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The Inspiration for FlexiBeatzII

Even before Microsoft shifted its very first copy of Windows 2000, an intrepid Dutchman by the name of Bram Bos released a little freeware drum pattern creator called Hammerhead Rhythm Station. It went on to become popular as it was easy and fun to operate, invited experimentation, and could be used to quickly generate some decent and very useable breaks and loops.

Online forums filled up with users lavishing praise on the program, but also requesting and eagerly anticipating additional features that would've made the program somewhat more flexible.

Fast forward a decade and the desktop music scene has changed considerably. But Hammerhead is still out there on countless machines, and the requested enhancements to the original freeware version never came. Why? Because the Hammerhead buzz caught the attention of a well-known music software house, Bram ended up signing a contract with them to turn Hammerhead into the commercial 'B.Box' product, and so evolution and support of Hammerhead had to cease.

These days are plenty of other feature-rich rhythm-making options out there, but I felt there was still room for a kind of 'enhanced' Hammerhead.

So I took note of the most-requested features and user-comments, thumbed through the user manuals of assorted drum machines and samplers I've still got stacked in a cupboard for additional ideas, and am pleased to announce FlexiBeatzII. After having worked with it for a while, I find it does inspire my creativity and help with my workflow, and hope it will do the same for you.

System Requirements

VB6 Runtime

Bram used Delphi to create Hammerhead, I've used VB6 for FlexiBeatzII, so it requires the Visual Basic 6 runtime library to be installed on your system - if you successfully run other VB6 applications on your system, then your system already has this library installed (as most systems do). Otherwise, you might get 'A required .DLL file was not found', 'Runtime error' or 'Error: the file msvbvm60.dll could not be found' message when you try to run FlexiBeatzII. If you do get such an error, simply search the Merrie Internette for VBRun60.exe - it's a self-extracting executable that installs all files required by VB6 applications.

DirectSound

Also, FlexibeatzII makes use of DirectSound. This is a Microsoft API which provides a direct interface between audio applications and the soundcard drivers on Windows XP and earlier Operating Systems. FlexibeatzII has been built to utilize Directsound Version 8 for passing audio data to the soundcard, mixing sounds, recording etc. The requisite DLL is already part of Windows XP, but for Windows 2000 (the only other OS I've tested FlexibeatzII on), one needs to register DX8VB.DLL first (by doing Start>Run>regsvr32 dx8vb.dll). This version can co-exist with any earlier DX versions (such as DX7) already part of Windows 2000. DX9 was the last officially supported version for Windows XP and 2000. Windows Vista and Windows 7 both have it installed. Windows Vista originally shipped with DX10, and Windows 7 includes DX11.

Besides the use of DirectSound for the purposes mentioned above, all FlexibeatzII features have been programmed from scratch, including the DSP algorithms for the synthesis, effects, EQ etc. No use was made of the preset effects which come with DirectSound, or sound libraries etc

FlexibeatzII vs Hammerhead Features Comparison

Since many people are already familiar with Hammerhead Rhythm Station, I thought it might be useful to present a table of features of each, to give a quick snapshot how FlexibeatzII 'extends' the concept of that fine little program:

Feature #	Feature	Hammerhead	FlexiBeatzII
1	User Instrument sounds	Conversion from standard wav files necessary using a separate tool. 6 user sounds possible per channel	Plays any wav file. No conversion necessary. Can choose from unlimited sounds per channel
2	Number of channels	6	10
3	Note resolution	16ths	32nds
4	Max number of measures	8. Designed to create a short pattern for use in a midi/audio sequencer or sampler	999. Can be used for either short pattern creation, or complete compositions
5	Play from a specified measure to a specified measure	No	Yes
6	Record as wav from a specified measure to a specified measure	No	Yes
7	Pan per channel	No	Yes
8	'Freehand' Pitch adjust per channel	No	Yes
9	'Lock to Semitones' Pitch adjust per channel	No	Yes
10	Swing adjust	Only depth of the effect can be adjusted, not the notes to be delayed/advanced	Both the depth of the effect, and the notes to be delayed/advanced can be adjusted
11	Note and Measure counters	No	Yes
12	Mute and Solo per channel	Only Mute per channel	Yes
13	Lock channel BPMs to master Tempo	No	Yes
14	Accent per Instrument per Note	No	Yes
15	Midi Clock Sync to other Sequencers	No	Yes
16	Dynamically alter sample start point of each Instrument in each channel	No	Yes
17	Dynamically specify which sounds can cut-off (mute) which other sounds (if	No. Closed Hi Hats are 'hard wired' to mute Open Hi Hats	Yes

	they are still playing)		
18	Hard or Soft Mute	No	Yes
19	Randomize Instruments in channels and Notes per Instrument	No	Yes
20	‘Lo Fi’ recording options	No	Yes
21	‘Auto Stretch’ Instrument samples to fit a specified fraction of a measure, or multiple measures	No	Yes
22	Auto-generate arpeggiator patterns	No	Yes
23	Ability to synthesize Instrument sounds from scratch in each channel	No	Yes
24	Ability to process and add effects to the Instrument sounds in each channel	-Reverse -Distortion -Level	-Reverse -3 types of Distortion -Level -Pan -Master LP Filter -Freely draw and apply a pitch envelope to a sound -Freely draw and apply a LP, HP or BP filter envelope to a sound -Freely draw and apply a volume envelope to a sound -Chorus -Delay -Compression -Change amplitude of positive and negative halves of the waveform independently -Reverb -Lo-Fi (bit and sampling rate reduction) -3-band Parametric EQ (featuring LP, HP, BP, Lo Shelf, Hi Shelf, Notch, Peak) -Stutter effect -Chop up waveform and randomly reassemble -Apply waveshaping by manipulating individual harmonics -Change pitch without changing duration -Change duration without changing pitch -Change both pitch and duration -Vocoder -Ring Modulator -AutoWah
25	Recording of knob/slider	No	Any changes to the

	movements while the sequence is playing		Level of the Instrument sound in each channel are recorded
26	Loop Slicing	No	Yes. Ability to automatically isolate the ‘hits’ in a drum loop and export each slice as a separate wav file
27	Graphical Time and Frequency display of each Instrument sound	No	Yes

Whatever happened to FlexibeatzI?

I never ended up releasing the original version of the application, as it lacked a lot of the editing, synthesis and effects features of FlexibeatzII, and in practice I found it was a bit too limiting as a self-contained application for music creation.

Overview

Initially, drum machines were nothing more than simple sequencers of a small palette of static sounds. The sounds were either pre-loaded samples or electronically generated through subtractive synthesis. Sometimes both techniques were used eg the hi-hats and cymbals may have been short samples, but the kicks and snares electronically generated. These machines offered very limited control over the individual sounds - changing the level, pan and tuning were about the only options available to the user.

Eventually, powerful sequencer-sampler-synth workstations emerged with the ability for users to load their own samples, twist and shape them to an extraordinary degree, assign them to key ranges and sequence them, such as the E-MU Emulator Ultra and Kurzweil K2XX series of the 90’s.

However, these instruments were complex beasts and much of the tactility and immediacy that came with the simple pattern sequencing interfaces of the original drum machines (such as the X0X interface which allowed quick construction of loops and patterns), was gone. The X0X style interface simply means the sequence is represented by a row of buttons – typically 16 or 32 – which can be pressed or depressed (or if the buttons are virtual and represented by checkboxes in the case of FlexiBeatzII – ticked or unticked) to reflect a ‘hit’ of an instrument or sound of some kind. It is a very intuitive and straightforward method of sequencing made famous by various vintage Roland bass and drum machines such as the 303, 606, 808 and 909 (hence the X0X reference).

The beauty of the X0X interface has always been real-time interaction in activating/deactivating each step of the measure while the loop is playing (thereby potentially creating infinite variations of the pattern on the fly) and the great visual reference provided about the on/off state of each step. Many people find this interface to be the best way to get nice beats and interesting loops happening, fast.

Since the emergence of those all-in-one-do-everything workstations, there have been attempts to bring the two worlds of X0X sequencing and versatile synthesis together through instruments such as Korg's Electribe series, Roland's Grooveboxes, the Radikal Spectralis, Elektron MachineDrum etc. Also, we now have in the mix great software like Ableton, Reason, Battery, Guru and a host of others.

Despite this, I have felt for some time that there was still room for an accessible software-based X0X sequencer that offers the user a powerful synthesis toolset for each sound, with the synthesis power being able to be applied dynamically while the loop is playing (but at the same time not compromising the ease of use of the sequencer). I know I am not alone in the opinion that such interactivity inspires creativity that cannot be achieved by using a separate sound editor/synth for creating the samples, and then loading them into a pattern sequencing application. Such an approach does not enable the spontaneity that comes from making changes to the sounds on the fly and getting immediate feedback for the changes.

For example, whilst the original inspiration for FlexibeatzII – Hammerhead - is a great little X0X drum sequencer application, it is extremely limited in the per-sound synthesis department. On the other hand, Stomper is still a good sound synthesizer for percussion sounds and sound effects, but it has no sequencer - you have to export the sounds you create with it, and then import them into something like Hammerhead.

Conceptually, what FlexiBeatzII does is combine both capabilities into the one application, but the resultant whole is far more than the sum of the parts.

You can open up a versatile synthesis panel for each channel independently via which you can draw in volume, frequency and filter envelopes for your sample, or waveshape your sample by manipulating individual harmonics, and combine the result with sounds created using subtractive synthesis on a variety of standard and specialized waveforms. This 'combining' needn't be limited to simply adding the processed sample with the processed waveforms - you can also FM one with the other.

You may decide your sound shouldn't even contain any aspect of the sample at all and that it should be purely synthesized and shaped from scratch.

You can put the sonic result through an arsenal of effects independently for each channel eg reverse, chorus, delay, reverb, various types of distortion, stutter, filter, compression, bit reduction, sample rate reduction and separate level settings for the positive and negative parts of the waveform.

A 3-band Parametric Equalizer is also provided to enable you to finely EQ the sound in each channel.

You can vocode, ring-modulate and auto-wah. You can dynamically vary the start and end points of the sound while it's playing (besides the standard adjustment of pitch, volume and pan). You can have a sample chopped into segments of varying lengths and have the segments pasted together again randomly. You can have a percussion sample chopped into individual ‘hits’ and then open up these up in separate channels.

When programming a sequence, you can opt to see notes to 32nd resolution instead of the standard 16, and you can copy and paste patterns around the composition.

You can have FlexibeatzII randomize sound selection and note selection in patterns. (FlexibeatzII can open up random sounds in each channel or randomize the notes set for the sound in each channel, for that extra bit of inspiration).

FlexibeatzII can simulate arpeggios by opening up the same sound in multiple channels but setting the Frequency slider of each sound, and the notes sequenced for each sound, in such a way as to sound like an arpeggio pattern when the sequencer is run.

Once you have created your pattern or song, you can record it direct to a .wav file as the sequence plays (together with any other sounds being played through the soundcard at the same time). In the case of a pattern, you can open up its .wav file in your sampler or midi & audio sequencer. If you save the pattern into the FlexibeatzII data folder, you can of course select the pattern again as a sound in a channel.

You also have the option of exporting just the FlexibeatzII sequence as a .wav file without actually playing it and recording it.

If it is a complete song you've created, the only thing you may need to do is master it in your favourite audio editor (typically all I do in an external editor is to trim, add fade in/out, normalize add perhaps a bit of dynamics compression) and burn the .wav file to CD, or compress it to mp3 and post on the Merrie Internette.

The .wav file can be of a song you have made completely in FlexiBeatzII, or FlexiBeatzII midi'd up with your other gear.

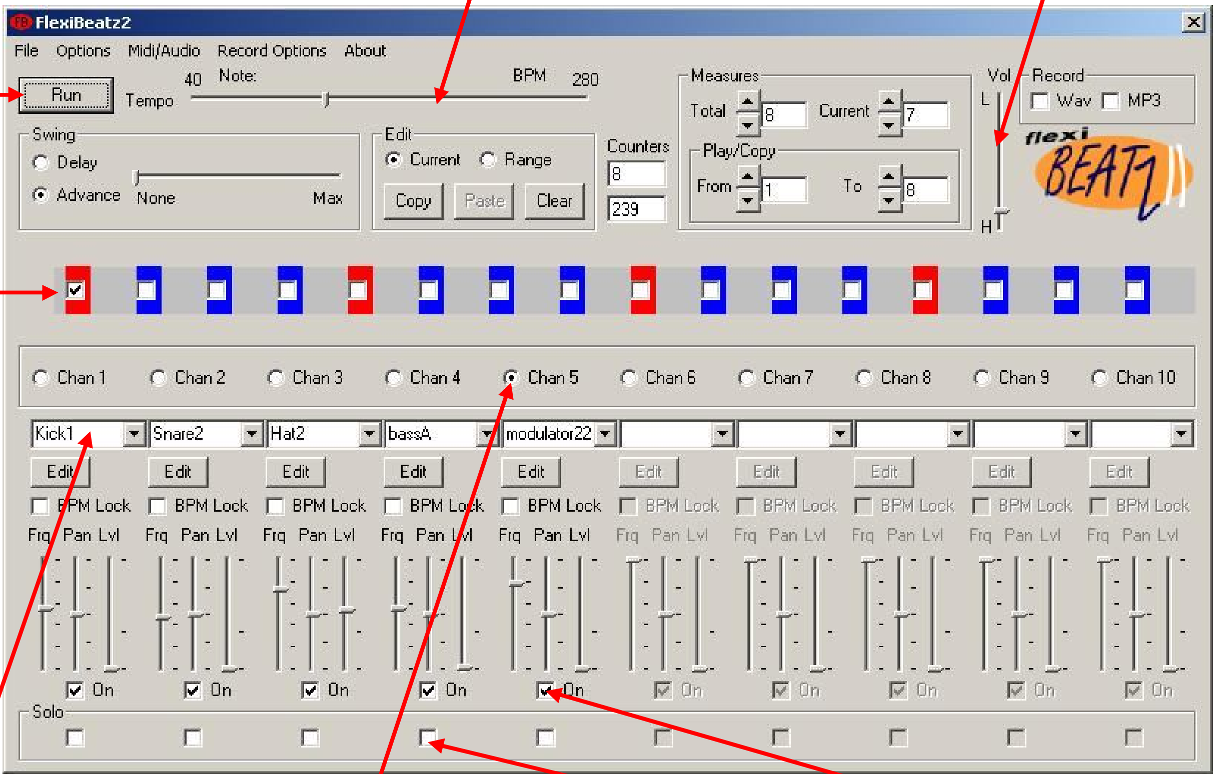
6. Adjust tempo to desired level

5. Adjust volume to desired level

Quick Start

4. Press Run to hear your sequence looping in the ‘From’ and ‘To’ Measures range

3. Sequence Notes in the grid for the selected channel



1. Press the dropdowns to see all wav files in the data folder. Select a sound

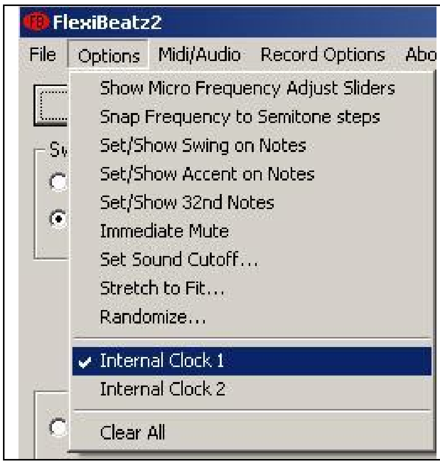
2. Select a channel

8. Tick and untick this ‘Solo’ checkbox to un-mute and mute ALL sounds (respectively), OTHER than the sound in this channel

7. Tick and untick this ‘On’ checkbox to un-mute and mute the sound (respectively) in THIS channel only

Simply follow steps 1 through 6 annotated on the above screenshot of the main screen, for a first experience of FlexibeatzII sequencing your sounds

About FlexibeatzII's Internal Timer



When FlexibeatzII is not being driven by an external Midi clock, the application uses one of two internal timing components – either 'Internal Clock 1' or 'Internal Clock 2' selected under the Options menu. This is the component used to sequence patterns and songs when you press the 'Run' button.

You probably won't need to try a different timing component than the one selected by default.

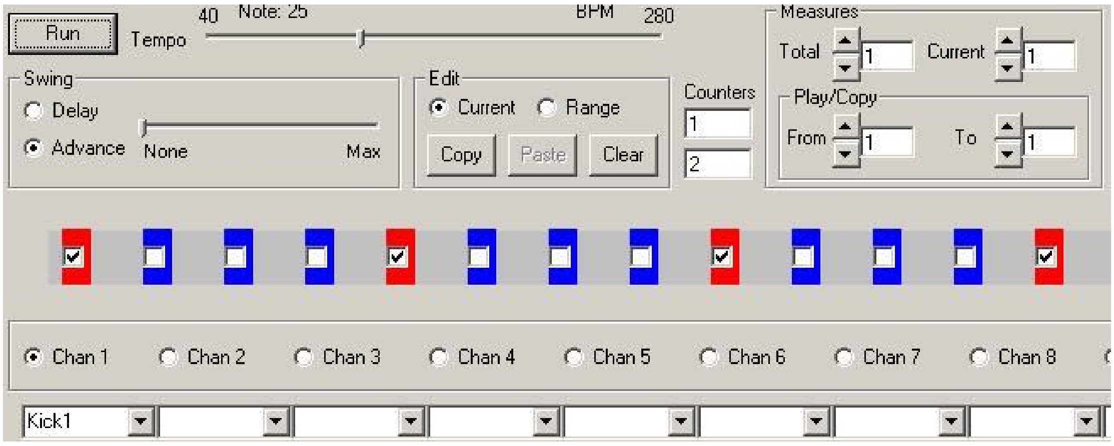
I implemented the Internal Clock 1 component when I experienced some 'sloppy' timing on one of my computers with the 'Internal Clock 2' component

Pattern/Song Creation

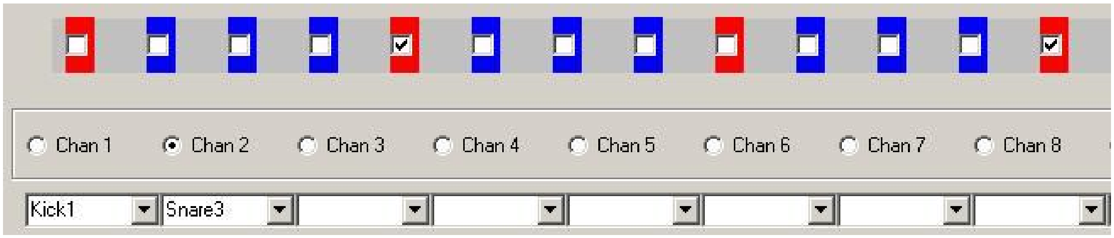
Select the radio button of any channel (from 1 to 10). Then select the sound for the channel from the drop-down list of sounds (which picks up all the wav files in the data directory). The sounds can be anything - you can for example set 10 kicks or 10 different sounds such as instrument notes, stabs, effects, vocal samples or loops. It's up to you. As long as you put your sounds in the data folder, they are selectable in any channel. Set your Lvl slider and Pan slider for the channel, then set the notes for the sound in each Measure.

To illustrate, let's create a basic drum pattern. Select Channel 1. You do this by clicking the radio button saying Channel 1. Choose your instrument eg select the sound "kick 1". You can find it in the drop-down menu underneath the Channel 1 button. All notes you enter in channel 1 will now be these kick drums.

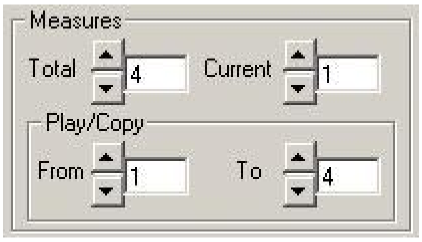
We'll edit this pattern on the fly, which means that you let the pattern play while entering notes. You can then hear immediately what you do. Do so by clicking on the 'Run' button. Look at the row of 16 checkboxes. This represents one measure. One measure consists of four beats, which in turn is made up of 4 ticks. Click on the first of every 4 checkboxes in the row. These are the red coloured ones.



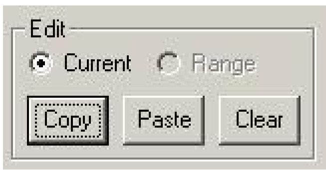
You should hear a basic "four-to-the-floor" bass drum rhythm repeat over and over again. We'll add snares on every other beat. Select the Channel 2 radio button. You should see that none of the notes checkboxes are now ticked. Choose one of the snares as your channel 2 instrument and click on the second and fourth of the red notes checkboxes. Now your beat should go "boom - tschuk - boom - tschuk..."



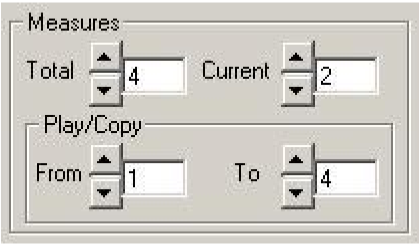
Nice, but it might feel a bit monotonous after a while. So, we'll place a fill-in at every fourth measure. Go to the 'Measures' group of controls and change the "Total" measures to 4.



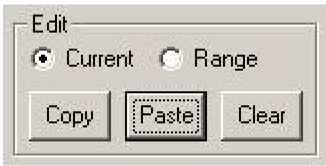
You should hear the beat play one measure, then stop for three measures before it starts over. To copy your existing pattern into the three empty measures, go to the group of controls titled 'Measures', scroll the 'Current' vertical scrollbar to 1. The notes should be showing whatever has been set in Measure 1, for the highlighted channel.
Next go to the group of controls titled 'Edit' and ensure the 'Current' radio button is selected. Then press the 'Copy' button. This copies everything in the current Measure to the clipboard.



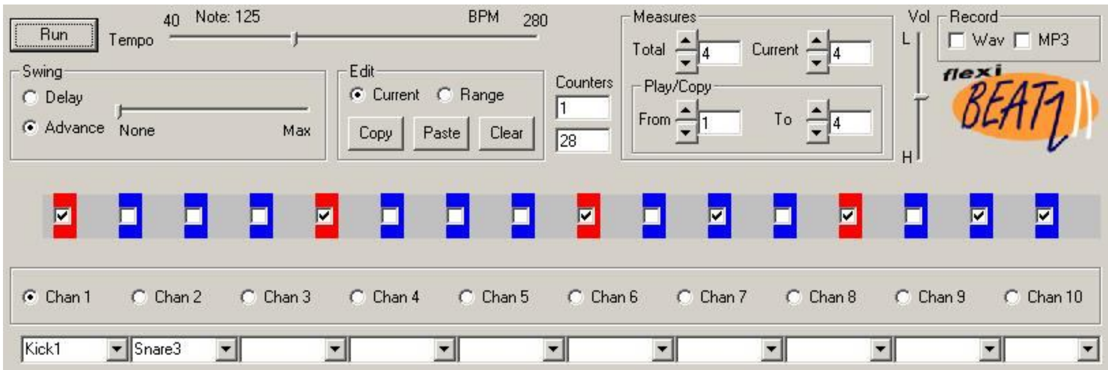
Then set the 'Current' scrollbar in the 'Measures' group of controls to 2.



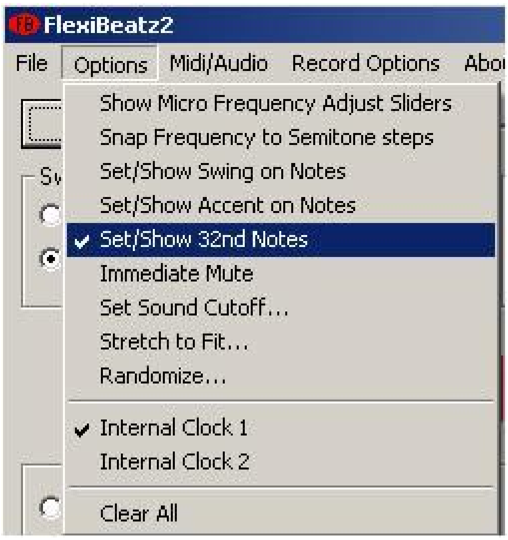
Now you can paste the pattern you copied from measure 1 into measure 2 by choosing Paste in the Edit group of controls.



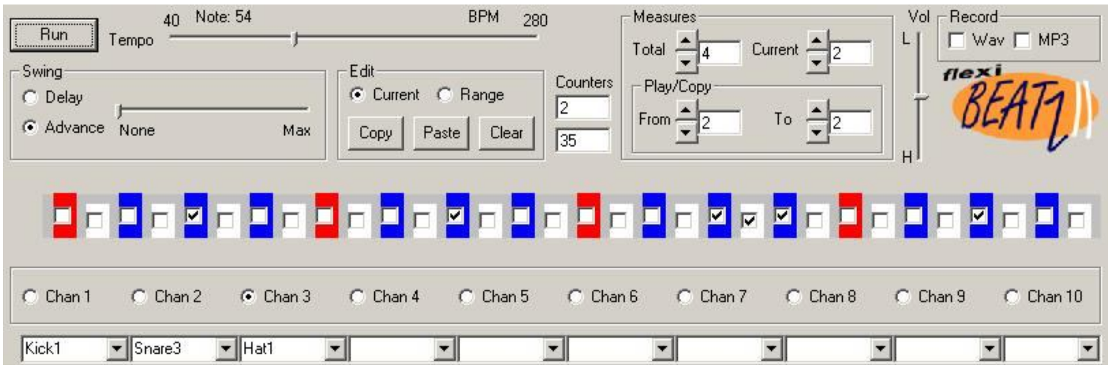
Repeat for measure 3 and 4. When you're done, the loop should be playing continuously again. We will now set current measure is set to 4, and add a funky bass drum variation to this measure. Select "Channel 1" (the bass drum) again. You should see all red buttons ticked. Then click on note checkboxes 11, 15 and 16.



Hear the difference on every fourth bar? Now we will add a hihat sequence in measure 2, including a small 32nd note pattern. We set the 'Current' scrollbar in the 'Measures' group of controls to 2. We load a hihat sound in Channel 3. Now, to add some notes in Measure 2 for the hihat sound, we select the Channel 3 radio button. We want to see 32nd notes in the Measure, so we select:



We also want the sequence to indefinitely loop Measure 2 after we fill the note checkboxes for the hihat sound and press ‘Run’, instead of looping through Measure 1 to 4. This is just so we can evaluate the pattern we have created in Measure 2 and immediately hear any changes we make to the pattern. In order to constrain the looping to Measure 2, we go to the 'Measures' group of controls and change and ‘From’ value to 2, and the ‘To’ value to 2. We now fill the note checkboxes with the hihat sound. So, we have this as the result:

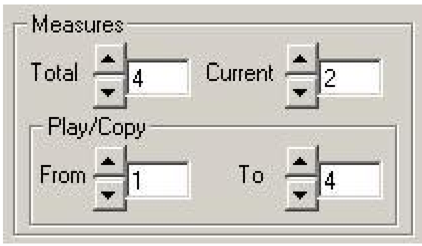


When we press Run, we now hear the drum pattern complete with hihat (and 32nd note flourish on the hihat) on Measure 2 repeated over and over.

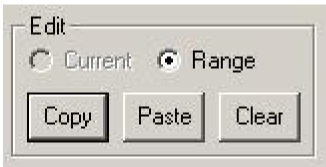
To hear the complete Measure 1 to Measure 4 pattern, we again go to the 'Measures' group of controls and change and ‘From’ value to 1, and the ‘To’ value to 4.

Let’s say we now want to copy and paste this 4 Measure pattern onto Measure 5, so that we end up with an 8 Measure pattern, consisting of two identical 4 Measure patterns one after the other. To achieve this, we do as follows:

In the 'Measures' group of controls, ensure ‘From’ value is 1, and the ‘To’ value is 4. This is the range we wish to copy.

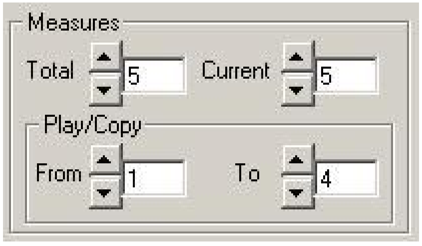


Next go to the 'Edit' group of controls and ensure the 'Range' radio button is selected (because this time we don’t want to copy the contents of the current Measure, we want to copy the Range showing in ‘From’ and ‘To’). Then press the 'Copy' button.

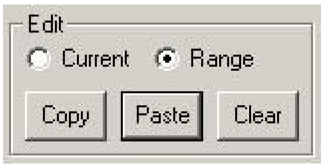


This copies everything in the Range to the clipboard.

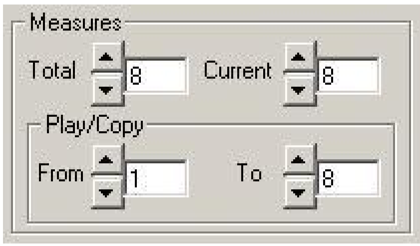
In the ‘Measures’ group of controls, you won’t be able to set the Current value as 5, until you set the Total to 5. So, do this adjustment till Current reads as 5. This is because we wish to copy the Range in the clipboard to Measure 5.



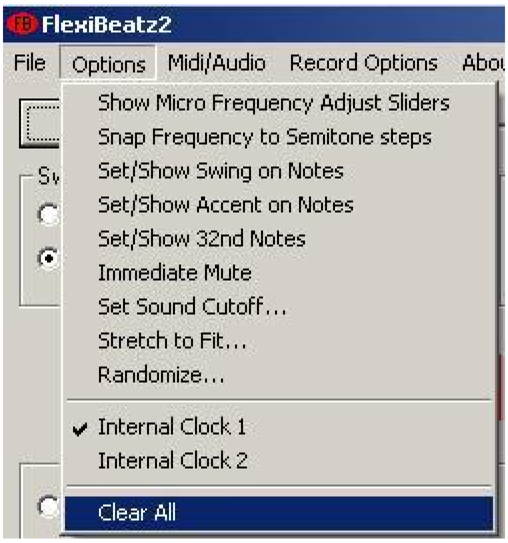
What has previously been copied is 4 Measures long, so when we paste it at Measure 5, we expect to see the total length of our sequence as 8 Measures. Press Paste in the Edit group of controls:



We can now see from our ‘Measures’ group of controls that our pattern has been pasted:



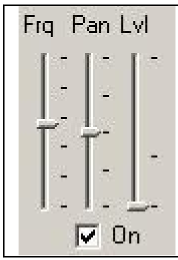
Press Run to loop through your new sequence now spanning Measures 1 through 8
Now you can start adding new instruments and play around with the notes settings to add some complexity to this basic drum beat. Experimentation is the key.
You can clear all your settings by selection Options>Clear All, allowing you to build up a fresh pattern.



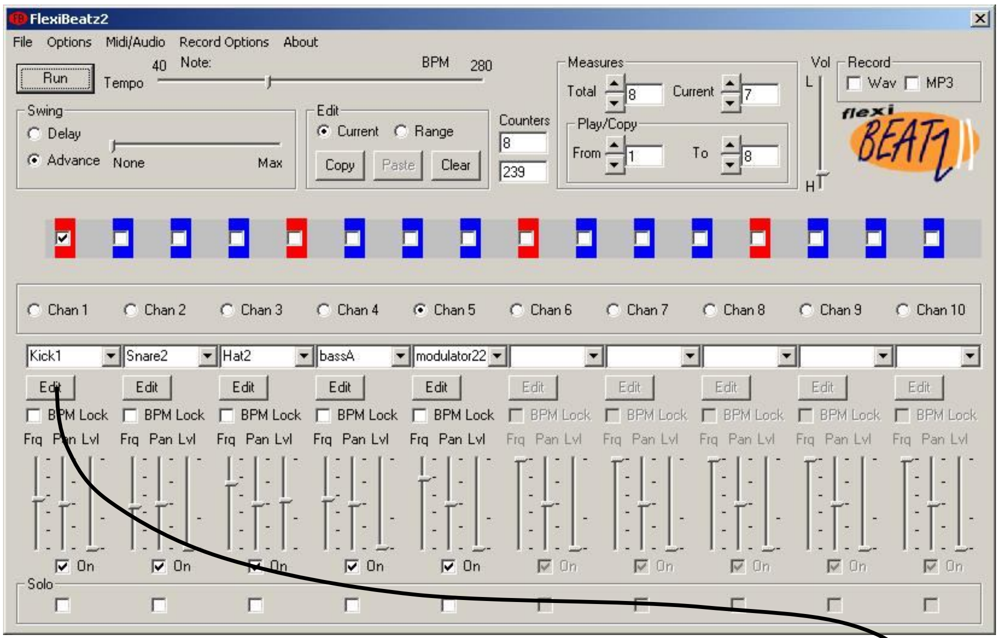
Editing/Synthesizing Sounds Overview

When you load a sample into a channel on the main screen of the application, the ‘Edit’ button for that channel is enabled, but this doesn’t mean you have to do any sound editing. It is entirely possible to create a complete song with just ‘as-is’ loaded samples.

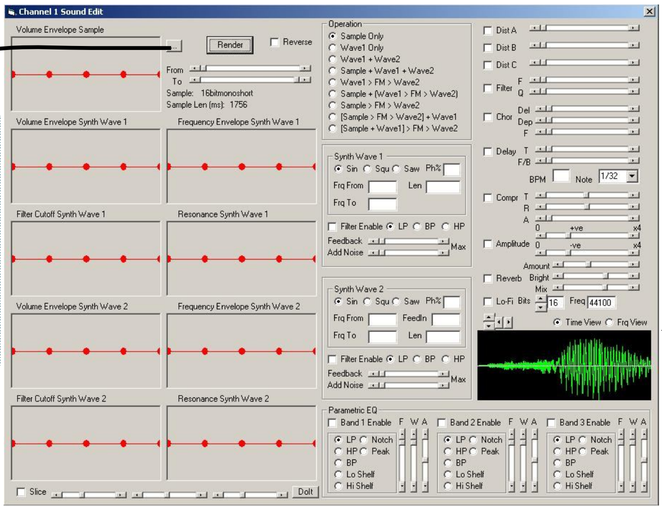
Note: If you have loaded a stereo sample, the only way to retain the stereo image of the sample is to actually use it as-is. FlexibeatzII converts the sound to mono when you use FlexibeatzII’s many DSP tools to process it

	Once the sample is loaded, changing the Vol, Pan and Lvl sliders do not perform any DSP editing of the sample; they simply affect how the sample is played-back. There are two panels available for Editing for each channel: the first panel that opens up when you press ‘Edit’ is the ‘Sound Edit’ panel. The second panel opens up when you press the button next to the ‘Render’ button on the ‘Sound Edit’ panel: called the ‘Advanced’ panel.

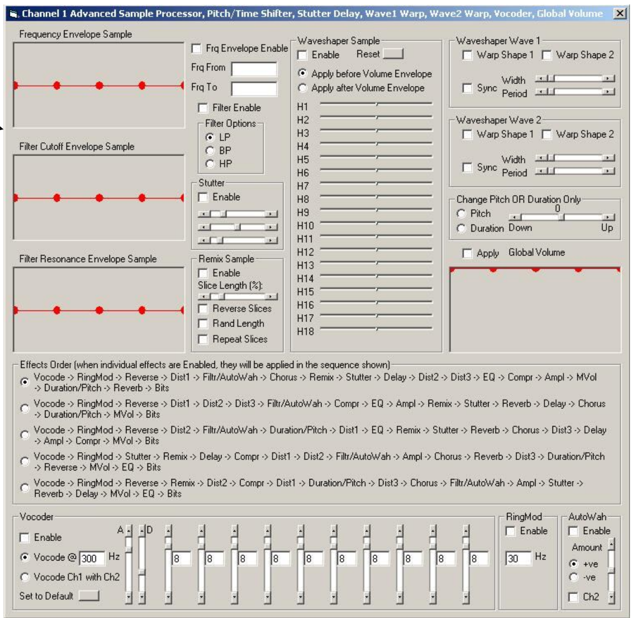
The following screenshots show navigation to the Sound Edit’ panel and ‘Advanced’ panel from the main screen of the application:



Loading a sound into a Channel enables the Edit button. Pressing the Edit button launches the Sound Edit panel

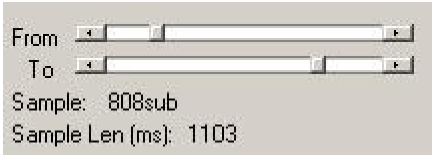


Pressing small button next to 'Render' launches the 'Advanced' panel



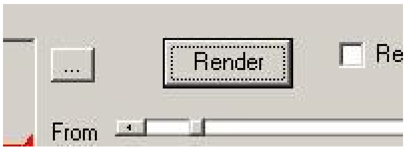
Between the ‘Sound Edit’ and ‘Advanced’ panels there is a lot you can do to the sample – from subtle processing to changing it beyond recognition. In fact, once in the Sound Edit and Advanced panels, you can even synthesize a new sound from scratch and use it instead of your sample, or combine the synthesized sound with the sample in various ways

Note: Changing the position of the From and To sliders on the Sound Edit panel does not perform any DSP editing of the sample; it simply affects what section of your sound is played-back.



What you do to the sound in any one channel is completely independent of all other channels, (unless you choose an effect which processes sounds in multiple channels together – such as the Vocoder and AutoWah effect, where the sound in Channel 1 can be set to affect the sound in Channel 2).

You commit your edits performed in the Sound Edit and Advanced panels, by clicking on the ‘Render’ button on the Sound Edit panel.

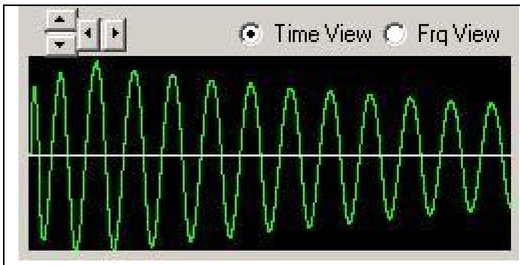


Depending on the nature of your edits, the processing of the edits can take anywhere from less than a second to many seconds. For example, applying any of the following to a sample many seconds long, could take some time:

- Frq Envelope Enable
- Waveshaper Sample Enable
- Change Pitch OR Duration Only
- Stutter Enable

The shorter the sample, the less processing time taken.

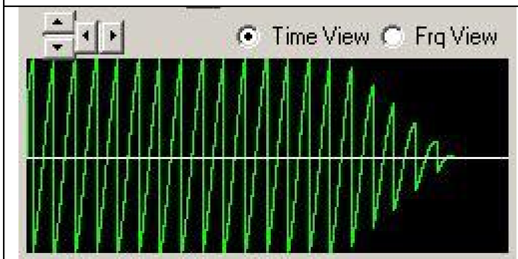
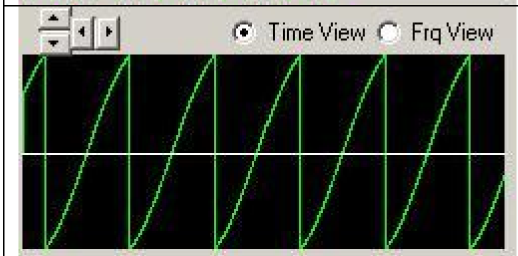
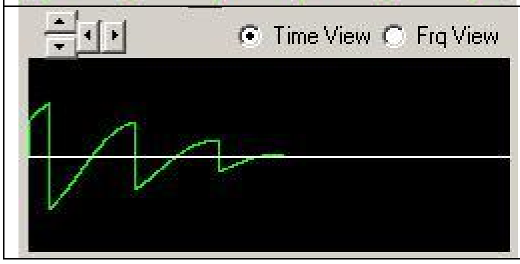
Once the edit has been committed, you will see a Time/Frequency chart of the waveform appear in the Sound Edit panel. You can hear the result and play the sound by running the sequencer, choosing the channel and entering a note in one of the Note checkboxes.

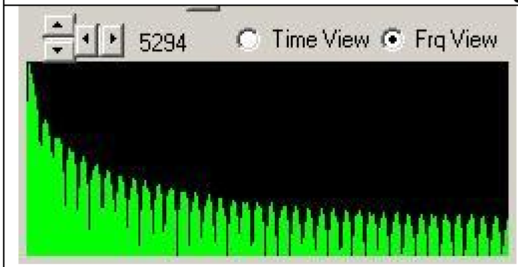
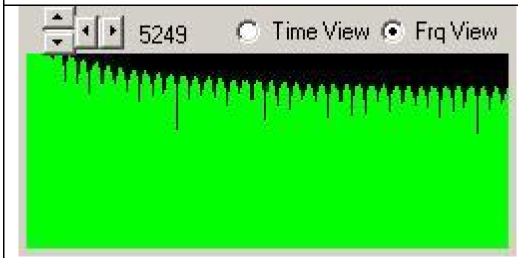
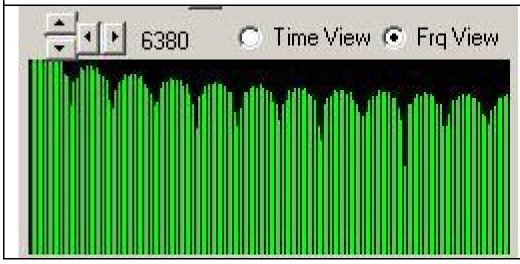


Here, a note is checked so that the ‘808sub’ sound can be heard as the sequencer plays, and the Time representation of the 808sub waveform is shown

As mentioned previously, any 8 bit Stereo or 16 bit Stereo .wav file loaded into a channel, gets converted into a 16 bit Mono sound once ‘Render’ is pressed in the Sound Edit panel, and all edits are performed on this converted sound

Graphical Time and Frequency Display Navigation

TIME VIEW	
	This is a display of a whole example waveform, as seen when the ‘up’ arrow of the vertical scrollbar is pressed a few times to ‘zoom out’
	Pressing the ‘down’ arrow on the vertical scrollbar magnifies the display along the horizontal axis, effectively ‘zooming into’ the waveform. This screenshot shows a section of the waveform after having pressed the ‘down’ arrow a few times
	Pressing the left and right arrows on the horizontal scrollbar, moves you along the waveform. In this screenshot, the right arrow has been pressed a few times to move the view to the end of the waveform

FREQUENCY VIEW	
	Pressing the ‘up’ arrow on the vertical scrollbar results in a maximum contrast display between the ‘highest highs’ and ‘lowest lows’ of the spectrum. Pressing the ‘right’ arrow on the horizontal scrollbar results in maximum frequency range display (in this screenshot the chart spans 0 Hz to 20kHz. 5294Hz is being displayed only because that is where the mouse pointer was at the time. Clicking on any position of the waveform displays the Frequency at that point)
	Pressing the ‘down’ arrow on the vertical scrollbar shifts the view down to the ‘lowest lows’. This is useful when you want to see spectrum detail without some extraordinarily high peaks skewing the display. (5294Hz is being shown only because that is where the mouse pointer was at the time. Clicking on any position of the waveform displays the Frequency at that point)
	Pressing the left arrow on the horizontal scrollbar, results in display of a progressively narrower frequency range. In this screenshot, the left arrow has been pressed a few times, such that the display spans 0Hz to 6380Hz. The 6380 value appeared when the extreme right edge of the waveform display was clicked)

Editing/Synthesizing Sounds Detail

 Screenshot of Sample and Wave Operation Options	Sample Only: Processes only the sample you have loaded Wave1 Only: Processes only what you have set for Synth Wave 1 Wave1 + Wave2: Adds together what you have set for Synth Wave 1, and Synth Wave 2 Sample + Wave1 + Wave2: Adds together your processed Sample, with Synth Wave 1 and Synth Wave 2. Note: If you just wish to add the Sample with Synth Wave 1 only, leave settings for Synth Wave 2 blank Wave1>FM>Wave2 : Frequency modulates Synth Wave 2 by Synth Wave 1 Sample + [Wave1>FM>Wave2]: Adds Sample with Synth Wave 2 frequency modulated by Synth Wave 1 Sample>FM>Wave2: Frequency modulates Synth Wave 2 by the Sample [Sample>FM>Wave2]+Wave1: Adds Synth Wave 1 with Synth Wave 2 frequency modulated by the Sample [Sample+Wave1]>FM>Wave2: Frequency modulates Synth Wave 2 by the Sample and Synth Wave 1 added together

Note: You must load a sample to not only process the sample itself, but also to synthesize a sound from scratch (i.e. perform a ‘wave’ operation), even though it doesn’t use the sample itself.

The length of the wave file you are able to synthesize will be limited by the length of your sample eg even if you put Len = 2000 for your wave but sample length = 200, your synthesized wave sound will only be 200 ms long

Sample Only Operations Overview

If you choose any operation which has the word ‘Sample’ in it, there are certain processes which apply ONLY to the Sample, namely:

On Sound Edit panel: Volume Envelope Sample

On Advanced panel: Frq Envelope Enable, Filter Enable, Remix Sample, Waveshaper Sample, Vocoder, RingMod, AutoWah

Note:

- When ‘Frq Envelope Enable’ and ‘Waveshaper Sample Enable’ are ticked, the ‘Frequency Envelope Sample’ curve only gets applied if ‘apply after Volume Envelope’ radio button is pressed
- When ‘Filter Enable’ and ‘Waveshaper Sample Enable’ are ticked, the filter curves (i.e. ‘Filter Cutoff Envelope Sample’ and ‘Filter Resonance Envelope Sample’) only get applied if ‘apply after Volume Envelope’ radio button is pressed
- When ‘Filter Enable’ and ‘Waveshaper Sample Enable’ and ‘Remix Sample Enable’ are ticked and ‘apply before Volume Envelope’ radio button is pressed, the waveshaped sample is remixed without filter applied
- When ‘Filter Enable’ and ‘Waveshaper Sample Enable’ and ‘Remix Sample Enable’ are pressed and ‘apply after Volume Envelope’ radio button is pressed, the waveshaped and filtered sample is remixed
- When ‘Frq Envelope Enable’ and ‘Waveshaper Sample Enable’ and ‘Remix Sample Enable’ are ticked and ‘apply after Volume Envelope’ radio button is pressed, the waveshaped and frequency-enveloped sample is remixed
- When ‘Frq Envelope Enable’ and ‘Filter Enable’ and ‘Waveshaper Sample Enable’ and ‘Remix Sample Enable’ are ticked and ‘apply after Volume Envelope’ radio button is pressed, the waveshaped and frequency-enveloped and filtered sample is remixed

Wave 1/Wave 2 Only Operations Overview

When any of the wave1 or wave2 operations without the word ‘Sample’ in them is pressed, then these are the processes available:

On Sound Edit panel: All control groups with ‘Synth Wave 1’ or ‘Synth Wave 2’ in their captions

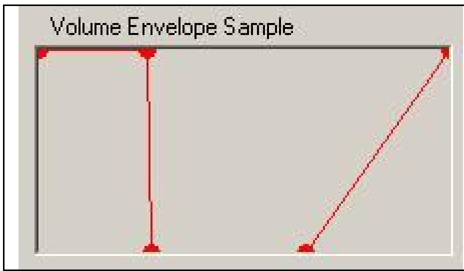
On Advanced panel: Waveshaper Wave 1, Waveshaper Wave 2

Operations Common to both Sample and Wave 1/Wave 2 Overview

Stutter, Change Pitch or Duration Only, Global volume, Effects, EQ

Sample Only Operations Detail

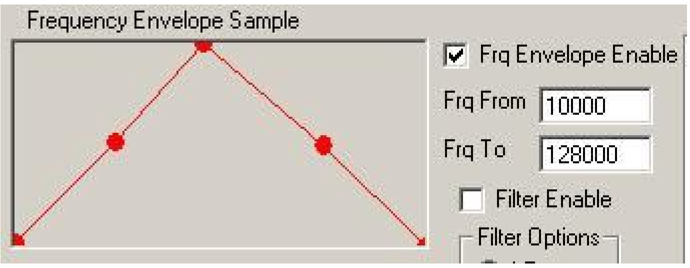
Volume Envelope Sample



Whatever volume curve you draw here will be imposed on the sample sound. In the adjacent example, a section of the sample has been silenced. The sound plays at maximum volume for a while then abruptly stops for a while. Finally, it sharply ramps up to maximum volume again.

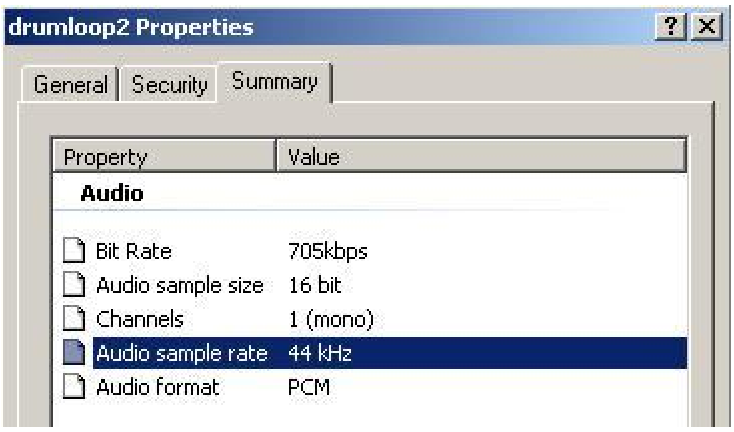
Note: All Envelope windowboxes in FlexibeatzII allow you to create an envelope shape simply by 'dragging and dropping' the red nodes anywhere in the window area

Frequency Envelope Sample



The Frequency Envelope Sample feature speeds up playback (and increases the pitch) or slows down playback (and decreases the pitch) of the sample according to the curve you draw between the Frq From and Frq To values.

The Frq From and Frq To values represent the rate of playback of the sample. In the screenshot, the sample has a sampling rate of 44.1kHz. We ascertain this by right clicking on the .wav sound file, choosing Properties>Summary, and then reading the value of 'Audio Sample Rate':



The Frq From value has been entered as 10000 in this example: more than 4 times less than the sampling rate of the sample. At this value, the sample will play back much more slowly and the pitch will be much lower. The Frq To value has been entered as 128000: more than 3 times the sampling rate of the sample. At this value, the sample will play back much quicker and the pitch will be much higher.

The drawn ‘ramp up-ramp down’ curve in the Frequency Envelope Sample windowbox results in the sample playback rate going from 4 times less to 3 times more for half the duration of playback, and then going from 3 times more to 4 times less for the remaining duration of playback.

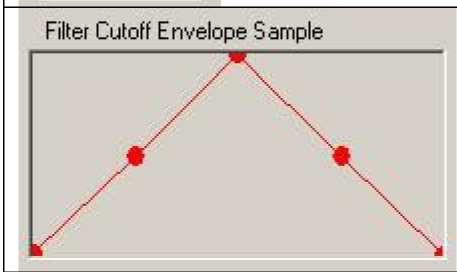
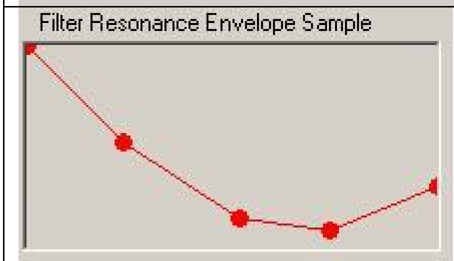
Filter Envelope Sample

<table>
<tr><td>
</td><td rowspan="3">On the Advanced Panel, you can apply a Filter envelope to your sample.

Tick Filter Enable, choose from one of three filter types:

• LP (Low Pass)
• BP (Band Pass)
• HP (High Pass)

Then draw your Envelopes for Filter Cutoff and Filter Resonance.

In this screenshot, a very pronounced filter-sweep effect on a vocal sample is obtained by applying Band Pass filtering on it and setting the Cutoff and Resonance of the filter over the duration of the sample, as shown</td></tr>
<tr><td>
</td></tr>
<tr><td>
</td></tr>
<tr><td colspan="2">Note: Resonance has the effect of emphasising harmonics at the Cutoff frequency of the filter. Frequencies around the Cutoff tend to 'ring' with high Resonance settings, which makes high Resonance settings useful for creating bell, gong etc type sounds. If you 'sweep' the filter Cutoff frequency by drawing an increasing/decreasing curve in the Filter Cutoff Envelope Sample windowbox, but set a constantly high level in the Filter Resonance Envelope Sample windowbox (say a straight line close to the top boundary of the windowbox), various overtones are 'picked out' of the sound and amplified as the resonant peak sweeps over them.</td></tr>
</table>

Remix Sample

Chops the sample into many pieces and randomly re-arranges the pieces

Slice Length: determines the length of each piece. The further the slider is to the right, the longer each piece will be (and therefore fewer the number of pieces)

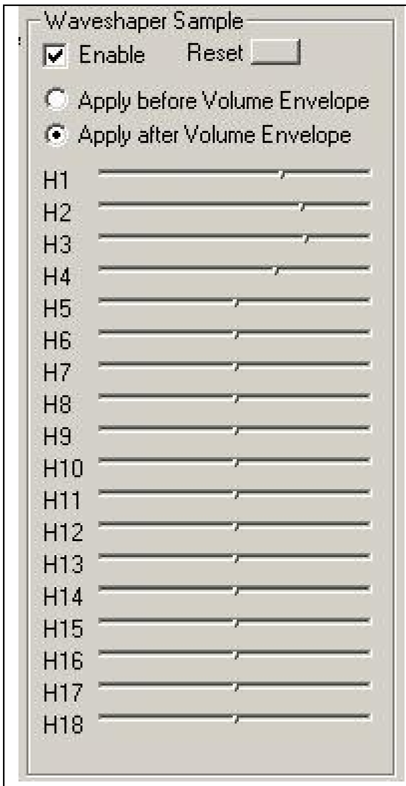
Reverse Slices: Ticking this checkbox results in some pieces randomly being reversed

Rand Length: Ticking this checkbox results in each piece being of random length (irrespective of the setting of the Slice Length slider)

Repeat Slices: Ticking this checkbox results in some pieces randomly being repeated

Each time you 'Render', the remixed result will be different; even when you open a saved FB file

Chebychev Waveshaper



Chebyshev polynomials have a special characteristic in that if you apply a Chebychev polynomial of order n to a pure sine wave of amplitude 1 (where the positive peak of the sine wave is +1 and the negative peak is -1), the resulting output signal will contain only the nth harmonic.

Here, the first slider H1 adds a Chebychev polynomial of order 1 to the output signal, the second slider H2 adds a Chebychev polynomial of order 2 to the output signal, and so on.

This means:

- If you slide only H1 to the right, and leave all the other sliders zero'd, then a sine wave of amplitude 1 with frequency F at the input results in a sine wave with frequency F x 1 = F at the output
- If you slide only H2 to the right, and leave all the other sliders zero'd, then a sine wave of amplitude 1 with frequency F at the input results in a sine wave with frequency F x 2 = 2F at the output
- If you slide both H1 and H2 to the right, you would expect to see only F and 2F in the output.

In order to validate the behaviour of the Chebychev waveshaper described above, we first load a sine wave sample and ensure it has amplitude of 1 as follows:

In this example, the sine wave has a frequency of 200 Hz.

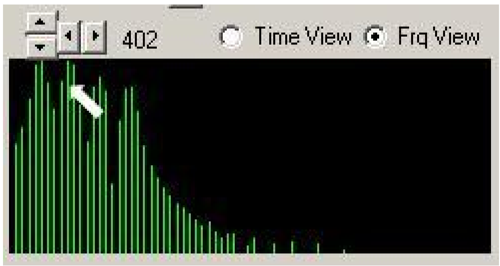
Now we shift the first four sliders of the Chebychev waveshaper H1 through H4 to the right as shown in the screenshot. When we apply these settings on our sine wave of amplitude 1 and frequency 200 Hz, we would expect to see only the following frequencies in the output:

1x200 = 200 Hz
2x200 = 400 Hz

3x200 = 600 Hz
4x200 = 800 Hz.

The ‘strength’ of each frequency in the output would be determined by how far each of the sliders is moved.

With this expectation, we Render then look at the Frequency spectrum:

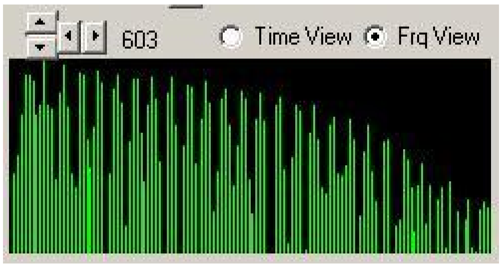


We indeed see 4 equally spaced peaks – the cursor is on one of the peaks and the screenshot shows the Frequency of that peak to be 400 Hz. The one before it is 200 Hz and the two after it are 600 Hz and 800 Hz respectively.

So, here we have a neat mechanism for ‘dialling in’ the exact harmonic content we want the output to contain.

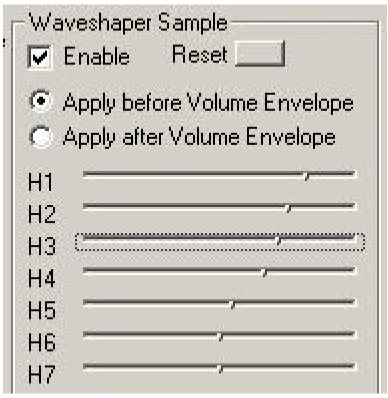
Here’s another example, where we set the harmonic content to make the sine sample sound more like a saw wave:

First, we need to know what the harmonic content of a saw looks like. To do this, we get FlexibeatzII to synthesize a saw wave and look at its frequency graph:

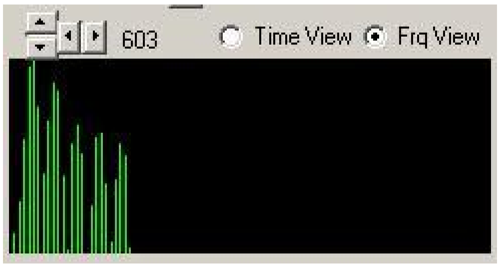


If we call each of the harmonics a partial, the first partial on the left is the fundamental and determines the pitch of the waveform. The other partials are frequency multiples of the fundamental i.e. overtones. The fundamental is the loudest sinewave of all the harmonics. The two key observations here are
1. Every frequency multiple is present in the output
2. As the overtones get higher in pitch, their amplitudes decrease.

So, if in the Chebychev waveshaper we set the amplitudes of the first few harmonics to decrease like this:



We can start to 'assemble' a saw wave from scratch. Here's what the frequency graph looks like when the sliders are set as above on a 200Hz unity amplitude sine wave sample:

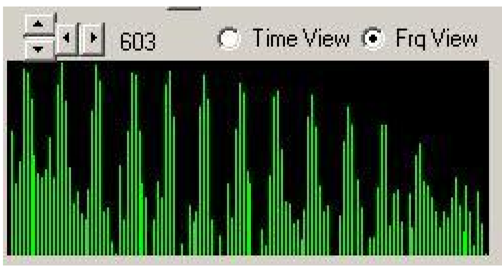


It already begins to sound a little like the saw wave synthesized by FlexibeatzII. Keep following this pattern with all the other harmonics and it will sound more and more like one.

Note: This ability to 'dial-in' particular harmonics holds true only for unity amplitude sinusoids. If the amplitude is not 1 or if the waveform is not a sine, then each of the Chebychev polynomials invoked by each of the sliders H1 through H18 will manipulate the output harmonics in complex and indeterminate ways. This of course doesn't mean the result won't be interesting or musically useful, it just means you won't be 'predictably assembling' an output waveform.

For example, if we set the Volume Envelope Sample of our sine wave sample to roll-off from 1 to 0, and then reapply the Chebychev waveshaper (ensuring the option 'apply after volume envelope' is chosen), we hear something similar to a filter sweep effect as the harmonic content changes with the level of the input

Just out of interest, let's have a look at the harmonic content of a square wave. To do this, again we get FlexibeatzII to synthesize a square wave and observe its frequency graph:



Here we see, just as with the saw wave, that as the overtones get higher in pitch their amplitudes decrease, but this time every frequency multiple is not present in the output – only the odd-numbered partials are present. So, we have 200Hz (fundamental) but not 400Hz. We have 600Hz, but not 800Hz and so on.

Vocoder

The vocoder is available only on the 'Advanced' panel of the first channel

A vocoder requires one sound sample to be a Modulator and another a Carrier. Using bandpass filters, the modulator sample and carrier sample are separated into multiple frequency bands. The amplitude from an envelope follower on each band of the modulator is used to control the level of the corresponding band on the carrier. The resulting carrier bands are mixed together to the output signal.

Typically a human voice is used as a modulator and an instrument as the carrier. It makes the instrument speak or sing in a robotic-sounding way.

But really you can use anything as the modulator – such as a rhythm loop. Best results are achieved with a string, pad or brass sound as the carrier. It is not necessary for the carrier sound to be playing a constant note either – try a melody or chord(s). The only real requirement is that the carrier sound be rich in harmonics and have a constant dynamic (try compressing the carrier sound if the dynamic variations are too great)

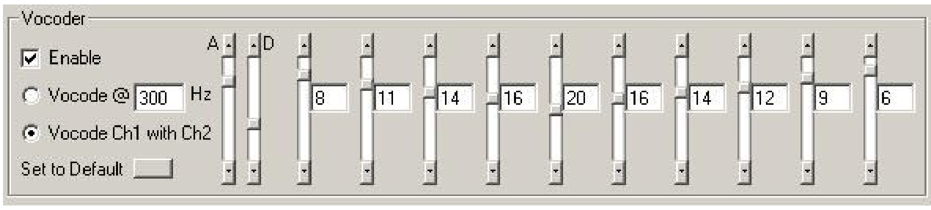
Whatever sample you use in channel 1 will act as the modulator, but FlexibeatzII offers you two options for the carrier:

- A waveform high in harmonics synthesized by FlexibeatzII itself if you choose option 'Vocode @ ___ Hz'. You only need to specify the frequency of the Carrier (the default is 300 Hz)

- The sample you have in channel 2, if you choose option 'Vocode Ch1 with Ch2'. You must Render the sample in Ch2 first

A Slider: Attack of the Envelope Follower
D Slider: Decay of the Envelope Follower

Altering these sliders changes the resolution of the extracted Envelopes. Try a short-ish Attack (A slider level close to the top) and mid-ish Decay (D slider) level to begin with. The next 10 sliders change the width of the Frequency bands of the Vocoder. Try setting the sliders so they form a sort of 'smile', with the 'corners' of the smile set to about the 4 level, and the 'bottom' of the smile set to around 20.



⌂Tip: If the level of the vocoded sound is low or it lacks 'bite', try filtering out the low frequencies by setting EQ to 'HP' at around 200 Hz or more

The vocoder will only vocode with the sample loaded in channel 2, regardless of whether you have set operation in channel 2 to 'wave'

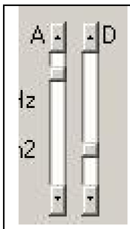
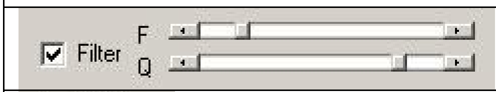
RingMod

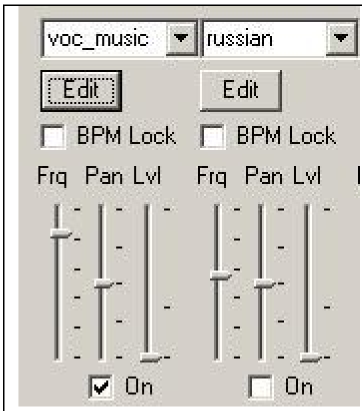
RingMod is available only on the 'Advanced' panel of the first channel

	The Ring Modulator effect is achieved by multiplying two audio signals together, which creates additional harmonics not present in the original tones; the output consists of the sum of the two input frequencies and the difference between the two input frequencies. The result is a metallic/robotic type sound. The Flexibeatz**II** Ring Modulator implementation is to allow you to multiply your audio sample with a sine wave. The default frequency of the sine wave is set to 30Hz, simply because if your audio sample is a vocal snippet and you ring modulate it with a sine wave of around 30Hz, you get a sort of Dr Who 'Dalek' effect (arguably one of the most famous uses of ring modulation)

AutoWah

AutoWah is available only on the 'Advanced' panel of the first channel

	Auto-wah can also be called an 'envelope following filter'. The auto-wah effect uses the envelope follower of the vocoder (but otherwise it has nothing to do with the vocoder) The envelope follower extracts the envelope of the sample in channel 1
	This envelope then varies the cutoff frequency of the filter
	The range of variation of the cutoff frequency will be from where the F slider is positioned, to the right edge of the slider (if the option '+ve' is chosen) or to the left edge of the slider (if the option '-ve' is chosen). Change the position of the A and D sliders of the envelope follower to vary the resolution of the extracted envelope.

	In the AutoWah effect settings, if you don't tick the 'Ch2' checkbox, you will hear the sound in channel 1 being filtered with the cutoff frequency varying according to the envelope of the sound in channel 1. If you tick the 'Ch2' checkbox, you will hear the sound in channel 2 being filtered with the cutoff frequency varying according to the envelope of the sound in channel 1. For this case, do not tick the 'On' box for the sound in channel 2, make sure you have 'Rendered' the sound in channel 2, and that the length of the sound is equal to or greater than the sound in channel 1

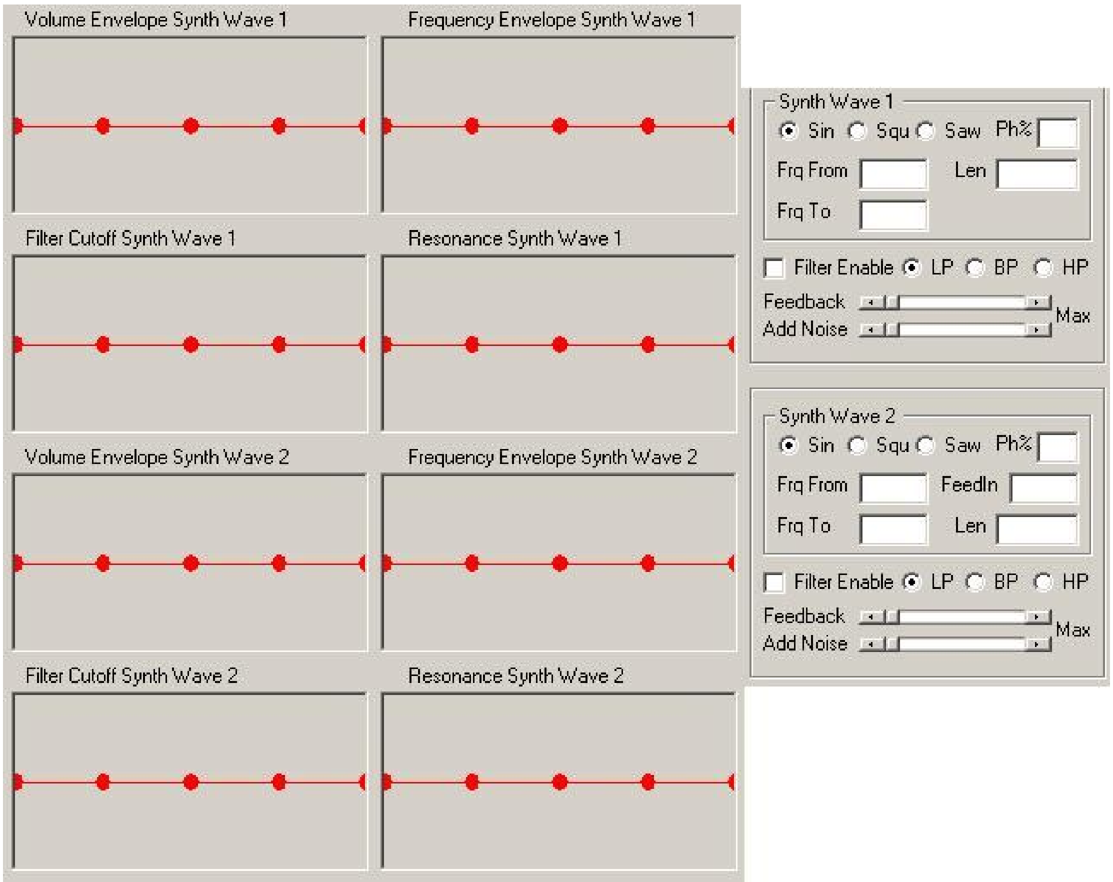
Wave 1/Wave 2 Only Operations Detail

Historically, some synthesizers featured only one oscillator, a few had many, but the most common configuration was two oscillators that could be independently set to different waveforms. These were a good balance between economy and ability because they were cheaper to produce than multi-oscillator synths yet capable of producing much richer and more interesting sounds than single oscillator synths. With these, it was possible to emulate strings, brass, woodwinds, and even vocal-like and inharmonic sounds (bells, clangs etc) and all manner of synthetic sounds.

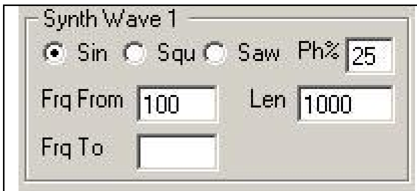
FlexibeatzII is effectively a 3-oscillator per channel synth, in that you can create sounds with two oscillators which can be set to internally-generated waveforms and you can combine these with the sample itself, which can be thought of as a third oscillator which can be set to any 'user loaded' waveform. In this section however, we will focus on the use of just the two internal oscillators and associated controls to produce sounds

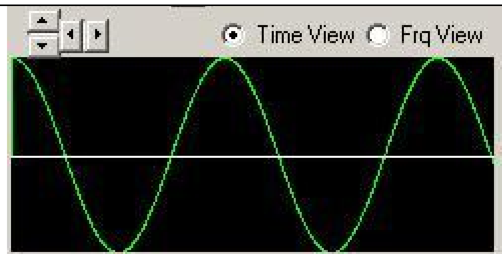
All control groups with 'Synth Wave 1' or 'Synth Wave 2' in their captions

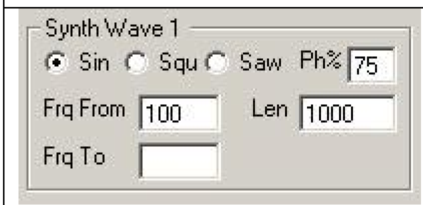
The following screenshot shows all controls applicable ONLY to Synth Wave 1 or Synth Wave 2 on the Sound Edit Panel. These are the controls with which you create sounds from scratch.

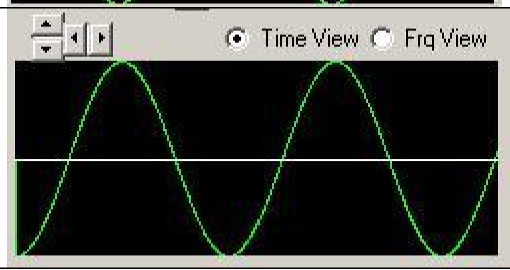


To have FlexibeatzII generate a sound, choose operation 'Wave1 Only'. In the Synth Wave 1 group of controls, select a waveform type – say Sin. Ph% denotes Phase, and here you can specify what point you want the wave to start from in its cycle. Leaving it blank or entering 0 will make the wave start at 0, entering 25 will make the wave start at its highest point (good for percussion sounds for 'maximum impact'), entering 50 will again make the wave start from 0, entering 75 will make the wave start at its lowest point

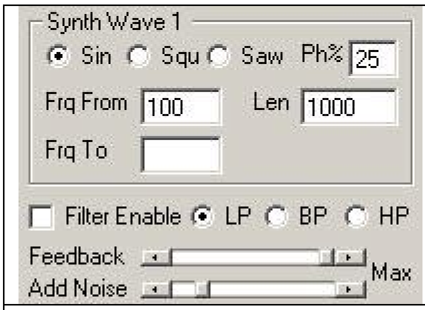


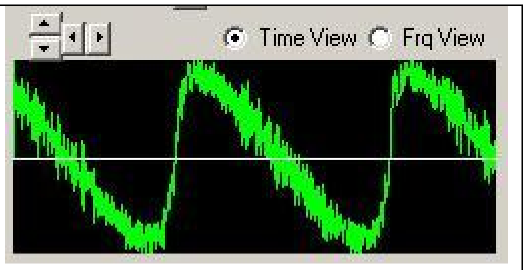






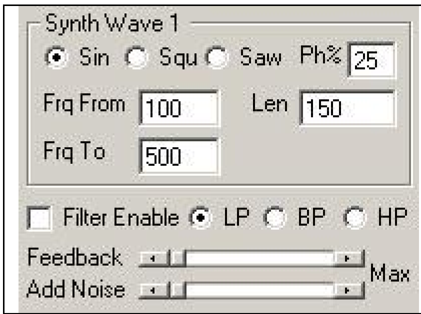
In the Frq From textbox, enter the desired frequency of the waveform in Hz. If you want the waveform to have a constant frequency, leave the Frq To textbox blank. In the Len textbox, enter the desired length of the waveform in milliseconds (entering 1000 will result in a 1 sec waveform). Pressing Render results in the waveform being generated, and you can confirm it is a sine wave by looking at Time View graph.

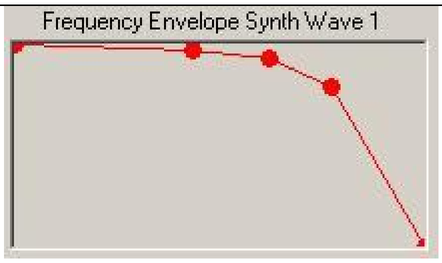


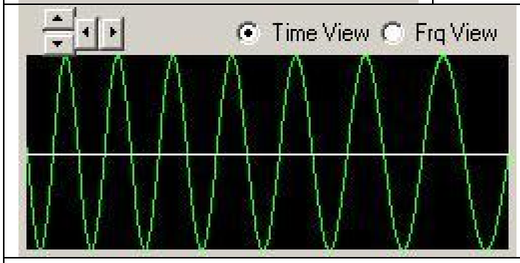


The ‘Feedback’ slider progressively skews the waveform and thus adds ‘harshness’ to the sound as it is slid to the right.

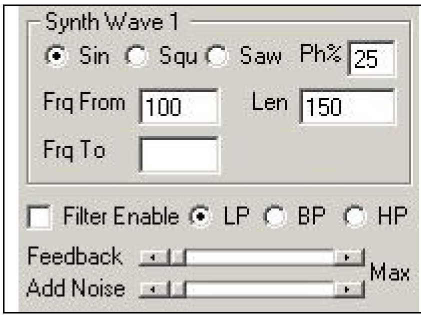
The Add Noise slider transforms the waveform increasingly to white noise as it is slid to the right.

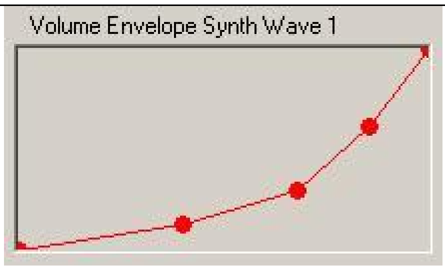


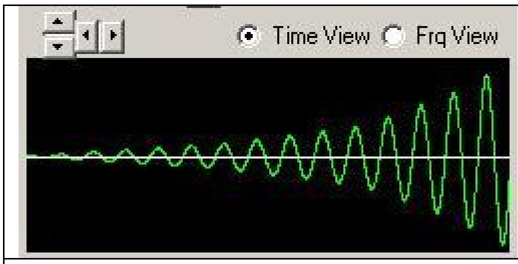




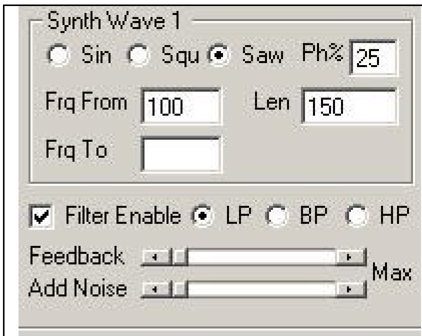
In this example, the Frq From value of 100Hz represents the bottom of the Frequency Envelope windowbox. The Frq To value of 500Hz represents the top of the Frequency Envelope windowbox. The Frequency envelope is drawn as a curve from 500Hz to 100Hz over the length of the sound. Therefore, we would expect to see the period of the sine wave increasing as we scroll left to right in the Time View of the waveform. The Time View screenshot shows this to be the case.

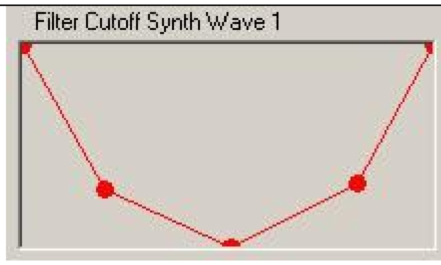


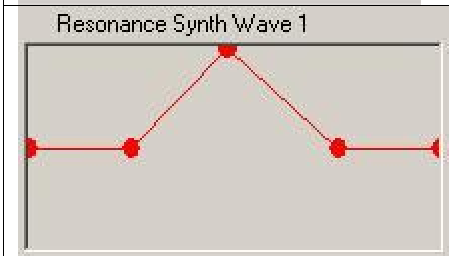


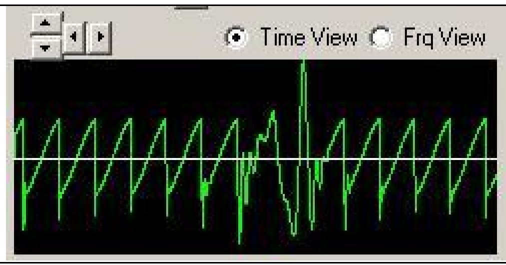


In this example, the volume envelope is drawn as a curve from 0 amplitude to maximum amplitude (1) over the length of the sound. The Time View representation of the waveform shows the level of the sine wave increasing according to the Volume Envelope curve.









In this example, the Filter Enable checkbox has been ticked, and the LP radio button has been selected, which results in the waveform getting low pass filtered according to the Filter Cutoff and Resonance Envelope curves. We see on the Time View representation of the waveform, that as expected, most of the 'filtering activity' on the waveform occurs as the filter cutoff is brought down to a low level, around the middle of the waveform

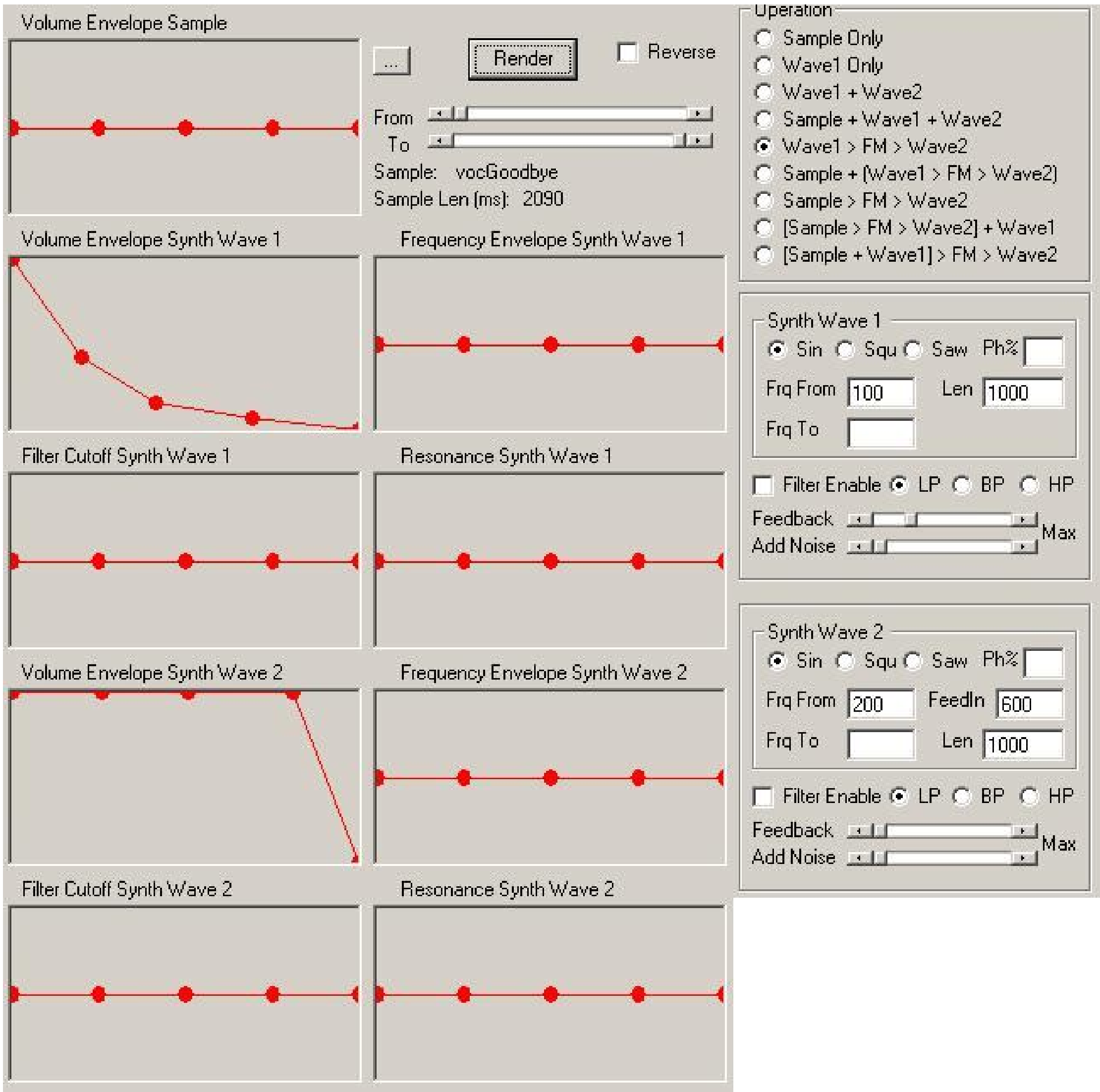
⌂Tip: It is not just the amplitude envelope which changes the amplitude of the waveform, the filter envelope ALSO does so. But whereas the amplitude envelope changes the amplitude of ALL harmonics equally, the filter envelope changes the amplitude of PARTICULAR harmonics. For example, a low-pass filter envelope will change the amplitude of the higher harmonics more dramatically, and a high-pass filter will have the opposite effect. So, removing frequencies results in an overall lower amplitude (a filtered signal isn't as loud as an unfiltered signal). To emulate some sounds (woodwinds for example), you may want to use the filter envelope first to control the amplitude of the higher harmonics and then use the amplitude envelope to control the amplitude of the lower harmonics not affected by the filter

FM Synthesis Example

FM synthesis is a form of 'distortion synthesis' where the timbre of a waveform is changed by frequency modulating it with a modulating frequency that is also in the audio range, resulting in a more complex waveform and a different-sounding tone. The frequency of a waveform is altered or distorted in accordance with the amplitude of the modulating signal. Also, as the amount of frequency modulation increases, the sound grows progressively more complex. FM is great for producing bell-like and dissonant "clangs", "twangs" and "boing" noises

In the following screenshot we are Frequency Modulating Synth Wave 2 (the Carrier) with

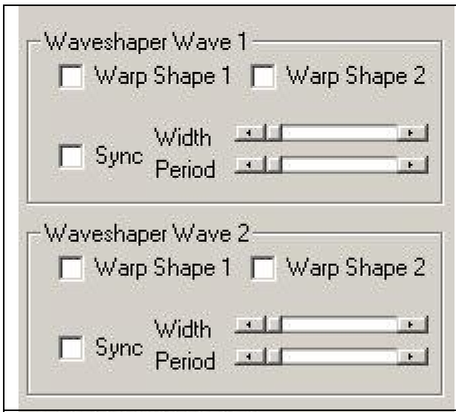
Synth Wave 1 (the Modulator), by virtue of the fact that we have chosen as the Operation Wave 1 > FM > Wave 2



Even though we have chosen the simplest waveform (sine wave), for both the Modulator and Carrier, when they are FM'd together, a waveform rich in harmonics results. This is the beauty of FM – production of many sine waves from the interaction of just two original waves. We see that the Modulator is 100 Hz and 1 sec in duration, and the Carrier is 200 Hz and also 1 sec in duration. Synth Wave 2 has an additional parameter which Synth Wave 1 does not: the 'Feed In' value. This value can be thought of as the 'strength' with which Wave 2 is modulated by Wave 1. Best results are obtained when the Feed In value is between 100 and around 3000. In this example, a Feed In value of 600 has been entered (which results in reasonably strong modulation). We also see a decaying volume envelope drawn in for Synth Wave 1. This has the effect of changing the final output substantially; the Frequency Modulated sound changes as Wave 1 decays (and it would change again if a volume envelope were drawn in for Synth Wave 2). The net result of our settings here is the production of a 'Boing' type sound.

Note: If the Frequency of one of the Waves is a multiple of the other, one gets a harmonic timbre (instrumental sound). If not, the timbre produced is inharmonic like many percussion sounds. Essentially, the Carrier is responsible for the overriding volume envelope and pitch of the final waveform, and the Modulator supplies the change in timbre.

Advanced Panel Synth Wave 1/Synth Wave 2 Operations

<table>
<tr>
<td>
</td>
<td>If you have chosen a Synth Wave 1 operation on the Sound Edit panel, by ticking Warp Shape 1 or Warp Shape 2 checkboxes under the Waveshaper Wave 1 group of controls on the Advanced panel, you are selecting a new special waveform type instead of the Sine, Square and Saw waveforms available on the Sound Edit panel. Even if one of Sine, Square or Saw is selected on the Sound Edit panel, the selection will be over-ridden by your Warp Shape 1 or Warp Shape 2 selection.

Similarly the Waveshaper Wave 2 group of controls apply if you have chosen a Synth Wave 2 operation</td>
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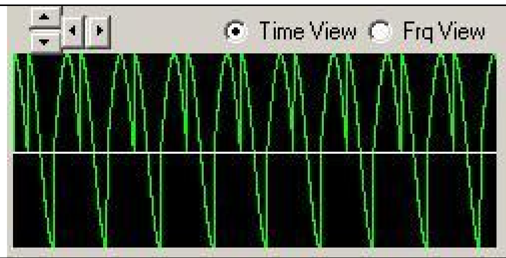
<td>
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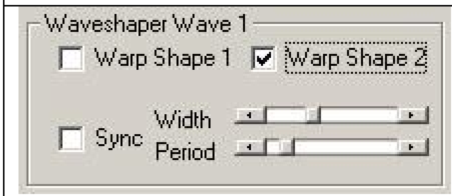
<td>
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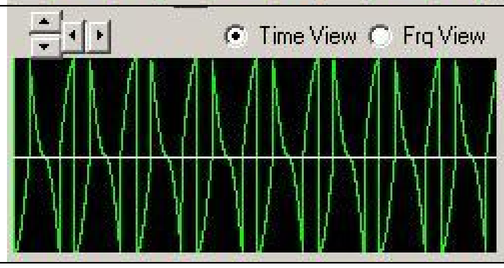
<td>
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<table>
<tr>
<td colspan="2">If you tick the Sync checkbox, then if for example you have chosen the Saw waveform for your Synth Wave 1 operation on the Sound Edit panel, the Saw wave will reset every few cycles. At what point in the cycle the reset occurs is dependent on the setting of the ‘Width’ slider. How often the reset occurs (i.e. after how many periods the reset happens) is determined by the setting of the ‘Period’ slider.

In the second set of screenshots, the ‘Width’ slider is set to a lower value than in the first set of screenshots, consequently we see the reset occurring earlier in the cycle.

Also, in the second set of screenshots the ‘Period’ slider is set to a higher value than in the first set of screenshots, consequently we see the reset occurring less frequently in the second set of screenshots</td>
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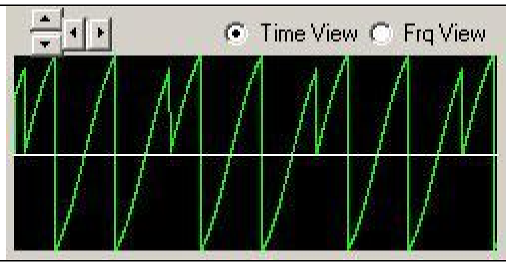
<td>
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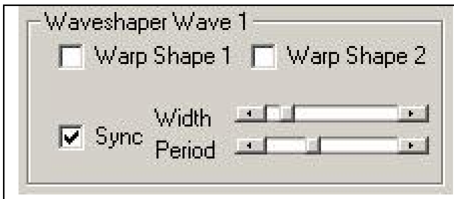
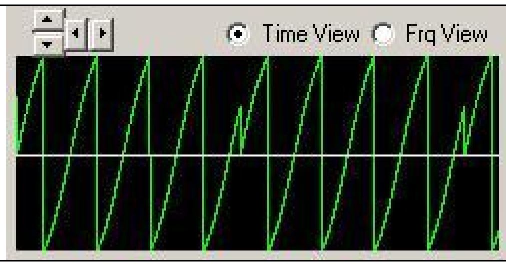
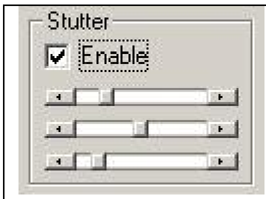
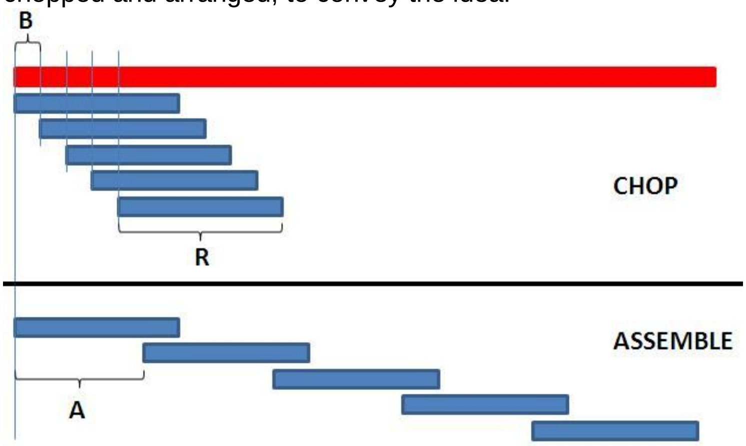
	
Note: In traditional analog synths, the sync effect was created by taking two oscillators and using one (the Master) to reset the other (the Slave) each time it starts a new cycle. The effect was most noticeable when the two oscillators were out of tune. The sync effect can create piercing and metallic sounds often used as lead sounds. FlexibeatzII does not need two oscillators to create this effect; instead, it is simulated by means of a couple of sync parameters (Width and Period).	

Operations Common to both Sample and Wave 1/Wave 2 Detail

Stutter

	Chops the sample into many overlapping pieces and plays them First Slider from top to bottom (A): determines the amount of overlap between pieces when they are arranged Second Slider from top to bottom (R): determines the size of each piece Third Slider from top to bottom (B): determines how far forward into the original sound each piece starts from

Here is diagrammatic representation of what the sliders are doing. The red bar represents the original waveform, and the small blue bars represent each piece. Only 5 pieces are shown chopped and arranged, to convey the idea:

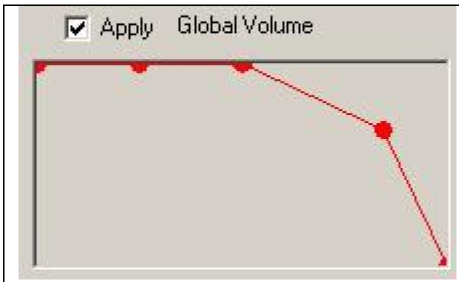


Change Pitch or Duration Only

Change Pitch	Changes the Pitch of your sound while keeping the Duration constant. If you are applying this on a sample, any leading or trailing silence should be trimmed first.

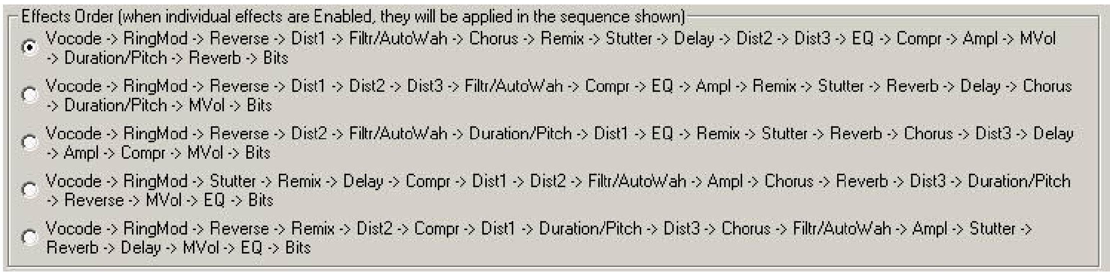
	 May not work on some samples, or at extreme settings of the slider. Move the slider 'Up' for increased pitch, and 'Down' for decreased pitch. For no effect, move the slider to '0' position (if you cannot move and drop the slider exactly onto '0', you can click the Up/Down arrows)
Change Duration	Changes the Duration of your sound while keeping the Pitch constant. If you are applying this on a sample, any leading or trailing silence should be trimmed first.  May not work on some samples, or at extreme settings of the slider. Move the slider 'Up' to stretch the sound, and 'Down' to shorten it. For no effect, move the slider to '0' position (if you cannot move and drop the slider exactly onto '0', you can click the Up/Down arrows)

Global Volume

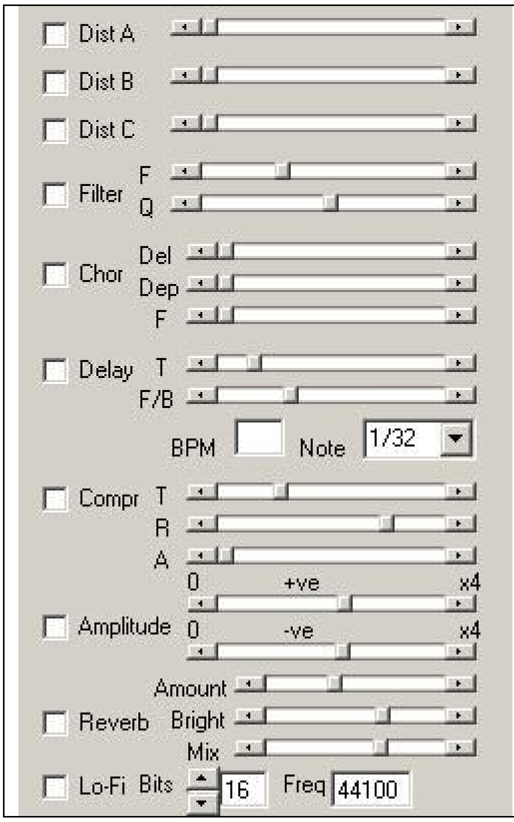
	The Global Volume (also denoted as MVol short for 'Master Volume') curve is one of the last processes to be applied, as can be seen by position of MVol in the Effects Order (see Effects section for Effects order). If you enable Global Volume, then whatever volume curve you draw will be imposed on the final sound (the resultant sound after most other effects, if selected, have been applied). In the example here, the amplitude of the sound is rolled off at the end, because the other effects applied had resulted in the sound finishing abruptly (causing clicking when the sound was sequenced)

Effects

The order in which Effects are applied to your sample, synthesized wave(s) or combination of these (based on the Operation chosen), is given by:

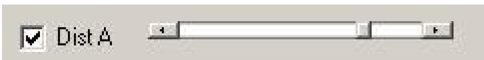
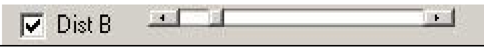


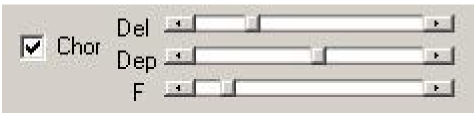
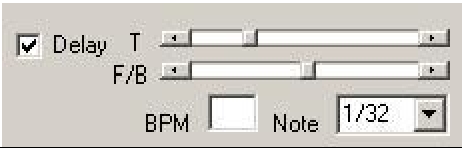
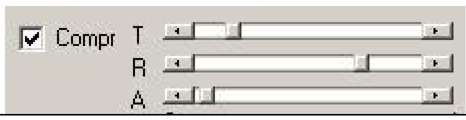
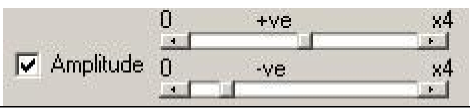
For example, if you click the first radio button (thereby opting for the first order of Effects), and you select Distortion 2, Delay and Compressor as your effects, they will be applied in the sequence Delay first, then Distortion 2 and finally Compressor

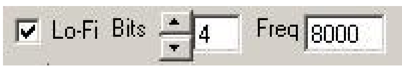


In previous sections, a number of the Effects in the Effects Order have been examined.

In this section, we will explore the remaining Effects, available for selection from the group of controls on the Sound Edit panel shown in the adjacent screenshot

Effect	Description
Distortion A	Foldback Distortion. The ‘gentlest’ of the three Distortion types available. I like to think of it as adding a bit of ‘fuzziness’ around the edges of a sound 
Distortion B	Waveshaping Distortion. The severest Distortion type available – things can get really loud and dirty very quickly with this one (there are points along the slider where the audio ‘breaks up’; if you experience this, just shift the slider a bit to either side of that point 
Distortion C	Soft Waveshaping + Clipping Distortion. The intensity of this “overdriver” is somewhere between Distortion type A and B, but again the sound of this Distortion is different to either of those. There are points along the slider where the audio ‘breaks up’; if you experience this, just shift the slider a bit to either side of that point) 
Filter	A Low Pass filter whose characteristics are very loosely modelled on a Moog LPF. F = Cutoff Frequency Q = Resonance ‘Quality’. This is a peaking or accentuation of the frequency response of the filter at a specific frequency. Try bringing the level of the Q Slider close to that of the F Slider to really hear the effects of this ‘peaking’

	
Chorus	This is a modulation effect used to create a richer, thicker sound and add subtle movement. It simulates the slight variations in pitch and timing that occur when multiple performers play or sing the same part. To create this effect, the audio is split and run through a very short delay, then mixed with the original audio. To add movement, the delay time is slowly modulated with a low-frequency oscillator (LFO). Del = How much the signal is Delayed Dep = Depth (intensity) of the Chorus effect F = Frequency of the modulating LFO 
Delay	Creates the sound of a repeating, decaying echo. T = Time interval between each distinct echo F/B = How quickly the echoes decay BPM: If you optionally set a value say, ‘120’ in this box, the interval between each distinct echo will be synched to 120 BPM, and each echo will occur according to the note spacing selected in the ‘Note’ drop down box 
Compression	Compression reduces the dynamic range of a sound. All signal values above a certain adjustable threshold are reduced in gain relative to lower-level signals. This creates a more even sound level, reducing the level of the loudest parts. T = Threshold, above which the level of the audio signal is reduced. The farther the slider is to the right, the higher the Threshold level R = Ratio. Determines the amount of level reduction. The farther the slider is to the right, the more severe the reduction A = Attack. Determines the responsiveness of the compressor. The farther the slider is to the right, the slower the response speed, thereby ‘smoothing’ the effect So, in order to experience the result of extreme compression, try setting the level of the T slider close to the left, the level of the R slider close to the right and the level of the A slider close to the left 
Amplitude	One slider changes the level of the positive part of the waveform from anywhere between 0 and 4 times the normalized level, the other does the same thing with the negative part of the waveform. Reducing the amplitude of one side and increasing the other side can alter the sound of your sample quite a bit 

Reverb	This is the effect of many sound reflections occurring in a very short space of time. The sound of clapping in an empty hall is a good example of reverb. This effect can be used to restore the natural ambience to a sound, or to give it more fullness and body. With the sliders provided, you can simulate anything from a very short subtle reverb in a small room to a long and pronounced reverb in a large cavern Amount = The reverberation time. The farther the slider is to the right, the longer is the reverb Bright = This is basically a filter on the reverb such that the farther the slider is to the right, the harsher/metallic sounding the reverb is. Conversely the farther the slider is to the left, the softer/warmer sounding the reverb is. Mix = Determines the balance between the original sound and reverberated sound. The farther the slider is to the left, the less the reverberated sound is mixed in 
Lo-Fi	This is a distortion effect that intentionally reduces audio quality to emulate early digital audio gear, or even beyond to the absolute depths of grittiness and grunginess. This distortion effect sounds different to the other Distortion types featured in FlexibeatzII (Distortion A-C), so warrants its own inclusion. There are two parameters you can change to reduce audio fidelity: Sample Rate reduction (Freq value) and Resolution Reduction (Bits value). Sample Rate reduction: To accurately represent a smooth waveform, digital audio requires a large number of samples at a high sample rate. The higher the rate, the more accurate the waveform - a sample rate of 44100 Hz is ‘CD quality’. Early digital gear used much lower sample rates than this to conserve memory for stored audio. Sample rate reduction intentionally reduces the sample rate to degrade the quality of the audio; as the sample rate is reduced, waveforms become coarser and high frequencies are lost. Resolution reduction: Samples in digital audio are recorded as integers or floating-point (real) numbers, and these numbers are encoded using a series of on and off memory bits. The greater the number of bits, the more accurately a sample encodes the instantaneous volume level of a sampled audio waveform. Early digital audio gear used 8 bit or even 6 bit integer samples (compared with 16 bit ‘CD quality’). Resolution reduction intentionally reduces the number of bits used for audio samples, and as the bit depth goes down, waveforms become more stair-stepped and subtle volume variations are lost.  Want to hear what a sampling rate of 8000 Hz or 4000 Hz or even less, and bit resolution of 5 bits or even less sounds like? Give it a try!
Reverse	Does as it says on the tin – reverses the sound 

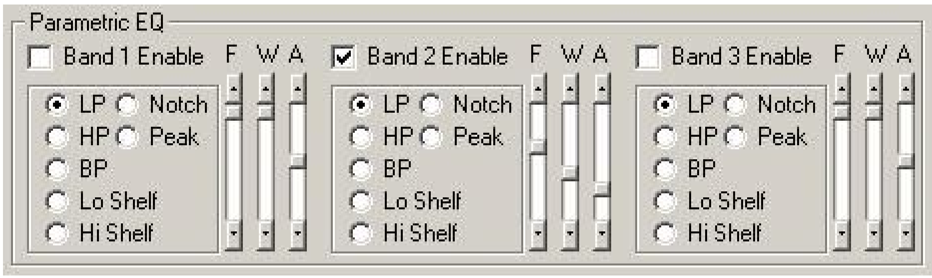
Parametric EQ

The Parametric EQ is a multi-band variable Equalizer, which allows you to control the three primary parameters of filters operating on bands of frequencies – amplitude, centre frequency and bandwidth. The amplitude of each band can be controlled, and the center frequency can be shifted, and widened or narrowed.

You can use the Parametric EQ to improve the fidelity of a sound, to emphasize or de-emphasize the sound in a mix and make it sit the way you want, to reduce "boominess" or "ringy" tones or "muddiness", to remove undesired noises, or as a creative effect to completely change the characteristics of your sound.

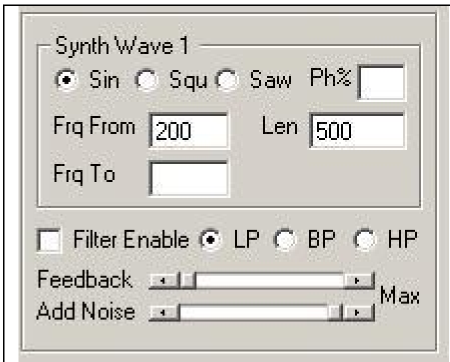
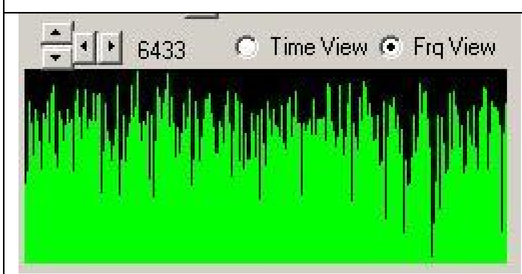
FlexibeatzII gives you a bunch of EQ filters you can apply across three frequency Bands:

Band 1: 39-408 Hz
Band 2: 212-2350 Hz
Band3: 1800-20000 Hz



The range of each F slider is from the minimum frequency of the Band at the uppermost position of the slider, to the maximum frequency of the band at the bottom position of the slider. The frequency Bands overlap.

You can of course enable the same or different EQ filter types in each Band. The role of each of the sliders is to process the signal at F Hz by A Db over a bandwidth of W Octaves (note however that the A slider is not applicable to all the EQ filter types). Shown below is what each EQ filter type does:

	First we generate a short burst of noise with which to demonstrate the effect of the various EQ filters (Add Noise slider is pushed to maximum, resulting in a half second noise signal being generated)
 **Unmodified**	This is what the noise waveform looks like with no EQ applied. We see that it contains pretty much every frequency in the range of human hearing in equal amounts

<table>
<tr>
<td>
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<td>We now look at the result of applying each of the seven EQ filter options to the noise waveform</td>
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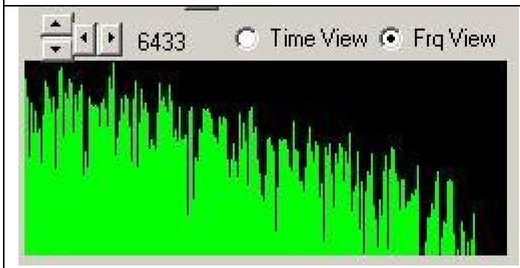
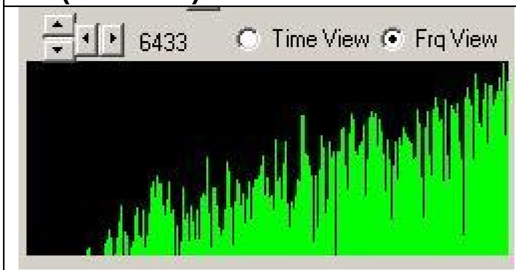
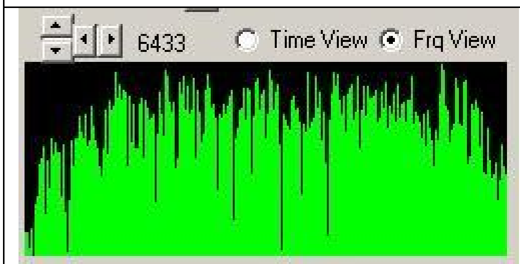
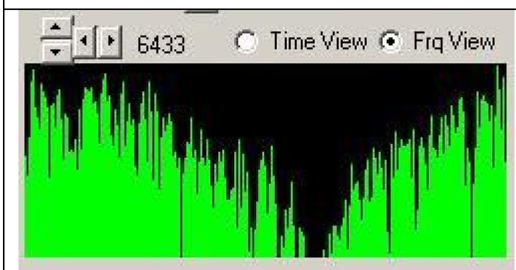
<td>

LP (Low Pass)</td>
<td>The LP filter allows frequencies lower than the cutoff frequency to pass while severely attenuating above it.
For example, applying LP filtering at around 8-9 kHz can reduce the effects of sibilance on vocal snippets
Note that only the F and W sliders are relevant for the LP filter. The setting of the A slider has no effect</td>
</tr>
<tr>
<td>

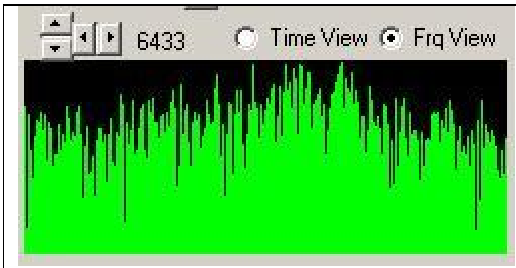
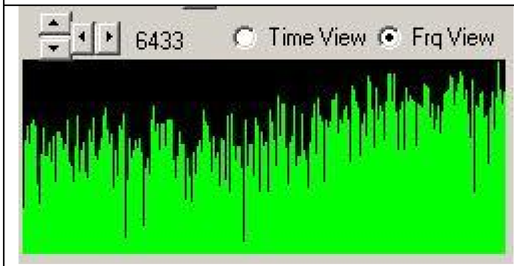
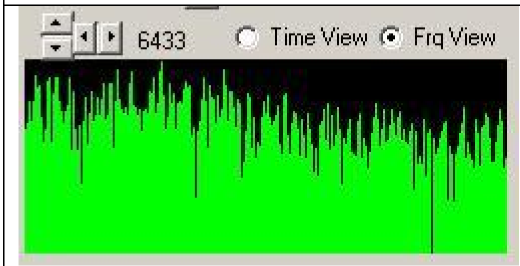
HP (Hi Pass)</td>
<td>The HP filter allows frequencies higher than the cutoff frequency to pass while severely attenuating below it.

Use it to remove any low lying tones that are interfering with the sound you want. For example, HP filtering can reduce the effects of electrical hum or mic plosives or knocks on vocal snippets

Note that only the F and W sliders are relevant for the LP filter. The setting of the A slider has no effect</td>
</tr>
<tr>
<td>

BP (Band Pass)</td>
<td>BP attenuates frequencies below and above the cut-off and leaves the frequencies around the cut-off. It is effectively a LP and HP filter together.
It’s great to use as an effect (it can give a mid-range type of old radio sound or simulate a telephone voice), or for isolating a band of frequencies in sounds that have too much low and high end.
Note that only the F and W sliders are relevant for the LP filter. The setting of the A slider has no effect</td>
</tr>
<tr>
<td>

Notch</td>
<td>Allows frequencies below and above the cut-off and attenuates the frequencies around the cut-off point.
Use it for eliminating the frequencies you don’t want, and for creating a new flavour to a sound. For example, it can be useful on drum sounds that have a muddy or heavy mid-section, or on sounds that have a little noise or frequency clash in the mid section
Note that only the F and W sliders are relevant for the LP filter. The setting of the A slider has no effect</td>
</tr>
</table>

 **Peak**	Boosts a specific band of frequencies as a bell shaped curve. Setting of F slider determines the centre Frequency, W slider determines the width of the bell, and A the height of the bell. It's good for making focused adjustments, like adding more crack to a snare drum, or adding presence to a vocal snippet.
 **Hi Shelf**	Useful when the overall balance of low and high frequencies is off. Boosts a broad range of frequencies above the cutoff frequency. If for example your sound is a little 'dark', you can use the Hi Shelf to add some high frequencies and 'brighten it up'* ⌂Tip: Set the A slider to a high value
 **Lo Shelf**	Useful when the overall balance of low and high frequencies is off. Boosts a broad range of frequencies below the cutoff frequency. If for example your bass sound is a little weak, you can use the Lo Shelf to bring it up a bit* ⌂Tip: Set the A slider to a high value
*: An example of use of both Hi Shelf and Lo Shelf is that if in your sequence the piano and bass tracks seem to be clashing because of overlapping frequencies, you can separate them sonically by applying a Hi Shelf to the bass and a Lo Shelf to the piano to keep them out of each other's way	

⌂Tips:

Don't boost when you can cut instead. It sounds more natural when you reduce unwanted sounds, rather than boosting the desired frequencies.

Human hearing is most sensitive to midrange and upper midrange frequencies. Because of this sensitivity, large boosts in this range can quickly make your sound seem harsh or shrill. Characteristics of certain frequency ranges:

20 to 100 Hz	'warms' an instrument, adds boominess
100 to 200 Hz	'muddy' for many instruments, adds 'fullness' to a few
350 to 450 Hz	sounds boxy
700 to 850 Hz	adds depth/body
1 to 2kHz	adds attack or punch to some instruments and creates nasally sound in others
2 to 5kHz	increases presence of instruments
5kHz to 8kHz	sounds harsh in some instruments
>8kHz	adds airiness or brightness

How to apply this to a vocal snippet:

To add fullness	Boost a bit at 150Hz
To get rid of muddiness	Cut a bit at 200-250Hz
To add clarity	Boost a bit at 3kHz
For more presence	Add a bit at 5kHz

To brighten	Boost at 10kHz
To get rid of sibilance	Cut a bit at 7.5kHz and 10kHz

Drums and bass:

To ensure both kick and bass can be clearly heard	In order to make room for both, it is common to cut frequencies from the bass to add to kick
To add depth to snare	Boost at 200Hz
If snare sounds too boxy	Cut at 800Hz
To increase attack of snare	Boost at 5kHz
To add crispness to snare	Boost at 10kHz

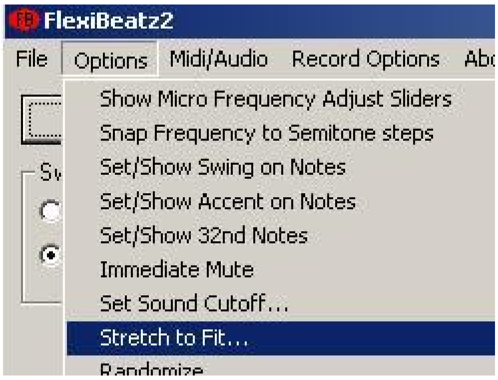
Synchronize sound loops in multiple channels to the Master Tempo

Let's say you have a FlexiBeatzII pattern running at a low Tempo. You find a pre-made loop online, or on a sample CD, or on a dusty cassette still lodged in your ancient four-track. You think the loop will mesh well with your pattern. To sync it, increase the Tempo of your pattern close to that of the loop, open the pre-made sound loop in a spare channel, assign it to one or more notes and Run. If your pattern and the new loop don't play in sync, first try altering the 'Frq' slider of the channel and set it to what sounds to be the closest match possible, then choose option 'Show Micro Frequency Adjust Sliders' - moving the Micro Frequency slider will 'fine tune' the setting around the 'Frq' setting. If you want even finer adjustment, you can click above or below the 'Frq' slider handle on the slider guide rail itself – each click will minutely adjust the Tempo of the pre-made loop up or down.

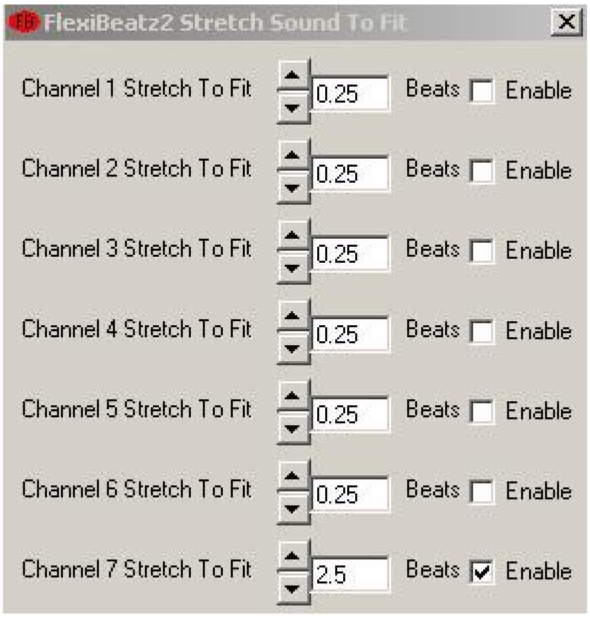
Once it sounds as tight as you want it, tick the 'BPM Lock' checkbox above the channel. Now when you adjust the master Tempo back down, the pre-made loop remains in sync with your pattern. You can of course repeat this process with additional pre-made loops if you've got spare channels, and build up quite a complex pattern. Note that FlexiBeatzII will change pitch of the pre-made loop when you change its Tempo.

FlexiBeatzII gates its channels, which means that the most current playing of a sample cuts out the playing of any previously playing sample on the same channel. You can use this feature to make new loop variations. Make sure to set each loop to Stretch To Measure or tempo will matter. The Stretch To Measure option eliminates the necessity for knowing a loop's tempo as long as the loop is not required to match a certain pitch. Adjusting the tempo will not affect the pitch of the normal one-shot samples, but will do so with the samples in the channels locked to BPM. In Hammerhead, the "stretch to measure" option shifts the pitch of the sample to fit in exactly one measure, and if you want to make a user sample of a breakbeat longer than one bar, the recommendation is to use a sample/wave editor to cut the sample in two equally sized halves, then import them as to two separate user samples. You do not have this limitation with FlexiBeatzII - the equivalent of stretch to measure in FlexeBeatzII can stretch a sample over multiple measures or sub-measure.

In the following example, we use Stretch To Measure to fit a drum loop in Channel 7 to a pattern. Choose Options>Stretch To Fit.

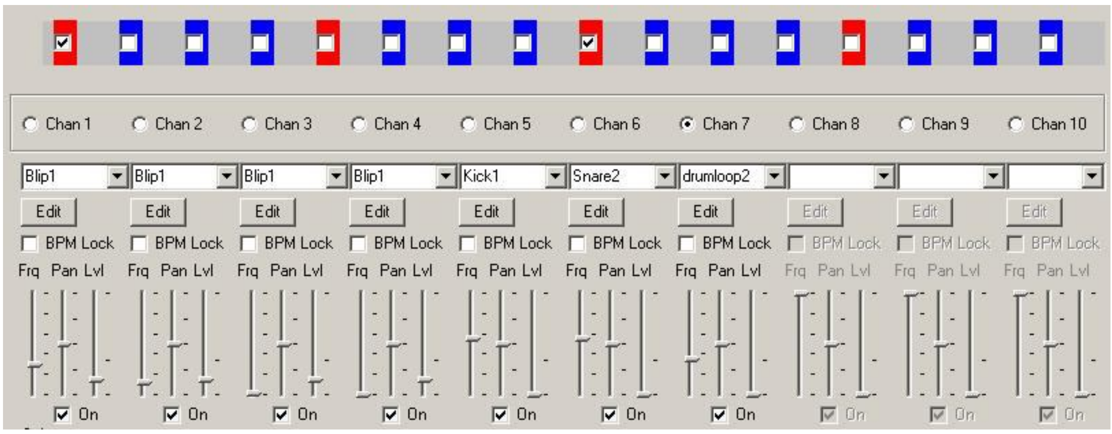


This opens up the 'Stretch Sound to Fit' panel where we select the channel of our sample, enable stretch by ticking the corresponding checkbox, and scrolling to multiples of a measure, or a fraction of a measure, in 1/4 measure steps. FlexiBeatzII automatically calculates the maximum multiples of a measure, and minimum fraction of a measure you can scroll to. As our sequence plays our selected Measure, we scroll through values and find 2.5 sounds good.



We then mark the Notes in the Measure we want to play this sound.

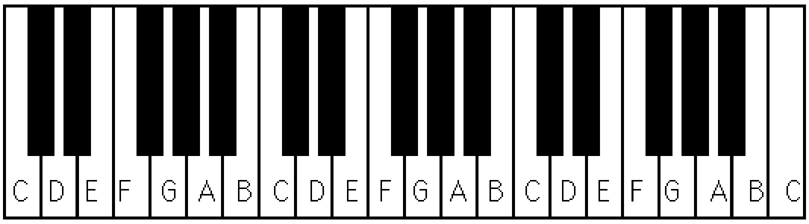
Once FlexiBeatzII has shortened or stretched the sample (see the position the Frq level slider was placed in for Channel 7), don't forget you can still 'fine tune' the stretch using the Micro Frequency slider:

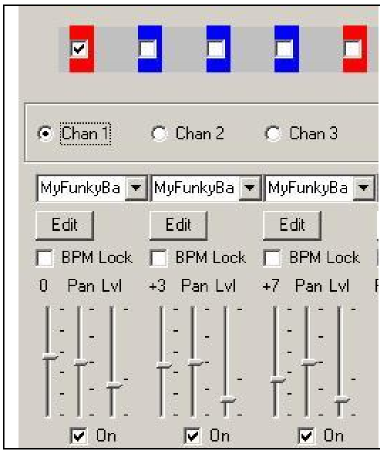


Play Basslines and Melodic Riffs

Let's say you open your favourite 'MyFunkyBass' sound on 3 consecutive channels. If you select the option 'Snap Frequency To Semitones', moving the Frq slider of each channel will 'snap to' a positive or negative integer value eg +1, +2 etc or -1, -2 etc. This facility allows you to play a 3 Note Bassline with your 'MyFunkyBass' sound.

First, a quick recap on keyboard layout. There's a pretty good chance you've noticed your synth keyboard is a collection of white and black keys. Chances are you've also noticed the black keys are grouped together in alternating groups of two and three. The black notes help you understand 'where you are' on the keyboard. The note 'C' is the white note that is just to the left of each group of two black notes. If you sit roughly at the middle of the synth and look down, you should be looking at MIDDLE C. In other words, there are several C's as you glance up and down the keyboard, but the one in the middle is MIDDLE C. The smallest interval between two notes is a semitone. Thus, a semitone above the note 'B' is 'C'. A semitone above 'C' is a black note that is lowest of a group of two black notes – called 'C-sharp'. Any black note can be named for the white note just below it in the same manner by suffixing a '-sharp' to it. There are 12 semitones in an octave, so the distance from MIDDLE C to the next C up on the keyboard is 12 semitones:





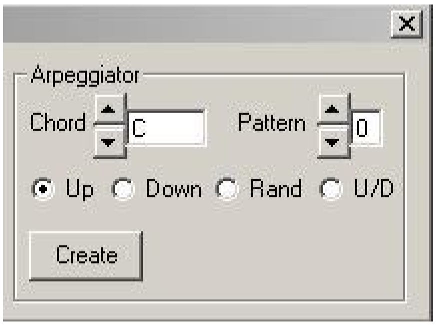
Now let's say you either know or are told that the 'MyFunkyBass' sound you've got is playing a Middle C note (frequency 261.6 Hz i.e. C4).

If you want to play a Bassline that goes C to D-sharp to G, you can see from the keyboard picture that D-sharp is 3 semitones above C, and G is 7 semitones above C. Thus, you would set the Frq sliders for the 3 MyFunkyBass channels as 0, +3, +7 respectively

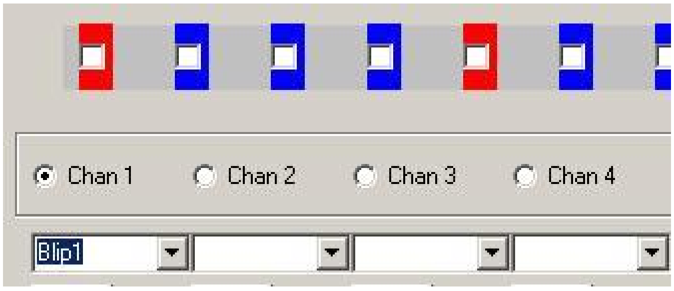
The accompanying screenshot shows these settings

Arpeggiate

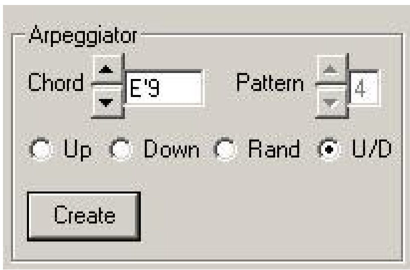
Go to Options>Randomize. This opens a panel captioned 'Randomize/Arpeggiator'.



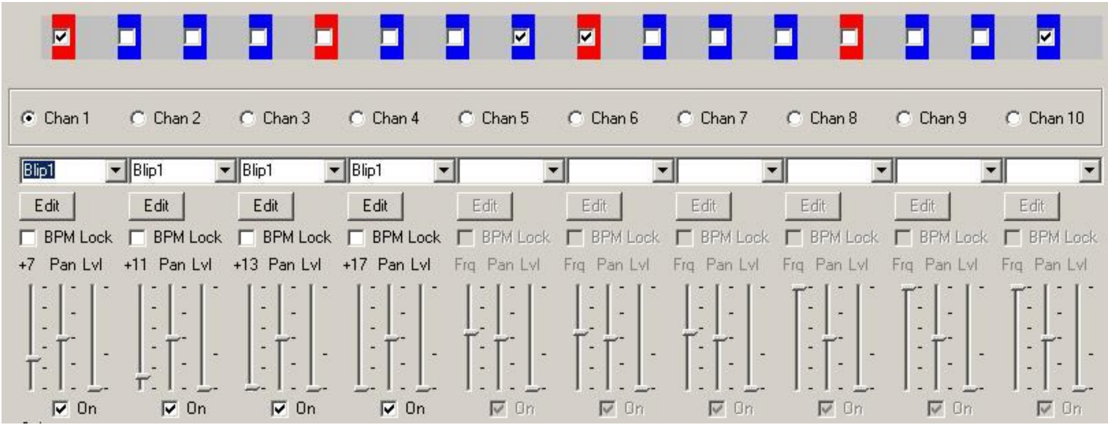
In the group of controls titled 'Arpeggiator', pick a chord from the scroll box (eg C, D7+ etc). Then choose how you want the notes of the chord to be played in the arpeggio - up, down, up then down or arranged randomly and repeated. Once you've made your selection, pressing the 'create' button does the following things:
1) Erases the notes in all channels in the current measure
2) Opens the sound in channel 1 in additional consecutive channels; the number of channels in which the sound is selected is equal to the number of notes in the chord - for example if there are 4 different notes in the chord, then the sound in channel 1 is also opened in channels 2-4
3) Automatically adjusts the frequency of the sound in each channel to correspond to the note of the chord
4) Sets the appropriate notes pattern in the current measure for each channel. You can then have other sounds playing along with the arpeggio in the free channels, or you can create a .wav file of the arpeggio and open it up as a sound in a channel. You could export multiple arpeggios corresponding to different chords, and open them up in spare channels to play an arpeggiated chord sequence.
To illustrate the arpeggiator in action, we open a sound in Channel 1:



We now open the 'Randomize/Arpeggiator' panel, set the chord we wish to play and the sequence in which the notes in the chord will be played – in this example we choose ‘U/D’ i.e. ‘Up Then Down’



We now press ‘Create’, which results in FlexibeatzII opening the Channel 1 sound into Channels 2 to 4, setting the Frq slider level for each sound, and setting the appropriate notes in the Measure for each sound, such that when we press ‘Run’; we hear the desired arpeggio:

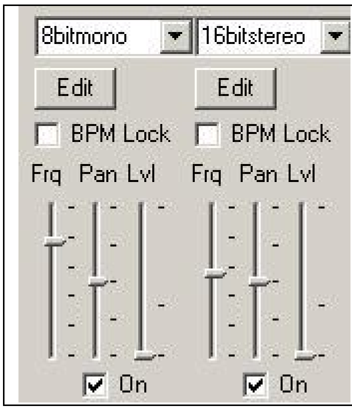


Here are the notes patterns set up for each arpeggio option. 1, 2, 3, 4 below pertain to the 4 notes of the chord selected in the Chord scrollbox (in other words they relate to the 4 sounds loaded into Channels 1 through 4, with their Frq slider levels appropriately set):

Up, Option 0	1 2 3 4 1 2 3 4 1 2 3 4 1 2 3 4
Up, Option 1	1_2 3 4_1 2 3_4 1 2_3 4
Up, Option 2	1_2_3_ _4 1_2_3_ _4
Up, Option 3	1 2_3 4_1 2_3 4_1_2_
Up, Option 4	1_2_3 4_1 2_3_4 1_2
Up, Option 5	1_2_3_4 1_2_3 4_1_
Down	4 3 2 1 4 3 2 1 4 3 2 1 4 3 2 1
U/D (Up then Down)	1 2 3 4 4 3 2 1 1 2 3 4 4 3 2 1
Random	The sequence of the first four notes is randomized, and the next 3 blocks of four are the same repeating pattern as the first block of four. For example, say the random function comes up with the following sequence for the first four notes: 3 1 4 2. Thus the full sequence becomes: 3 1 4 2 3 1 4 2 3 1 4 2 3 1 4 2

‘Hot Repitch’ Sounds

Move the channel Frq sliders while the pattern is playing and you may end up with interesting results that inspire you further or help you move in a different direction.

	The range of the Frq sliders goes from 100 (top) to 100000 (bottom). When you open a wav file in a channel, the level of the Frq slider will default to the sampling rate of the wav file. In the accompanying screenshot, the sampling rate of the first sound (8bitmono) is 22.05 kHz, so the Frq slider level is around 22000. The sampling rate of the second sound (16bitstereo) is 44.1kHz, so the Frq slider level is around 44000. From these ‘load time’ levels, when you move the Frq slider towards the top, the pitch of the sound lowers (and duration of the sound lengthens). When you move the Frq slider towards the bottom, the pitch of the sound increases (and duration of the sound reduces)

- Repitching the same vocal sound in two channels, can give you instant soprano with baritone backing.

- Repitching drum samples down can result in some dark, brooding soundscapes, and repitching up can result in Kraftwerk-esque pops, pings, clicks and bleeps.

- Pitch a tambourine sound an octave down and you get a submarine. Pitch a paper crumpling sound an octave up and you get a crackling fire.

- You don't need a new set of samples to accommodate different musical styles. House music pitches drums lower, whereas drum 'n' bass often pitches them up.

- You can create multiple drum sounds from one. For example, play a two-hand shaker part, from only one shaker sample. Copy it, then detune the copy by a semitone or so to provide a slight sonic variation. Detuning can also create a family of toms out of one tom sample.

- You can tune drums to the song's key, particularly beneficial with toms and resonant kick drums. If the kick is out of tune with the bass, the sound can be muddy, and weaken the rhythm. Sometimes simply tuning a semitone up or down makes all the difference, but if fine tuning is needed, that is also available

- You can make radical transpositions to create new sounds. For example, create a gong out of cymbals. Take a long cymbal sound and detune it by 12 to 20 semitones. Create another version of the cymbal and detune it by about 3 semitones. Trigger the two together; the slightly detuned cymbal gives a convincing attack, while the highly detuned one provides the necessary sustain

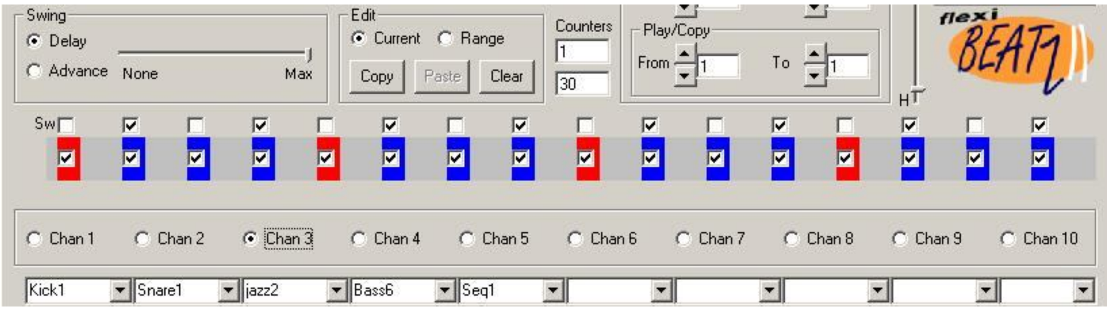
⌂Tip: By playing around with the Frq slider of the different channels while the sequence is playing, you can completely change the feel of the sequence. Be aware though that if you decrease the pitch of a sound too much, it can become too 'boomy' and overwhelm the bottom end of your music when you record your sequence. So, in order to use the sound in your final mix, you should simply record the pitch-decreased sound, open it up in a channel, leave the Frq slider untouched once opened, and use FlexibeatzII's Parametric EQ and other processing tools on the sound to make is sit just right with respect to the other sounds in your mix.

Born To Swing

Choose option Options>Set/Show Swing On Notes.

A set of checkboxes appears above the Note checkboxes – one swing checkbox above each note checkbox. Simply tick the swing checkboxes corresponding to the Notes you want to delay or advance. Once you're done ticking, go to the 'Swing' group of controls, choose 'Delay' or 'Advance' and move the slider to the amount you want the Notes delayed or advanced. Some common settings are:

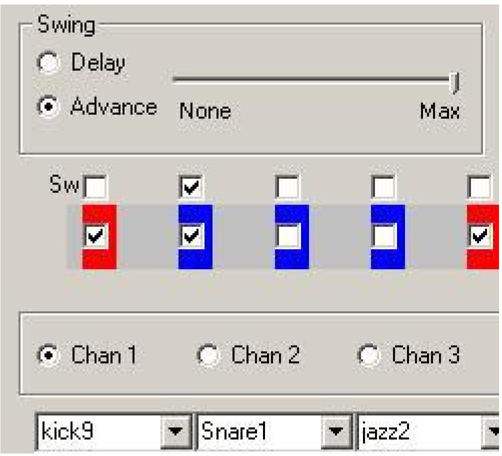
- Classic 16^{th} Note Shuffle: Tick the Swing checkbox corresponding to every other 16th note, and choose 'Delay'. To help 'lighten' the delayed Notes in a natural way, you can try putting an Accent on every Note other than the Delayed ones.



- Tick the Swing checkbox corresponding to the Notes on the first and third beat of the 4/4 bar, and choose ‘Advance’. This can create a sense of urgency, suited to styles where there is usually a bass drum hit on the first beat of the bar.

- Tick the Swing checkbox corresponding to the Notes on the second and fourth beat of the 4/4 bar, and choose ‘Delay’. This will create a lazy or 'laid-back' feel

- You can set Swing on adjacent snare or kick hits for example, to create 'stutter' or 'flam' effects (sounding like double hits):

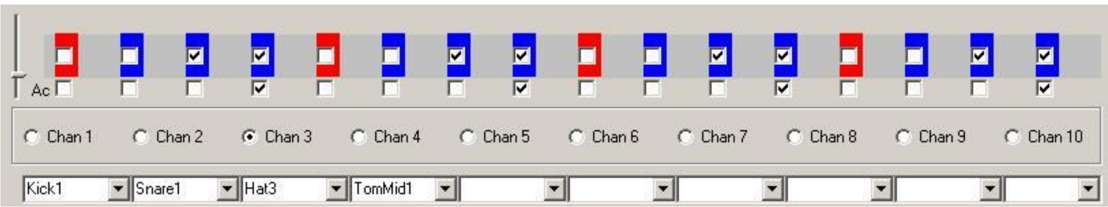


Note: All instruments with a sound on a Swung Note, will Swing

Nice Accent

Choosing option Options>Set/Show Accent on Notes:

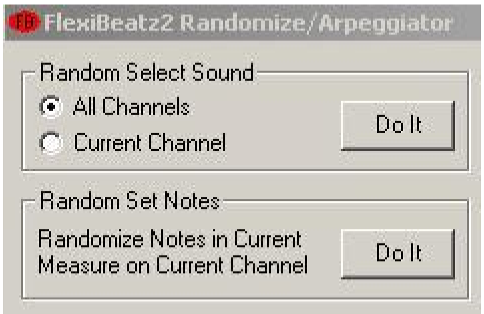
displays a row of checkboxes beneath the Notes checkboxes, and a slider to the left of the checkboxes (Accent Slider). These allow you to make specific notes sound louder than the other notes. Choose a channel, tick the checkboxes corresponding to the notes you want to Accent, set the Lvl slider of the channel about mid-way and the Accent Slider close to Max (50).



This results in all notes other than the Accented notes play at mid-volume, but the Accented notes play at a level between mid-volume and maximum volume determined by the setting of the Accent Slider (in this example since Accent Slider is set close to Max, the Accented Notes play at near-max-volume). You can set note Accents independently for each channel.

‘Hot Swap’ Sounds

You’ve laid a kick pattern on channel one, snare on two, closed hat on three, open hat on four, claps on five. Now take these ‘textbook assignments’ and swap completely different sounds in and out on the various channels while the pattern is playing – not just replacing ‘kick 3’ with ‘kick 4’ on channel one or ‘snare Y’ instead of ‘snare X’ on channel 2 – try really different sounds. You can either swap out sounds manually from the drop-down lists under the Channel radio buttons, or you can use FlexiBeatzII's Random function. To use the Random function, go to Options>Randomize. This opens a dialog box captioned 'Randomize/Arpeggiator'.



In the group of controls titled 'Random Select Sounds', you can choose one of two options - 'All Channels' or 'Current Channel'.

If you choose 'All Channels' and click on the button 'Do It', a random selection of sounds opens up in all channels.

If however you choose 'Current Channel' and click on the button 'Do It', a randomly selected sound opens up in the current channel only.

The group of controls titled 'Random Set Notes' contains a 'Do It' button which when clicked firstly erases whatever notes are set for the current channel in the current measure, and then replaces them with a random pattern of notes.

You can stumble on some intriguing patterns this way.

Free Your Channels

Let's say you set two sounds in independent channels to trigger on the same note, and you like the sound of these two layered sounds eg a gong and a snare. Why not mute all channels except the two, set the note for the two sounds to be the same, tick the ‘Wav’ checkbox, and hit 'Run'. This allows you to record the two overlapping sounds as a single sound. You can now open this sound in its own channel, and use the gong and snare channels for something else. For more about recording, see section ‘Just For The Record’.

Audio Device Selection

<table>
<tr>
<td>
</td>
<td>If you have multiple soundcards or audio interfaces installed on your computer, you can select one to play the sound output of FlexibeatzII through.

Simply go to Midi/Audio>Midi/Audio Settings, and in the dialog box that pops up, select your Audio device from the ‘Audio Device’ dropdown listbox.

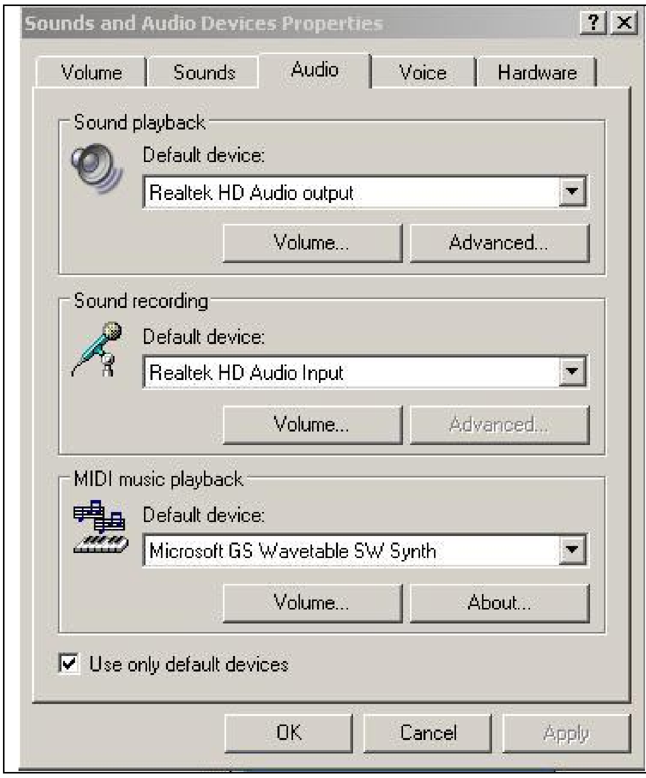
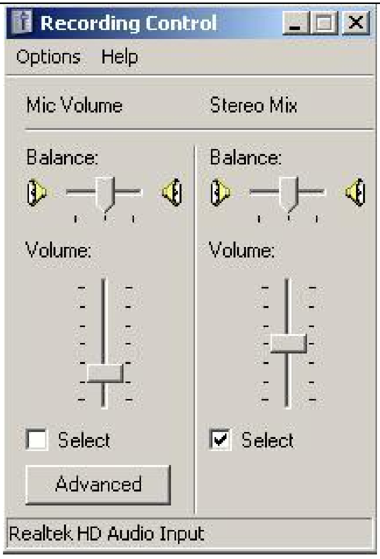
In this screenshot, a Zoom H4 portable recorder plugged into the USB port of the computer and acting as an audio interface, has been selected</td>
</tr>
<tr>
<td>
</td>
<td>After you have selected your Audio Device, the Audio Device selection dropdown listbox becomes greyed-out when you first load a sound into a channel. If you subsequently wish to select a different Audio Device, you first need to go to Options>Clear All. This re-enables the Audio Device dropdown listbox</td>
</tr>
</table>

Just For The Record

In order to record the output of FlexiBeatzII to a .wav file, you must first ensure your Windows Control Panel settings are correct for recording.

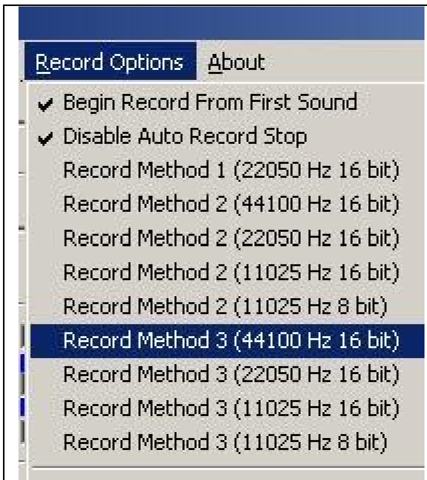
Go to Windows Control Panel>Sounds and Audio Devices>Audio tab. In the Sound Recording section, choose your soundcard as default recording device. Pressing 'volume' will open up a dialog box with recording sliders. Ensure that the checkbox against the slider labeled 'Stereo Mix' or 'What You Hear' or equivalent, is ticked (as opposed to checkbox against 'Line Volume', 'Mic Volume' etc).

The following screenshots show these Control Panel settings:

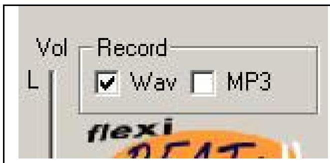
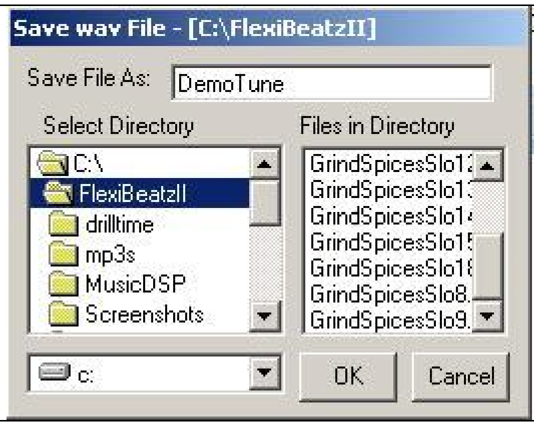
	

Recording to a .wav File

To enable recording in FlexiBeatzII, proceed as follows:

	Record Options>Record Method1 uses a multimedia control component for recording. Record Options>Record Method 2 uses Windows API code to perform the recording. With this Method, alternatives are provided to perform the recording at lower sample rates and bit resolution if desired. Record Options>Record Method 3 uses DirectSound streaming to perform the recording. With this Method, alternatives are provided to perform the recording at lower sample rates and bit resolution if desired.

Recording is enabled by ticking the checkbox captioned 'Wav' under the ‘Record’ group of controls. When you tick this checkbox, a dialog box pops up for you enter the title of the recording, and path to save to.

If you don't have Midi>Midi Sync ticked, recording begins when you press the 'Run' button and finishes when the total number of measures have been played (i.e. what is set in the Total box in the Measures group of controls).

If you do have Midi>Midi Sync ticked, recording begins when you rewind your sequencer to the start and press play, and finishes when your sequence finishes.

In either case, not just the FlexibeatzII output, but also any sounds playing along or in parallel with the FlexibeatzII output, will be recorded.

The following are additional recording options available:

If you tick Record Options>Disable Auto Record Stop, recording begins when you press the 'Run' button, but finishes only when you manually press the button again (i.e. press 'Stop'). This way, you can for example have a 1 measure pattern loop over and over, and mute and un-mute instruments with each pass, and have all the variations recorded in one long multi-measure pattern. This option is not available if you have Midi>Midi Sync ticked.

Note that if you tick Options>Immediate Mute, then whenever you untick the 'On' checkbox for a given channel to mute the channel, the sound will stop immediately. If you don't tick Options>Immediate Mute, then whenever you untick the 'On' checkbox for a given channel to mute the channel, the sound - if it is currently playing - will play to the end, then not play again until the 'On' checkbox is ticked again.

When you want to layer two or more sounds and record them as one sound: Note that even if you set the sounds to trigger on the first note of the measure, FlexibeatzII records from the start of the measure, not from the first note position (hence not from the start of the sound), which means a little leading silence will be recorded. This means that if you want to use the recorded sound as an instrument in a channel, you should first either trim the leading silence in a sound editor like Audacity, or set the 'From' slider on the Sound Edit panel to the point where the sound starts.

Recording to a .mp3 File

	As described in the 'Recording to a .wav File' section, after you tick the 'Wav' checkbox, a dialog box pops up for you enter the title of the recording, and path to save to. Once you enter these details and close the dialog box, you have the option of ticking the 'MP3' checkbox in addition to the 'Wav' checkbox. If you tick the 'MP3' checkbox then once recording completes after you press the 'Run' button, BOTH a .wav file AND a .mp3 file (with the name and path you specified in the dialog box), will be created. This is handy when you want to quickly share your loops, patterns and sound sketches online, as it means you don't have to separately convert the .wav file to a .mp3

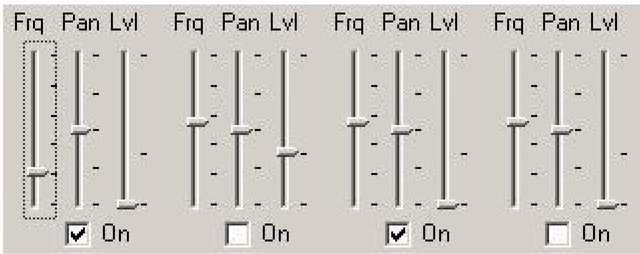
Loop Export

Using the loop export capability, you can have FlexibeatzII assemble a wav file from all the sounds in all the notes between 'From' Measure and 'To' Measure in the 'Measures' group of controls, at the Tempo you specify, without having to play and record the sequence.

Notes:

- The sounds have to be Rendered first. Just loaded sounds in channels will not appear in the export
- Those settings which only affect how a sound is played back and are not part of the DSP processing of the sound when the sound is Rendered, will not have any effect on the sounds in the exported loop. These settings are the Vol, Pan and Lvl sliders beneath each channel, and the 'From' and 'To' sound playback sliders on the Sound Edit panel.
- Any other sounds being played through the audio device at the time you perform the loop export, will not be recorded in the exported loop
- Any unticked 'On' checkbox beneath a channel will prevent the sound in that channel from appearing in the export

As an example to illustrate the last dot point, if a loop is exported with sounds in 4 channels loaded and Rendered, and the below sliders are set, the sounds in channels 2 and 4 will not appear in the exported loop wav, and the high pitch setting of the sound in channel 1, will have no effect on the channel 1 sound in the exported loop wav



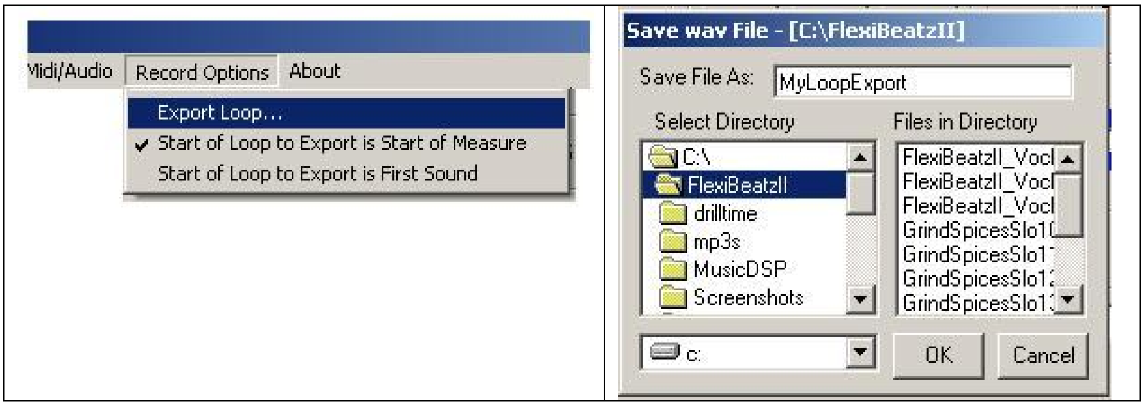
To export a loop:

First choose:
Record Options>Start of Loop to Export is Start of Measure
OR
Record Options>Start of Loop to Export is First Sound

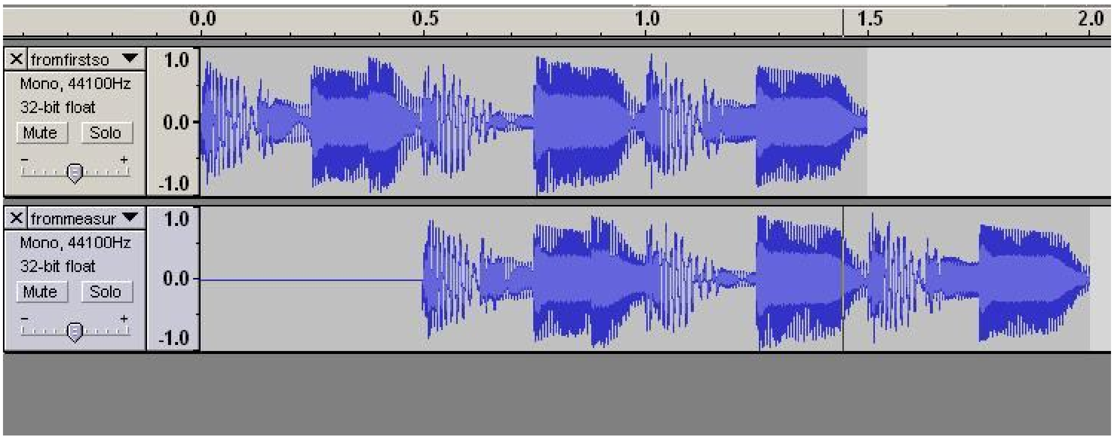
Then choose Record Options>Export Loop…

This will open up a dialog box where you specify the name and path of the Loop to be saved.

Pressing 'OK' exports the loop to the specified path

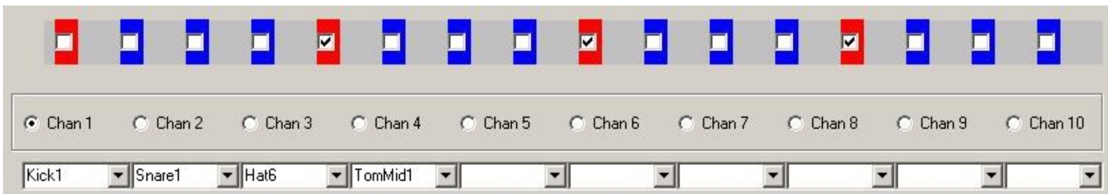


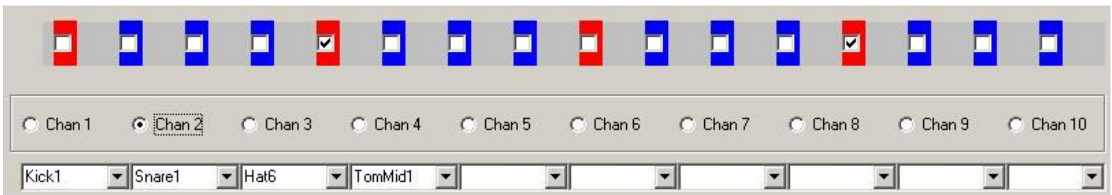
In the following screenshot, you can see the difference in the exported loop depending on whether
Record Options>Start of Loop to Export is Start of Measure
is chosen (second wav image)
OR
Record Options>Start of Loop to Export is First Sound
is chosen (first wav image)

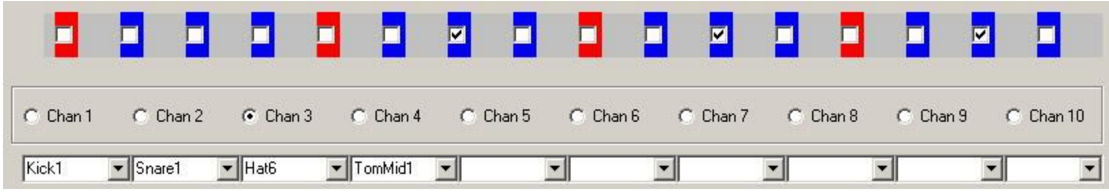


This screenshot was taken after importing the exported loops into Audacity. Pressing “Shift-Play” (continuous play) in Audacity on the first imported wav file above will play the loop over and over with perfect looping.

Here is the sound data in each of the channels in FlexibeatzII that got exported as the above wavs:

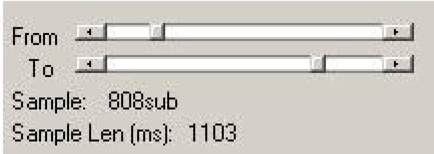






Slice n' Dice

As the pattern is playing, you can dynamically change the sample playback start and end points in each channel to obtain some interesting effects, and even completely change the feel of your pattern or song at some settings. To do this, go to the Sound Edit panel of the channel containing your sample, where you will see From and To sliders underneath the Render button

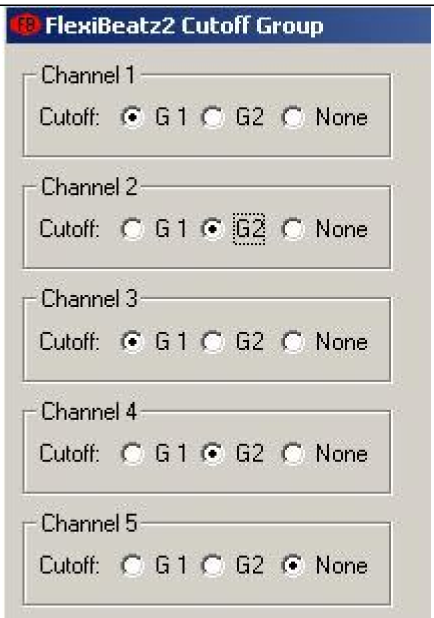


The length of each slider represents the length of the sound in the channel. Thus when the From slider is at the extreme left, the sound in the channel triggers from the start of the sample, and when the slider is at Max, the start point for sound triggering is set to the end of the sample (so you won't hear anything). Similarly, when the To slider is at the extreme right the endpoint of the sound playback is set to the end of the sample, but when the slider is at the extreme left, the endpoint for sound playback is set to the start of the sample (so you won't hear anything). By changing the sample start point and endpoint for playback, you can play any section of a sample without having to create and load multisamples. For example, in the screenshot, the mid-section of a sample is being played (forgoing a bit of the start and finishing a bit before the end of the sample)

If you go to Options>'Set Sound Start/Cutoff' panel, you will see 'cutoff' radio buttons labeled 'G1' (Group 1), 'G2' (Group 2) and 'None' (No Group assignment) against each channel. If you set multiple channels to the same cutoff Group, triggering a sound that is part of the Group will cut off any other sound in the Group that's still playing. This type of feature is traditionally used in programming hi-hats, such that playing a closed hi-hat sound shuts off an open hi-hat. All you need do is assign the channels playing closed and open hi-hats to 'G1'. But there are other possibilities:

- Assign toms with long decays to the same cutoff Group. Too many simultaneous tom decays can muddy up a track. When they are assigned to the same cutoff Group, the tom rolls sound cleaner
- Choke a Cymbal by another sound
- Cut off sustained sound effects eg a looped thunderstorm is cutoff with a crack of lightning.
- If you have some rhythmic loop samples playing as well as individual drum sounds, make the loops part of a cutoff Group. This means you can have a four-measure loop

playing, but are able to switch between the first measure, and first two measures, of various loops. Assigning loops to the same cutoff Group means you can start them whenever you like, knowing that the other ones will stop



In this screenshot:

Sounds playing in Channel 1 and Channel3 will cut each other off

Sounds playing in Channel 2 and Channel 4 will cut each other off

The Sound playing in Channel 5 will not cutoff any other sound, nor be cutoff itself

Other Time Signatures

FlexiBeatzII makes loops in 4/4 (or 2/4, 8/8 etc) time. But it is also possible to make loops in eg 3/4, 5/4, 6/8, 7/8 time.

Here's how to make a rhythm loop in 3/4 time:

If we use FlexiBeatzII's red checkboxes for the quarter notes, we see that every measure has four quarter notes. But since there are only three quarter notes per measure in 3/4 time, it is not possible to create a 3/4 drum loop without increasing the total measures to three. Four quarter notes per measure and three total measures makes a total of 12 quarter notes. In 3/4 time 12 quarter notes is the same as 4 measures, so that's the smallest number of bars we can have in our 3/4 loop.

The first kick would then be on 1st red note of the 1st measure, the second one on the 4th, the third on the 3rd red note on the 2nd measure and the fourth on the 2nd red note on the third measure. Put another way, if there are 48 notes in 3 sequential measures, then the kicks are on notes 1, 13, 25, 37. To complete the loop, put snare hits on the quarter notes between the kicks i.e. on notes 5 and 9, 17 and 21, 29 and 33, 41 and 45, and put the hats on every eighth note throughout the pattern i.e. on notes 1, 3, 5, 7, 9, 11, 13, 15, 17, 19, 21, 23, 25, 27, 29, 31, 33, 35, 37, 39, 41, 43, 45, 47.

So, to make four bars in 3/4 time, you will need to set the total measures to three.

To make four bars in 5/4 time, you will need to set the total measures to five

To make eight bars in 7/8 time, you will need to set the total measures to seven

MIDI Sync

You can have FlexiBeatzII start and stop with your Midi sequencer. This way you can opt to have:

a) FlexiBeatzII provide just the beats/rhythm track, with the sequencer playing all the other parts.

Or

b) FlexiBeatzII playing the bulk of the composition, with the sequencer providing a few sounds/effects here and there - 'icing on the cake' if you like.

Or

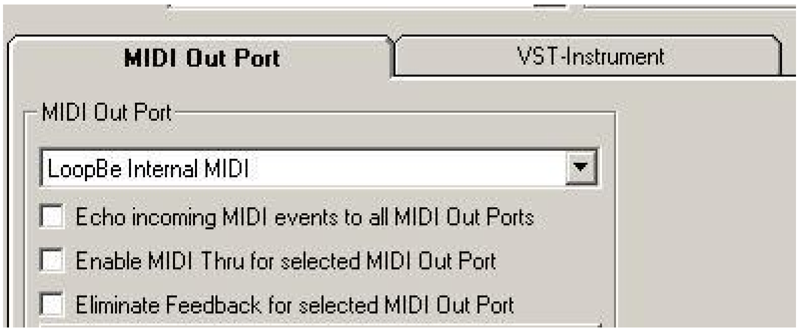
c) Anything in-between a) and b).

A good freeware Midi sequencer I use with FlexiBeatzII is Anvil Studio. In order to link FlexiBeatzII with your Midi sequencer, all you need is a 'virtual Midi cable' between the two. An example is the utility LoopBe. It's free and can be downloaded from many sites. It's a very simple install, and when you run it you get this dialog box:

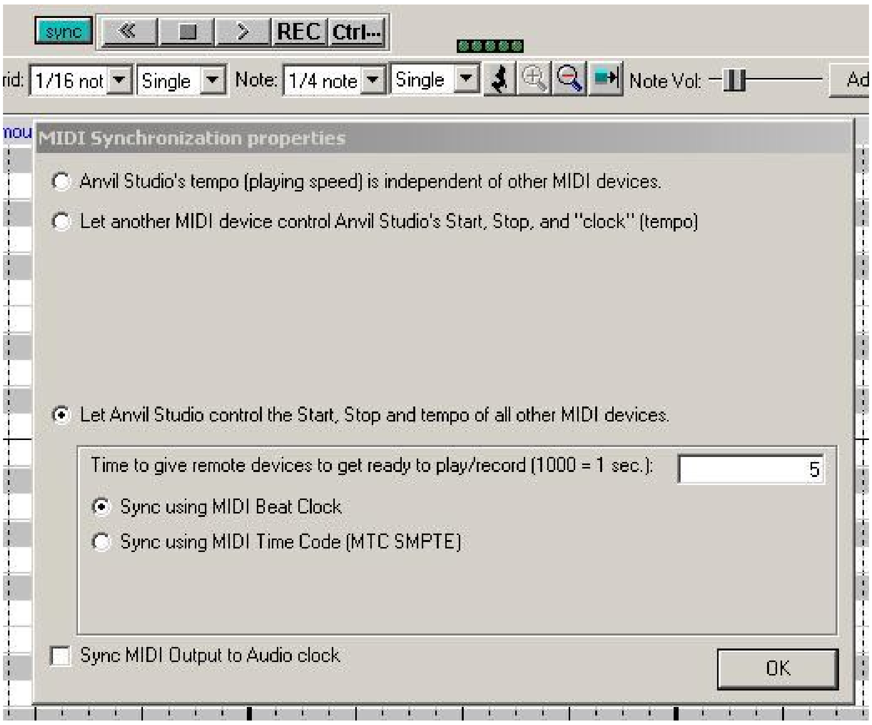
Make sure you don't tick the 'Mute' checkbox. When you press 'OK', an 'internal MIDI port' icon gets placed in your system tray, as shown by the first icon in this screenshot:

Then to synchronize your external sequencer with FlexibeatzII, all you do is as follows: Select Midi/Audio>Midi/Audio Settings. A dialog box pops up. For 'Midi In Device', select 'LoopBe Internal Midi'. For 'Midi Out Device', select whatever device you want the external sequencer to play - such as your soundcard's synthesizer (eg on my old desktop PC, I select 'SB Live! Midi Synth', on my laptop I select 'Microsoft GS Wavetable SW Synth'):

In Anvil Studio, select View>Synthesizers, Midi + Audio Ports. In the dialog box that pops up, select 'Midi Out Port' as 'LoopBe Internal MIDI':



In Anvil Studio, press the 'sync' button (highlighted in blue in the below screenshot), and in the dialog box that pops up, select the following radio buttons:
'Let Anvil Studio control the Start, Stop and tempo of all other MIDI devices'
and
'Sync using MIDI Beat Clock':



In FlexiBeatzII, select Midi/Audio>Midi Sync

The Run button becomes disabled, because FlexibeatzII will now run and stop when you start and stop your external sequencer (in this case Anvil Studio). FlexibeatzII will run at the tempo determined by the external sequencer.

Note that each time you launch FlexiBeatzII, if you wish to sync the application to an external sequencer, you need to set up Midi/Audio>Midi Settings as described in Step 1 above. However, once you've done so, you can enable/disable Midi/Audio>Midi Sync to the external sequencer simply by toggling the Midi Sync flag.

Tips n' Tricks

Like just about every other piece of software ever devised, FlexiBeatzII has some quirks and...ahem...'hidden features'. The more the application is used in different scenarios, the more likely it is such characteristics will be uncovered. I'll itemize them in this section as they are discovered.

1) Occasionally, you might find a sound in a channel 'cuts off' or plays intermittently. Slightly adjusting the Tempo or frequency of the sound can help solve this

2) If you want the Micro Frequency slider to show only a selected channel (and not on all channels or no channels - which are the only options when you tick or untick Options>Show Micro Frequency Adjust Sliders), simply tick the 'BPM Lock' checkbox of the channel, then untick it. Doing this will display the Micro Frequency slider on the channel.

3) When I ran FlexibeatzII on an Acer Aspire netbook using its internal audio device, I found the Pan sliders did not pan, and even slight adjustment of the From and To sliders on the Sound Edit panel resulted in the sound in that channel disappearing altogether or only a fragment of the sound being played. These issues disappeared when I routed the sound of FlexibeatzII through an external USB audio interface

4) When you are copying and pasting patterns around to create a song, under certain circumstances the application can freeze. Save your work to a .FB file as you go along (see Save and Open section) so that if this does ever happen to you, you can easily recover

5) The tempo of your sequence can slow down or fluctuate if you run FlexibeatzII on a heavily loaded system. So, it is recommended to close down as many unnecessary applications as possible when you work with FlexibeatzII

Save and Open

File>Save opens up a dialog box where you can specify name of the FlexiBeatzII file to save all the settings of your current session. The file gets saved with a .FB extension. To open your saved session, simply press File>Open and select the appropriate .FB file.

Beat Slicer

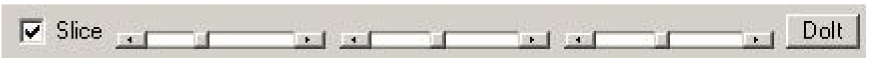
Beat Slicer is available only on the ‘Sound Edit’ panel of the first channel

The Beat Slicer detects distinct sounds within a loop. It then marks the start and stop of each sound on the Time View waveform of the loop.

Once the waveform is thus marked up, you can audition each separate sound, and make adjustments to the Beat Slicer parameters if you wish to further tweak the points where the slices occur.

Once satisfied, you can export each sound slice with a click of a button into the data folder, from where you can select the slices into separate channels and sequence them like any other sound in the data folder

Start first with loading your loop. In the adjacent screenshot, a loop wav file has been loaded into Channel 1 and a Note has been ticked so you can hear the loop as the measure plays over and over

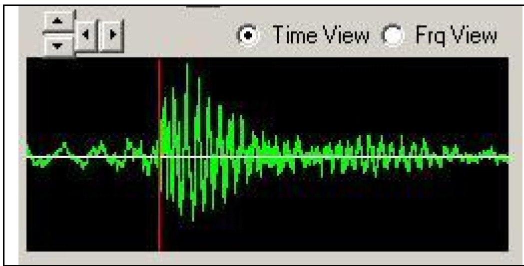


To separate out the sounds in the loop, the Beat Slicer first bandpass filters your loop. The first slider from the left lets you vary the bandwidth of the bandpass filter.

It then chops the filtered loop into small chunks. The middle slider lets you vary the chunk size (and therefore how many chunks the filtered loop gets broken up into).

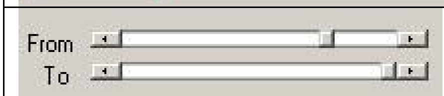
Finally, the RMS amplitude of each chunk is computed, and any chunk with an amplitude X times greater than the amplitude of a previous chunk is marked as the onset of a distinct sound. The last slider (third from the left) lets you vary the value of X.

Once you have set your sliders, tick the “Slice” checkbox, then press the “Render” button



Because you have ticked the ‘Slice’ checkbox, pressing ‘Render’ activates the Beat Slicer, which marks up the Time View representation of your loop waveform, with where the slices will be applied.

The red lines indicate each slice point



While FlexibeatzII is playing your loaded loop, you can hear what each slice will sound like, by clicking on the waveform area between each slice point on the Time View graph. This is because when you click the waveform area, the ‘From and ‘To’ playback sliders automatically adjust to the slice section, as shown by the adjacent screenshot

Once you are happy with how each slice will sound, press the “DoIt” button. This creates new wav files in the data folder called “SlicedN” corresponding to each marked-up slice, where the value N simply increments according to the number of

	slices created. The screenshot shows each new sliced sound opened up in a separate channel, ready for sequencing

Volume automation

Let us say we have a sound in Channel 5, we have 12 Measures in a pattern and we want to record our manipulation of the Lvl slider from start of Measure 2 to end of Measure 3

In order to do this, we open Options>Set Sound Cutoff…whereupon the ‘Cutoff Group’ panel opens.

Go to the Channel 5 group, in the ‘From’ textbox enter 64 (note pertaining to start of Measure 2) and in the ‘To’ textbox enter 128 (note pertaining to end of Measure 3). Tick the ‘Record’ checkbox in the Channel 5 group. We are now primed for recording the Lvl slider movement.

Run your sequence, keep an eye on the ‘Counters’ reading and make sure you move the Lvl slider of Channel 5 around during the 2nd and 3rd Measures.

Stop the sequence. Untick the ‘Record’ checkbox in the Channel 5 group. We now want to playback the Lvl slider movement, and if the ‘Record’ checkbox were to remain ticked during playback, we would record over our slider movements, so it is important the ‘Record’ checkbox is unticked. Now Run the sequence again, and we see the Lvl slider of Channel 5 automatically move during the 2nd and 3rd Measures in the same way as we had moved it.

OK now we want to paste the volume automation in the 2nd and 3rd Measures, to the 5th Measure.

The method to do this, is that in the Channel 5 group in the ‘Cutoff Group’ panel, we have to enter the From and To as the Source Measures we wish to copy. Then in the ‘To’ box following the Copy button, we enter a value – let us call it X, which represents the Measure we wish to copy the Source Measures to. Then we tick the ‘Record’ checkbox, and then press the ‘Copy’ button. This copies the Source Measures to position X. Now untick the ‘Record’ checkbox and Run your sequence. This is the formula to use to calculate the value of X:

X = The Measure you wish to paste TO – Number of Source Measures - 1

In our example, the Measure we wish to paste TO is 5. The Number of Source Measures is 2 (2nd and 3rd Measure). Therefore:

X = 5 - 2 -1 = 2. We convert this value to Notes: 2 x 32 = 64. This means we enter ‘64’ in the ‘To’ box following the Copy button

Now let’s say we want to paste the volume automation in the 2nd and 3rd Measures, to the 11th Measure.

The Measure we wish to paste TO is 11. The Number of Source Measures is 2 (2nd and 3rd Measure). Therefore:

X = 11 - 2 -1 = 8. We convert this value to Notes: 8 x 32 = 256. This means we enter ‘256’ in the ‘To’ box following the Copy button

Tracks you make with FlexiBeatzII

I am interested in hearing any commercially available tracks you create solely or partly with this application (by ‘commercially available’, I mean any music you make available to the public – whether you charge for it or not).

Also please email me, or let me know from where I can download both the .wav files and .FB files of any compositions you have created solely with this software, which you would like considered for inclusion as demo songs in future releases of the software. I acknowledge all contributions.

FlexiBeatzII Development

The majority of the application was developed in Melbourne, Australia

The algorithm which allows a pitch envelope to be applied to any wav sample was sketched out while on holiday in London and Oxford, and implemented in Paris. I think the ‘change of scenery’ really helped me in sorting that one out

The song ‘Welcome To Paris’ was assembled while travelling on Eurostar from London to Paris. The vocal snippets in the song were recordings made at stations and in the train with a Zoom H4 portable recorder (handy device, that one)

Finishing touches to the application and some more demo songs showcasing various features of the application were made in Sydney, Australia.

Acknowledgments

Bram Bos: For bringing us ‘Hammerhead’

Hakan ‘Master Zap’ Andersson: For bringing us ‘Stomper’

CVMichael: For DirectSound implementation tips

Professor Arun Chandra: For Reverb algorithm tips

Bram De Jong: For his waveshaping algorithm contributions to musicdsp.org. Some effects in FlexibeatzII were based on these. And thanks also go to Bram for founding Freesound.org, from where I sourced many of the sound files used during development of FlexibeatzII, and in the creation of demo songs

Borogove on KVR Audio for his hints for the beat slicing algorithm