, but also present in electronic music, mostly in Dubstep and Drum & Bass.

The basic components of distortion are mainly

- a preamplifier
- the waveshaping function
- filters

Preamplification changes the volume before the wave is shaped, and is indeed the amount of distortion. For example, if you clip a signal, the louder the input gets, the more distortion you will get. This can have different meanings for different types of distortions, as described below.

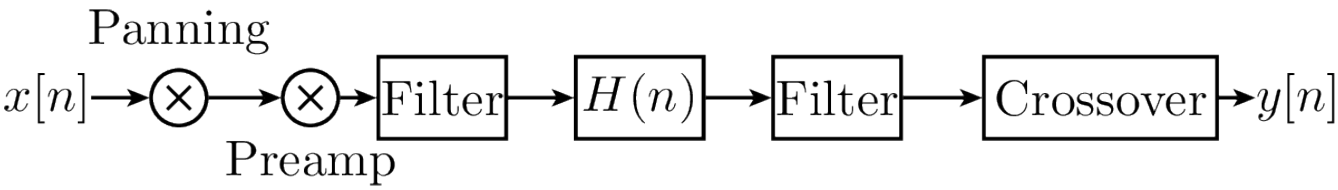
> The filters are practical. A reason for using them afterwards is that distortion can lead to waves with undesired high frequency parts. Those can be filtered out using the LPF. A reason for using filters before applying is to achieve multiband distortion. ZynAddSubFX has no "real" multiband distortion by now, however.

Types of Distortion

This topic is completely discussed in the Oscillator Section. Note that you can use the Oscillator editor in order to find out what your distortion effect does. Also note that while the Oscillator editor’s distortion is limited to some oscillators you can produce in the Oscillator editor, the distortion effect can be used on every wave that you can generate with ZynAddSubFX.

Function

We explain the functionality in a diagram and list the components below.



- Negation is the first thing to happen. If the **Neg** checkbox is activated, the amplitude is multiplied by -1.
- Panning is applied. Note, however, that you have to activate the Stereo Checkbox, labeled **St**, before.
- Preamplification is done next. The amount can be changed using the **Drive** nob. Indeed, this is the amount of distortion. For example, if you clip a signal, the louder the input gets, the more distortion you will get. This can have different meanings for different types of distortion, as described above.
- **HPF** and **LPF** are filters with 2 poles. Whether they are used before or after the waveshape, depends on the checkbox labeled **PF**.
- The next step is the wave shape. This defines how the wave is actually modified. The **Type** combo box lets you define how. We will discuss some types below.
- After the wave shape, we scale the level again. This is called output amplification. You can change the value using the **Level** knob.
- Crossover is the last step. This is controlled by the knob **LR Mix** and means that afterwards, a percentage of the left side is applied to the right side, and, synchronously, the other way round. It is a kind of interpolation between left and right. If you set the LR Mix to 0.0, you will always have a stereo output.

Dynamic Filter

Introduction

A dynamic filter is, as the name says, a filter which changes its parameters dynamically, dependent on the input and current time. In ZynAddSubFX, frequency is the only variable parameter. It can be used as an "envelope following filter" (sometimes referenced "Auto Wah" or simply "envelope filter").

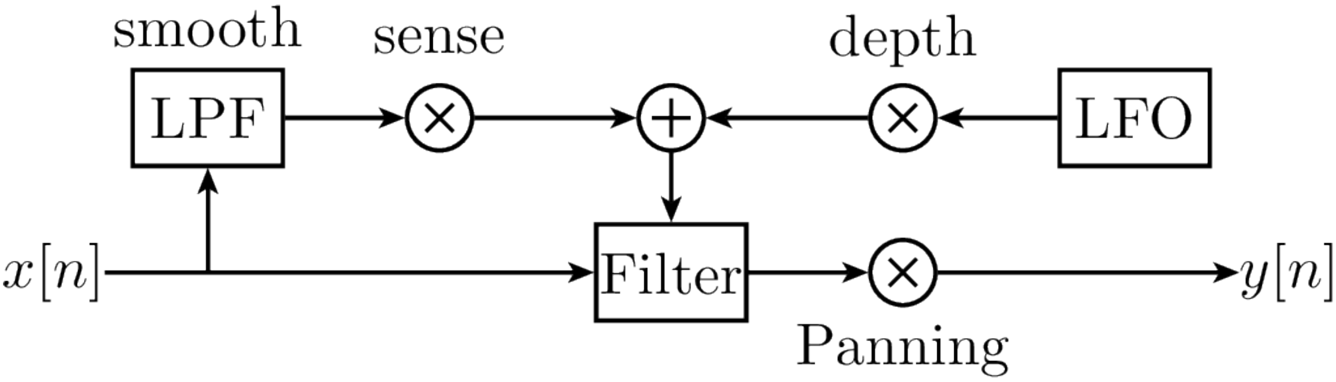
Function

Though this filter might look a bit complicated, it is actually easy. We divide the parameters into two classes:

- **Filter Parameters** are the ones you get when you click on **Filter**. They give the filter its basic settings.
- **Effect Parameters** are the other ones that control how the filter changes.

The filter basically works like this: The input signal is passed through a filter which dynamically changes its frequency. The frequency is an additive of:

- the filter's base frequency
- an LFO from the effect parameters
- the "amplitude" of the input wave



The amplitude of the input wave is not the current amplitude, but the so called "Root Mean Square (RMS)" value. This means that we build a mean on the current amplitude and the past values. How much the new amplitude takes influence is determined by the **Amplitude Smoothness** (see below).

> RMS value plays an important role in the term loudness. A fully distorted signal can sound 20 db louder due to its higher RMS value. This filter takes this into account, depending on the smoothness.

Usage

- The 4 knobs in the middle (**Freq**, **Rnd**, **LFO Type**, **St.df**) control the LFO.
- Two knobs let you control the way how the RMS value of the amplitudes is measured:
  - **A.M** sets the Amplitude Smoothness (this is described above). The higher you set this value, the more slow will the filter react.
  - **A.Inv.**, if being set, negates the (absolute) RMS value. This will lower the filter frequency instead of increasing it. Note that this will not have much effect if the effects input is not very loud.
- The following controls define the mix of the LFO and the amplitude.
  - **A.S** sets the Amplitude Sensing (i.e. how much influence the amplitude shall have).
  - **LfoD** sets the LFO depth.
- The filter button lets you choose the filter type.
- After the input signal has passed through the filter, **Pan** can apply panning.

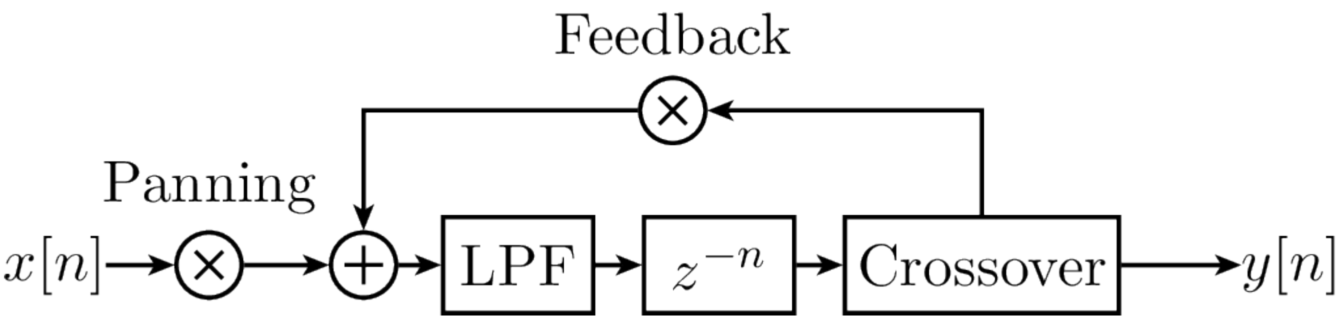
Echo

Introduction

The echo effect, also known as delay effect, simulates the natural reflection of a sound. The listener can hear the sound multiple times, usually decreasing in volume. Echos can be useful to fill empty parts of your songs with.

Function

In ZynAddSubFX, the echo is basically implemented as the addition of the current sound and a delayed version of it. The delay is implemented as in the picture below. First, we add the delayed signal to the effect input. Then, they pass an LP1. This shall simulate the effect of dampening, which means that low and especially high frequencies get lost earlier over distance than middle frequencies do. Next, the sound is delayed, and then it will be output and added to the input.



> The exact formula in the source code for the dampening effect is as follows:
>
> $Y(t) := (1-d) \cdot X(t) + d \cdot Y(t-1)$,
>
> where t be the time index for the input buffer, d be the dampening amount and X,Y be the input, respective the output of the dampening. This solves to
>
> $Y(z) = Z(Y(t)) = (1-d) \cdot X(z) + d \cdot Y(z) \cdot z^{-1}$
>
> $\Leftrightarrow H(z) := \frac{Y(z)}{X(z)} = \frac{1-d}{1 - d \cdot z^{-1}}$
>
> which is used in $Y(z) = H(z) \cdot X(z)$. So $H(z)$ is indeed a filter, and by looking at it, we see that it is an LP1. Note that infinite looping for d=1 is impossible.

Description

- **Pan** lets you apply panning of the input.
- **Delay** sets the time for one delay.
- **LRdl.** means Left-Right-Delay. If it is set to the middle, then both sides are delayed equally. If not, then the left echo comes earlier and the right echo comes (the same amount) later than the average echo; or the other way round. Set the knob to 0 to hear on the right first.
- **LRc.** applies crossover.
- Feedback describes how much of the delay is added back to the input. Set **Fb.** to the maximum to hear an infinite echo, or to the minimum to just hear a single repeat.

- The **Damp** value lets the LP1 reject higher frequencies earlier if increased.

Reverb

Introduction

A Reverberation actually expresses the effect of many echoes being played at the same time. This can happen in an enclosed room, where the sound can be reflected in different angles. Also, in nature, thunders approximate reverbs, because the sound is reflected in many different ways, arriving at the listener at different times.

In music, reverbs are popular in many ways. Reverbs with large room size can be used to emulate sounds like in live concerts. This is useful for voices, pads, and hand claps. A small room size can simulate the sound board of string instruments, like guitars or pianos.

Function

As mentioned, a reverb consists of permanent echo. The reverb in ZynAddSubFX is more complex than the echo. After the delaying, comb filters and then allpass filters are being applied. These make the resulting sound more realistic. The parameters for these filters depend on the room size. For details, consider the information about Freeverb.

Description

- The **Type** combo box lets you select a reverb type:
  - **Freeverb** is a preset. It was proposed by Jezar at Dreampoint.
  - **Bandwidth** has the same parameters for the comb and allpass filters, but it applies a unison before the LP/HP. The unison's bandwidth can be set using **bw**.
  - Random chooses a random layout for comb and allpass each time the type or the room size is being changed.
- The room size (**R.S.**) defines parameters only for the comb and allpass filters.
- **Time** controls how long the whole reverb shall take, including how slow the volume is decreased.
- The initial delay (**I.del**) is the time which the sounds need at least to return to the user. The initial delay feedback (**I.delfb**) says how much of the delayed sound is added to the input.
- Low pass filter (**LPF**) and high pass filter (**HPF**) can be applied before the comb filters.
- The dampening control (**Damp**) currently only allows to damp low frequencies. Its parameters are being used by the comb and allpass filters.
- **Pan** lets you apply panning. This is the last thing to happen.

Phaser

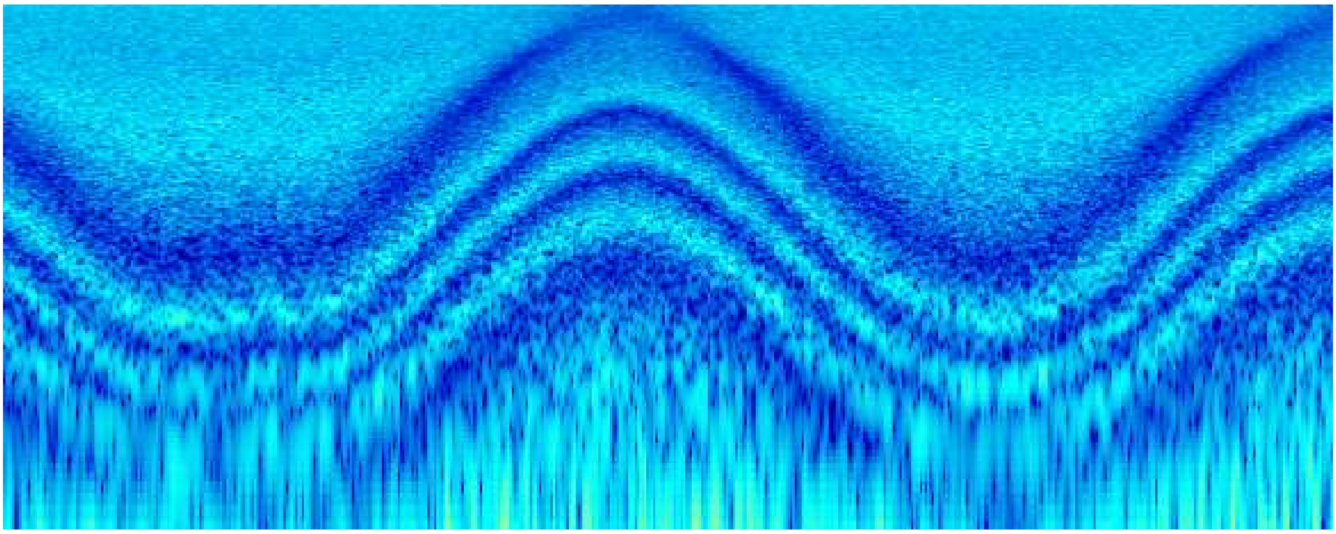
Introduction

The Phaser is a special dynamic filter. The result is a sweeping sound, which is often used on instruments with a large frequency band, like guitars or strings. This makes it typical for genres like rock or funk, where it is often modulated with a pedal, but also for giving strings a warm, relaxing character.

Function

The audio signal is split into two paths. One path remains unchanged. The other one is sent to a delay line. The delay time (the so called **phase**) is made dependent on the frequency. Therefore, an all-pass filter is applied to the signal, which **preserves** the amplitude, but determines the delay time. In the end, both paths are added.

The following picture describes how this works on white noise. Light blue signalizes that the frequency is not present at the current time, and dark blue signalizes the opposite. The dark blue peaks appear if the delay time is very short, because then, the second path almost equals the first one, which results in duplication of the signal. If the delay line is very long, then it is --- in the case of white noise --- totally at random whether the delayed signal currently duplicates the unchanged path, or whether it cancels it out to zero. This random effect results in white noise between the clear blue structures.



Phaser Types

ZynAddSubFX offers different types of phasers:

- Analog and "normal" phasers. Analog phasers are more complicated. They sound punchier, while normal phasers sound more fluently. However, analog filters usually need more filter stages to reach a characteristic sound.
- Sine and triangle filters. Note that an analog triangle filter with many poles is a barber pole filter and can be used to generate Shepard Tones, i.e. tones that seem to increase or decrease with time, but do not really.

- The LFO function can be squared. This converts the triangle wave into a hyper sine wave. The sine squared is simply a faster sine wave.
- TODO: Barber is deactivated, since PLFOtype is only 0 or 1?

Description

For the normal phaser, first, the LFO is generated:

- There are 4 controls (**Freq**,**Rnd**,**LFO type**,**St.df**) that define the LFO.
- **Phase** and **Depth** are applied afterwards in the usual way (TODO: I don't understand the code here for the normal phase...). For the analog phaser, **Phase** is not implemented, yet.
    - If **hyp** is being set, then the LFO function is being squared.

Next, the input is being used.

- **Analog** decides whether the phaser is analog or "normal".
- First, **Pan** applies panning to the original input in every loop.
- Next, barber pole phasing is being applied (Analog only).
- **Fb** applies feedback. The last sound buffer element is (after phasing) multiplied by this value and then added to the current one. For normal filter, the value is added before, for analog after the first phasing stage.
- Now, **Stages** phasing stages are being applied. **dist** sets the distortion for when applying the phasing stages. This has only effect for analog phasers.
- The feedback is taken now.
- In the end, **Substract** inverts the signal, multiplying it by -1.

Alienwah

Introduction

The AlienWah effect is a special, dynamic formant filter (TODO: is this true?). Paul Nasca named it AlienWah because it sounded "a bit like wahwah, but more strange". The result of the filter is a sound varying between the vocals "Ahhhhh" (or "Uhhhhh") and "Eeeeee".

Function

The way that the filter moves between the two vocals is mainly described by an LFO. A bit simplified, Paul Nasca has stated the formula (for $i^2=-1; R<1$) as

$fb=R*(\cos(\alpha)+i*\sin(\alpha))$

$y_n=y_{n-delay}*R*(\cos(\alpha)+i*\sin(\alpha))+x_n*(1-R)$.

The input x_n has the real part of the samples from the wave file and the imaginary part is zero. The output of this effect is the real part of y_n. α is the phase.

Description

- **Pan**
- The following 5 controls (**Freq**,**Rnd**,**LFO type**,**St.df**, **Dpth**) define the LFO.
  - **Fb**
  - **Delay** If this value is low, the sound is turned more into a "wah-wah"-effect.
  - **Phase** See α in the above formula. This lets you set where the vocal is between "Ahhhhh" and "Eeeeee".
  - **L/R** applies crossover in the end of every stage. This is currently not implemented for the Analog Phaser.

Persistence

As with most applications ZynAddSubFX allows for one to ave your work and reload it.

Saving it all

One of the simplest ways to save your work is to save the entire session. This can be done through the File menu and will result in the creation of an .xmz file. Once created, this file will hold the settings for all settings within that session, such as microtonal tunings, all patches, system effects, insertion effects, etc…

Saving Parts

In many cases saving everything is not what is desired. Saving a patch later on is one such example.

Patches

In order to save a patch, one can either save it from the instruments menu or through the bank window.

With the instrument menu, one can just save the file to any given location with the .xiz extension.

With the banks menu, one can assign a patch to a given slot with a bank. This instrument will remain here for future use until it is deleted. To see the physical location of the .xiz file, one should check the File→Settings→Bank_Root_Dirs window to see the paths for banks.

NOTE	You need to have write permissions to add instruments to the bank.

Presets

Have a favorite setting for an envelope, a difficult to reproduce oscillator? Then presets are for you. Presets allow for one to save the settings for any of the components which support copy/paste operations. This is done with preset files (.xpz), which get stored in the folders indicated by File→Settings→Preset_Root_Dirs.

Summary

Extension Summary

```
xmz Everything
xiz Instrument
xsz Scale Settings
xpz Presets
```

Appendex A: MIDI Defaults

Default MIDI Connections

```
001 - Modulation Wheel
007 - Volume
010 - Pan
011 - Expression
064 - Sustain
065 - Portamento Enable
071 - Filter Q
074 - Filter Cutoff
075 - Bandwidth(*)
076 - Modulation Amplitude(*)
077 - Resonance Center Frequency(*)
078 - Resonance Bandwidth(*)
120 - All Sounds Off
121 - Reset All Controllers
123 - All Notes Off
```

The entries with `(*)` are not within the General Midi specification

Appendix B: Building ZynAddSubFX

Introduction to CMake

> Note: This section is mostly copied from the OpenSceneGraph wiki, at: http://www.openscenegraph.org/projects/osg/wiki/Build/CMake

ZynAddSubFX uses CMake as its unified build system. CMake is able to read simple build scripts from the source tree and create from this a platform-specific build system. This build system can be in the form of VisualStudio project files, Unix Makefiles or XCode project files. CMake is able to automatically locate external dependencies, and allows you to toggle on/off module compilation and configure various build options.

The use of a unified build system has allowed to avoid build breakages that were common in the previous build method of maintaining three separate build targets for VisualStudio, Unix "make" and XCode. It also reduces the maintenance burden for core developers and contributors. Taken together usage of CMake should result in better consistency and more stable builds across all platforms for end users and a greater productivity in development of new versions. Hopefully with greater consistency of builds across platforms it will be easier for developers to use the development version of ZynAddSubFX and help contribute to its testing and refinement, leading to a high-quality code base.

Quick start guide

For the impatient ones, here is a quick guide on how to immediately build ZynAddSubFX from source.

> Note: This assumes that you already have a copy of the source.

```
#enter the source directory
cd zynaddsubfx

#make a directory for an out-of-source build
mkdir build
cd build

#generate a cmake build project here from the cmake root, which is
#found in the directory below the current one
cmake ..

#OPTIONAL: Adjust compile variables in the Cache file:
ccmake .

#And finally, build as usual using make
make
```

Appendix C: Getting ZynAddSubFX

Usually there are several methods to obtain a copy of ZynAddSubFX.

SourceForge

  http://sourceforge.net/projects/zynaddsubfx/files/

Distribution

  apt/yum/others

Git

  git clone git://git.code.sf.net/p/zynaddsubfx/code zynaddsubfx

Introduction to Git

For those who want to live on the bleeding edge or who want to assist with making sure that the next release has fewer bugs, you will want to get acquainted with git. Git is used to manage the source code for this project and can be used to quickly and easily get an up-to-date copy of the source code.

Getting the Source Code

In order to get a copy of the ZynAddSubFX source code, all that needs to be done is:

```
git clone git://git.code.sf.net/p/zynaddsubfx/code zynaddsubfx

cd zynaddsubfx

#Download additional resources
git submodule init
git submodule update
```

You should now be in the directory of the source code.

For simple steps on building, please see Appendix B of the manual.

Checking out a branch

Lets say that development has extended into the creation of a new feature that you want to preview. For the sake of this guide, lets assume that the name of the branch that the feature is on is foo.

```
#checkout the foo branch from sourceforge
git checkout --track -b foo origin/foo

#lets checkout the primary branch again
git checkout master

#hop back to the other branch
git checkout foo
----------------------------------------

Now one should be able to change branches and go into the build directory (as
described in Appendix B) and recompile ZynAddSubFX.

NOTE: When using branches other than the master be aware that stability may
      suffer
```