Introduction
Are you one of the many desktop musicians who rely only on the standard General MIDI instruments supplied with your Sound Blaster Live!, Sound Blaster Audigy, or Sound Blaster X-Fi series audio card, or those SoundFont banks made by others?

Do you find these instruments a little too familiar as everyone uses them? Would you like to spend a little time making some more "personal" instruments yourself?

If your answer is yes, you are probably wondering how you can make these customized instruments, or perhaps you do not even know you can actually make them yourself?

Well, you can!

In these next few lessons, I will reveal the secrets of Vienna SoundFont Studio and help you along with your first attempt at sound designing. Vienna SoundFont Studio can be rather confusing at first glance, but you'll quickly learn how powerful and easy to use the program actually is. Even if you are an experienced user of Vienna SoundFont Studio, I believe you will still learn something new. This tutorial will give you a step-by-step guide on how to create your own SoundFont file, also know as an SF2 file.

Note: this is not an attempt to teach you everything there is to know about Vienna SoundFont Studio. What we hope to do here is to show you how you can effectively create your own SoundFont file in the shortest possible time, while providing you with a better understanding of the essential basics.

To follow the examples given in the articles, all you need is a PC equipped with a Sound Blaster Live!, Sound Blaster Audigy, or Sound Blaster X-Fi series audio card and at least 128 MB of system memory, lots of patience, and the willpower to learn. The examples in these lessons revolve around a "SawPad" synth sample (**sawpad.wav** [125kb]) and three percussion samples (**snare.wav** [21.7kb], **kick.wav** [16.6kb], **hihat.wav** [24.2 kb]) that you can download for use with the examples. It is recommended that you use these samples as it will make the examples clearer and easier to understand.

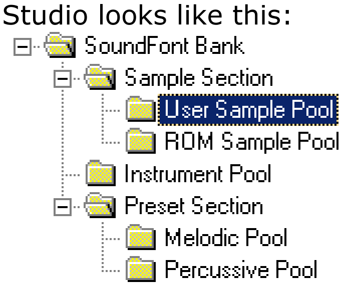
Good luck!

Samples: Importing The Raw Samples
To make a SoundFont bank, you will need a sample. More likely than not, you will actually need more than one sample. So let's assume you make one synth instrument and a percussion kit with four drums in it. To get started, you will need to lead the four needed samples into Vienna SoundFont Studio. The file format of the samples must be Microsoft's .wav format, Window's standard sound file format. .Wav files can be created using either a media recorder (Creative Media Source), or sample editors (such as Creative Wave Studio, Sony Sound Forge, or Steinberg WaveLab), dumped from sample CDs, or even downloaded from the internet. The process of creating .wav files will not be covered in this article.

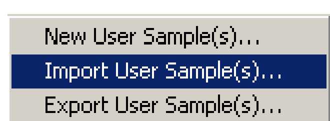
To get started, download the four samples needed for the examples here (ignore this if you have already downloaded these files from the previous lesson):

- **Sawpad.wav** (125kb)
- **Snare.wav** (21.7kb)
- **Kick.wav** (16.6kb)
- **Hihat.wav** (24.2kb)

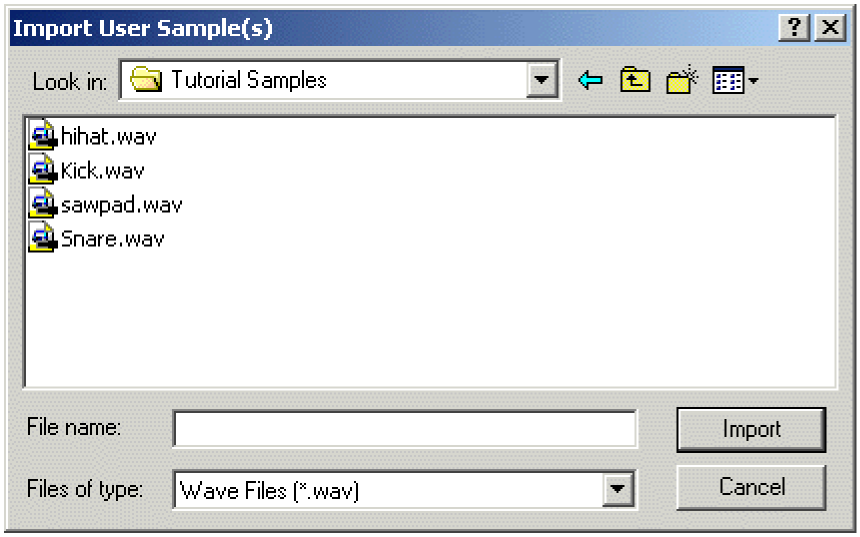
The SoundFont Tree View
The .wav files are first imported into Vienna SoundFont Studio. The SoundFont Tree Window in Vienna SoundFont Studio looks like this:



All samples imported into Vienna SoundFont Studio are located in the "User Sample Pool" folder in the SoundFont Tree Window. If you right-click this folder, you will get a context-sensitive menu with the options:



To import your samples, click the "Import User Sample(s)" menu item. From here, you will be able to navigate your drives and select the .wav files you want to import:



Find the four samples you downloaded from this page, mark them, and click "Import." You have now imported your .wav files into Vienna SoundFont Studio and should see the samples in the tree list.

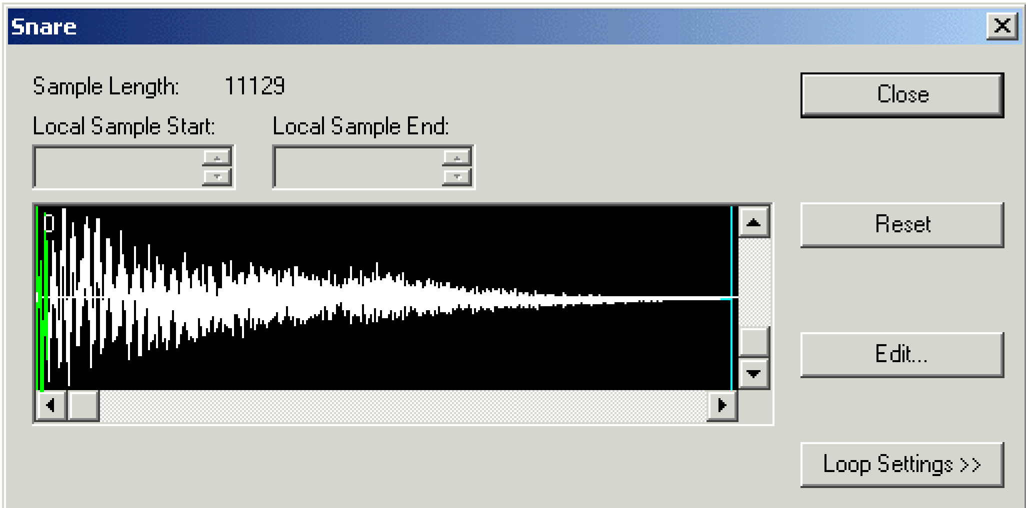
Global Sample Looping

To perfectly make a looped sample, I would recommend you use an advanced sample editor such as Sony Sound Forge, or something similar that has a cross-fading function. Vienna will successfully import the loop-points saved with the .wav file created by such programs.

Vienna SoundFont Studio will also let you loop a sample, but you will not be able to change the raw sample data which can only be done through a cross-fade operation. Vienna will only allow you to set the End/Start loop markers in the raw sample data which makes looping a little difficult (but it can be done anyway).

In general, looping is about making a portion of the raw sample data repeat itself again and again, resulting in a sustained sound that goes on and on.

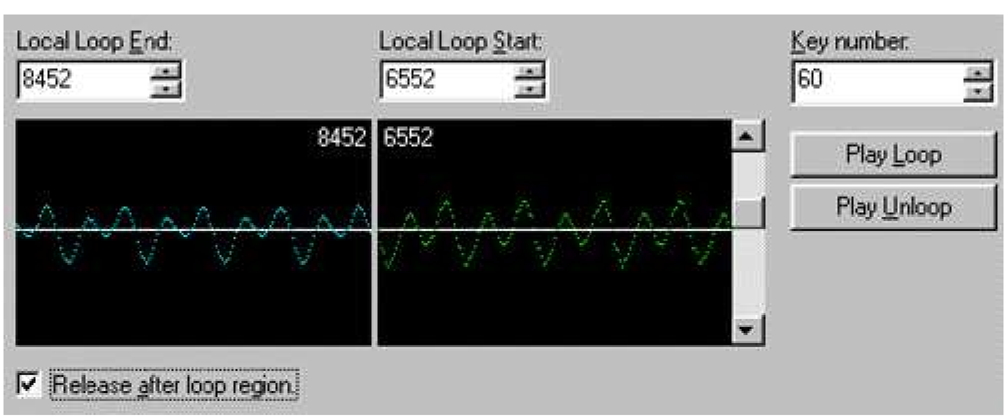
If you enter the looping screen, you will see the following (enter the looping screen by left double-clicking the relevant sample in the Sample Pool):



The green marker is the "Global Loop-Start" marker while the cyan is the "Global Loop-End" marker.

The sample data in between these two markers is the sample data that will be repeated over and over again. The green and cyan loop markers themselves can be dragged by the mouse for coarse loop marking, while fine-tuning of the markers can be done from the extra screen accessible from "Loop Settings" (explained below). The reason the loop markers are called "Global" loop markers is explained in the section on Local Sample Looping.

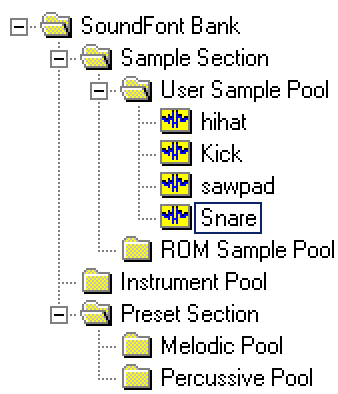
Normally, the coarse view of the sample is not good enough to make a good looping. You will need to be able to see a representation of every single sample "dot" in the sample data to make the loop perfect. To zoom in on the sample data, you can use the sliders beneath/beside the sample data view. It would of course be best if we could actually see the "splicing" point of the End/Start loop markers. To do this, click "Loop Settings." This will open up an additional area below the sample display. The extra space will show the "splicing" point of the loop markers and also every sample as needed (1:1 view). The loop markers can be changed in actual single samples here, which is essential in making a good loop. It is important that the two markers are located on two samples ("dots") that are very close to each other in value or you will hear an audible "click" as the loop repeats itself. Also, the sampled material has to be smooth. Here is how the extra space looks:



Try to change the SawPad sample's loop points and hear what happens. The SawPad sample was looped in Sound Forge using the CrossFade option which makes the loop extremely smooth (more or less perfect, actually). Don't count on your loops to be as good as the SawPad loop unless you are very lucky. The skill at making a good loop is an art in itself and requires keen analysis and experience (along with the proper tools, of course). After you have experimented with the loop markers, click "Reset" to set the markers back to their initial positions.

Remember that if you make your loops with a full featured sample editor (such as Sound Forge), you do not need to worry about looping in Vienna SoundFont Studio. The only thing you have to do is enable the looping.

Instruments
Once you have imported the samples you need and set their global looping (if necessary), you can now create the instrument. Let's begin with the synth sample named "**sawpad**," one of those you've just imported. To make an instrument of the sample, consult the SoundFont Tree window again:

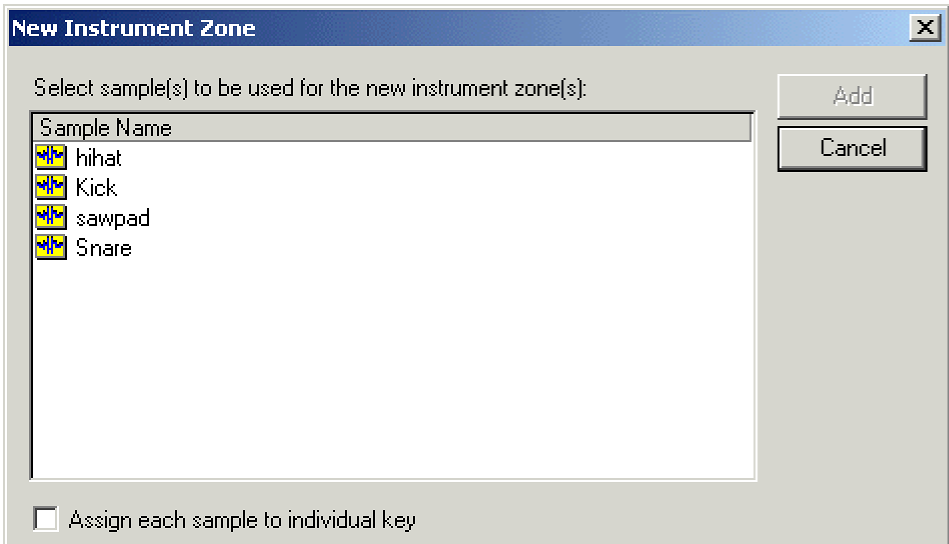


The instruments are all accessed through the "Instrument Pool" folder in the SoundFont Tree view. If you right-click this folder, you will get another context-sensitive menu with the option to select "New Instrument":

Click "New Instrument" and a new screen will pop up to prompt you for the name of the instrument you are creating. For the example, type in the name "SawPad - Dry" and click "OK":

This will bring you to the actual screen which allows you to choose which of the previously imported samples you would like to use in the instrument. All the samples that you have imported earlier will be shown here. You can mark several samples if you want the instrument to use more than one sample (as with a drum-kit for example), or you can mark just one sample if this is enough (as with most instruments).

The screen looks like this:



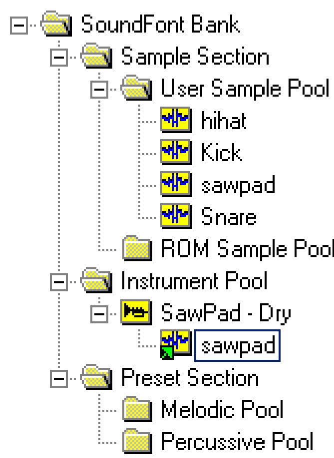
If you are making a drum-kit, you should check the "Assign each sample to individual key" box. This is a very useful feature - it opens up yet another screen that prompts you for the keyboard Key-Number of every drum sample you are adding. The Key-Number can be seen at the lower-right corner of Vienna when you move the mouse pointer across the graphical keyboard of Vienna.

When you have selected the samples you want to use (in this case, only the SawPad sample), click "Add" to create the instrument. After clicking "Add," you should see the newly-created instrument in the SoundFont Tree view along with all the samples you've included in the instrument directly beneath it. Such a sample is now called an "Instrument Zone."

You should be able to play the instrument by hitting a key on your keyboard. The sound will be that of the raw, unmodified "SawPad.wav" file you imported.

You should be able to play the Instrument by hitting a key on your keyboard. If you do not have a keyboard connected to your computer via a MIDI I/O device (usually the joystick port with a MIDI adapter), you can left mouse click on the user interface in Vienna that looks like a keyboard. The sound will be that of the raw, unmodified "SawPad.wav" file you imported.

The SoundFont Tree view should look like this now:



If you right-click this newly created instrument (the one with the trumpet icon), you will be given the option to add additional samples to this instrument just as when you created the instrument in the beginning.

You can also add something called a "Global Zone" this way. Any parameter set in a Global Zone will be the global setting for all the samples (Instrument Zone) within the instrument. This way, you could set a given parameter of all Instrument Zones by simply changing the Global Zone's parameter. Just remember that if you change the same parameter in an Instrument Zone, this will have the first priority and thus cancel the Global Zone's setting (for that particular Instrument Zone only, of course).

Local Sample Looping

As explained earlier, Global looping was something you did on the samples in the Sample Pool. Local looping is instead done on the individual Instrument Zones of an Instrument. You enter the Local Loop screen by left double-clicking the Instrument Zone for which you wish to change Local Looping. The changing of the Local loop markers is the same as with the Global Loop markers except that they are specific for that single Instrument Zone. Other Instrument Zones may have other Local Loop points even though they use the sampe sample. The only thing that all Instrument Zones have in common regarding the looping is that their Local Loop markers are set to the same as the used sample's Global Loop points. This will save you from having to set up the same loop marker positions each time you create an Instrument Zone that uses the same sample.

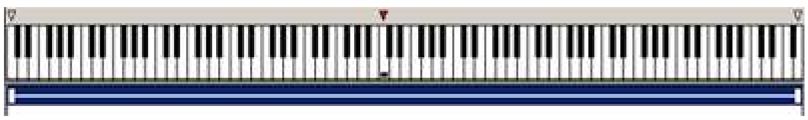
The only difference in the layout of the Global/Local looping screens is that the Local Looping screen has two additional check boxes:

and

The first check box lets you enable looping on the sample used by the Instrument Zone you are editing. The second check box lets you enable the "Release after loop region" feature which will play the sample data beyond the Loop-End marker when the key is released on the keyboard. This feature is not often used but there will come a time when you will need to use it. Remember that the later function is not available unless you have enabled the looping for the sample in the first place.

As you may have noticed, the instrument you created in the previous chapter (the "SawPad" synth) is abruptly stopped as it reaches the end of the raw sample data. It would be nice if the sample was in fact looped. Try and make the "SawPad" Instrument Zone loop the sample it uses. It should be possible for you to do this with the information from this section. The loop points have already been saved with the .wav file, so you should not need to set the Global Loop markers yourself.

Keyboard And Velocity Range
Highlighting the "SawPad" Instrument Zone in the SoundFont Tree view allows us to change its Keyboard and Velocity ranges. This can be done by dragging the low and high end of the Range markers located directly beneath the keyboard layout of Vienna SoundFont Studio:



Note the three arrows on top of the keyboard view. The two white arrows show the location of the low and high Key-Range, while the red arrow shows the "Root-Key" of the current Instrument Zone.

The Root-Key is the key at which the sample plays as originally sampled. The Root-Key needs to be the same as the sampled sound or the instrument will not play in tune with other instruments (that is, when you press the "C" on the keyboard, the instrument should of course play a "C" too). This means that if the raw sample you recorded is in fact a musical "D" and not a "C" as the red Root-Key arrow indicates above, then the sample's root-key is wrong. To change the root-key, just right-click on the key-range marker (the one highlighted in the illustration above) and a context-sensitive menu will appear with the following options:

Select "MIDI Set Root Key" and the following screen will appear:

Now play the proper Root-Key on your external keyboard. You will see the red arrow move accordingly to the location in the Keyboard View. You can also use this feature to set the Start and End Key-Range - just choose the "MIDI Set Range" menu item instead. If you do not have an external keyboard to do this, change the Root-Key using the "Properties...." menu item instead.

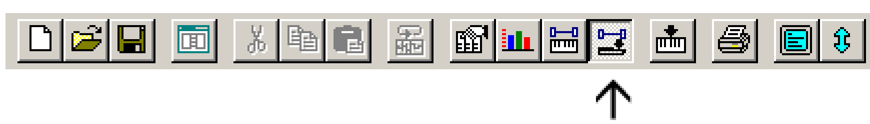
If your sample is still slightly out of tune after setting the Root-Key (the Root-Key is set in half-tone intervals), you can fine tune the Instrument Zone in the "Generator Parameters" view, but this I'll tell you about in a later section.

The Velocity Range
Vienna SoundFont Studio also allows you to change the "Velocity Range" of an Instrument Zone. Velocity is a measurement of how hard you play an external MIDI keyboard. If you play softly, the keyboard sends low velocity values to your computer and if you play hard, it sends high velocity values.

NOTE: To test the velocity range with the Vienna user interface, all you need to do is click on a different part of the note vertically. Within the Vienna user interface, if you left mouse click towards the top of the key the note will play softer and if you left mouse click the key towards the bottom it will play louder.

Setting the Velocity Range of an Instrument Zone will allow you to make it play only if the velocity values are in between two set values. Velocity is sent in vlues of between 0 and 127, with 0 being the softest and 127 the hardest. If you want the Instrument Zone only to play when pressing hard, you could set the Velocity Low Range marker to 64 and the Velocity High Range marker to 127. This will do the trick.

If you want to change the Velocity Range, you should switch the Keyboard View to the Velocity View by clicking the button which is shown depressed on the toolbar image below (to switch back to the Keyboard View again, click the button directly to the left of the Velocity View button):



You should now see the Velocity View instead of the Keyboard View:

Changing the Velocity Range is done in the same fashion as the Keyboard Range. Drag the Range markers' ends to adjust the Velocity Range of an Instrument Zone. The blue region shows the low Velocity values while the black region shows the High Velocity values.

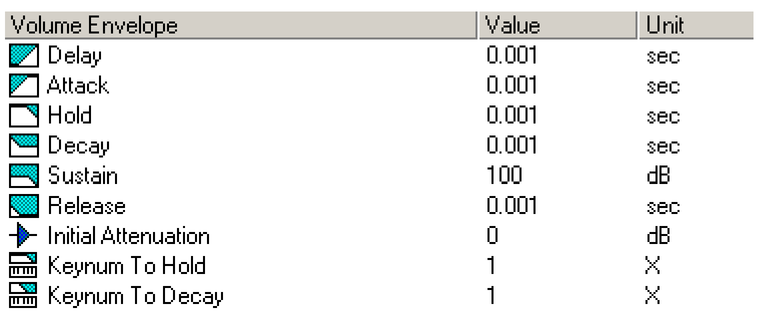
How do you make use of this feature? For example, if you have an acoustic drum sampled at, let's say, four different hits: low, middle, hard and very hard, you will be able to assign the four samples to their own Velocity Range, thus simulating the effect of the real instrument better. Just remember that the Keyboard Ranges of the four instruments must be set to the same range or they will not play on the same pressed key.

Now try and experiment a bit. Add a few Instrument Zones. Remove them again, and try to change the Key-Ranges, Velocity-Ranges and Root-Keys. Also try to make a drum-kit instrument with the three other samples you imported: "**Kick**," "**Snare**," and "**Hihat**." Every drum should be assigned to its own key on the keyboard.

After creating the drum-kit instrument, add one more sample to it (the "**Hihat**" sample again). When a sample has been imported into the sample pool, it is available to all instruments. In other words, the samples can be used more than once, which saves a lot of memory.

The Generator Panel, Part I
Exactly what is an "Envelope Generator?" Briefly stated, it is a curve that tells how a sound is modulated over time, that is, how a sound dynamically changes over time.

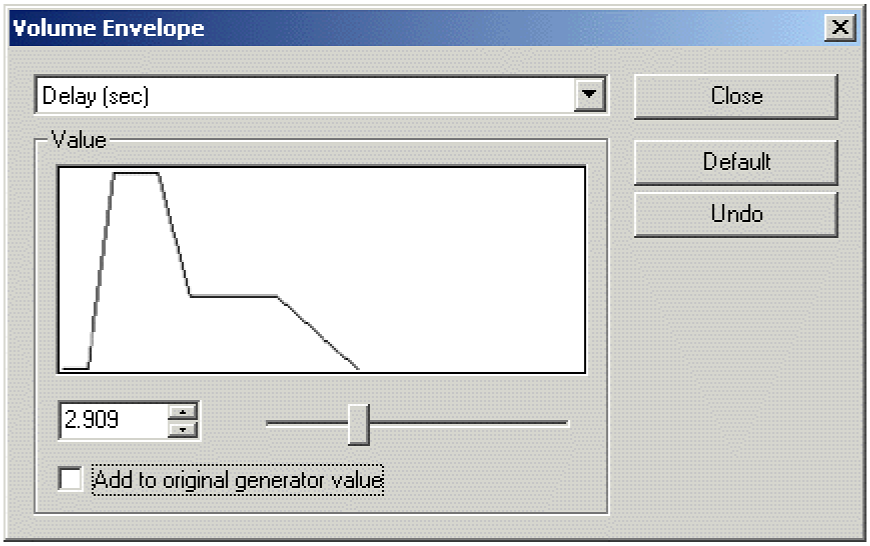
The Volume Envelope
The first Envelope we will look at is the "Volume Envelope Generator." This envelope can change the volume of a sound as it plays. The parameters for this envelope generator are found at the bottom of Vienna's screen and look like this (remember to highlight that particular Instrument Zone in the SoundFont Tree view for which you wish to change the Volume Envelope):



Volume Envelope	Value	Unit
Delay	0.001	sec
Attack	0.001	sec
Hold	0.001	sec
Decay	0.001	sec
Sustain	100	dB
Release	0.001	sec
Initial Attenuation	0	dB
Keynum To Hold	1	X
Keynum To Decay	1	X

When a sound is first initiated (by hitting a key on your keyboard), the volume will always start at "no sound" or zero, and it also ends at that value. The Volume Envelope then tells us how the volume will change over time from start to end.

When you edit the values of the Envelope Generators, you will see a graphical representation of the "curve." It looks like this (it shows the curve as a "whole" including all the envelope parameters):



Notice the different envelope phases: Delay, Attack, Hold, Decay, Sustain, and finally the Release phase. Here is an explanation of each parameter and phase in the Volume Envelope:

Delay
Before any change in volume starts, there is a delay; that is, no sound will be audible before the delay is finished (since the beginning volume is always zero). This delay can be set with the "Delay" parameter.

Attack
When the Delay is through; the volume starts to rise to its maximum and does so at a speed equal to the "Attack" parameter. The smaller the value, the faster the volume reaches its maximum. This is called the "Attack Phase." Remember that when the attack parameter is set to anything higher or equal to 0.007, the filter cut-off frequency (to be explained later) will be strangely altered to "soften" the sound when you press a key gently. It can be used for some nice effects though.

Hold
When the maximum volume has been reached through the Attack phase, the volume can be held there for a given period of time equal to the setting of the "Hold" parameter.

Sustain + Decay
When the "Hold" period is through, the volume will fade down until it reaches the value specified by the "Sustain" parameter. The speed at which it reaches the Sustain level is determined by the "Decay" parameter. When the volume reaches the value of the Sustain value at the end of the Decay phase, it will remain there for as long as you hold the key down. When you release the key, it will proceed to the "Release Phase."

Release
When the key is released on the keyboard, the "Release Phase" sets in. The Release Phase will take the volume from its current value (depends on when you release the

key) to zero volume. This Release Phase happens at a speed equal to the Release parameter.

It is important to note that the Release Phase will take its starting volume level from the volume's current value the moment you release the key (which is when the Release Phase will begin). This means that other Envelope parameters may be skipped (for example, if you release the key before the Attack Phase has ended, you would skip both the Hold, Decay and Sustain parameters; the Release Phase is always started when the key is released).

Initial Attentuation

Key Number To Hold/Decay
This parameter has a specialized use. It allows you to change the value of the Hold and Decay parameter from above according to the key pressed. If you play a key above or below the key #60 (the one marked with a dot in the Keyboard View), the Hold/Decay parameters will gradually change from their initial values. If the Hold/Decay Key-num value is greater than 1, the value of the Hold/Decay parameters will fall if a key is pressed above key #60, and it will rise if a key below key #60 is pressed.

If the Hold/Decay Key-num value is less than 1, the effect is of course reversed. A value of 1 is equal to no change of the Hold/Decay parameters (which is the default).

This effect can be used for many purposes - an example would be a piano. A piano-string in the high-end of a piano is physically shorter and will thus fade more quickly than a low-end piano-string. This effect can be simulated using this parameter.

The higher the values in these "Key-num" parameters, the more drastic the change will be. Experiment with them as it takes some time to get used to them.

The Modulation Envelope (for Filter and Pitch Envelopes)
It is also possible to change the Dynamic Filter and the Pitch of the sound being played in real-time. The way to do this is similar to the Volume Envelope. The Delay, Attack, Hold, Decay, Sustain and Release phases are exactly the same as with the Volume Envelope. You just need to know what a dynamic filter and pitch actually are in order to understand their uses. Remember that when filter enveloping is used, you cannot at the same time use pitch enveloping. The filter will somehow cancel out the pitch.

Here is the layout of the Modulation Envelope Generator:

Modulation Envelope	Value	Unit
Delay	0.001	sec
Attack	0.001	sec
Hold	0.001	sec
Decay	0.001	sec
Sustain	100	%
Release	0.001	sec
To Pitch	0	cents
To Filter Cutoff	0	cents
Keynum To Hold	1	X
Keynum To Decay	1	X

The Dynamic Low-Pass Filter (To Filter Cutoff)
A "Filter" is a piece of hardware that "filters" unwanted frequencies away. In short, frequencies that the human ear can hear are from about 5 to 20.000 KHz. Low frequencies are usually the deep sounds while high frequencies are the high sounds. Thus, a hi-hat has much high frequency material and a bass very little. The type of filter in the Sound Blaster Live! and Sound Blaster Audigy series audio cards is called a "Low-Pass" filter. It lets all frequencies below the "Cut-off Frequency" (selectable in the "Effects" Parameter section) pass through while higher frequencies will be removed. The best way for you to learn what the effect really sounds like is for you to hear it yourself.

The lower the value of the "To Filter Cut-off" parameter, the "duller" the sound will get during the Envelope phases. If the "To Filter Cut-off" value is a positive value, it will get "Brighter" during the Envelope phases (but this requires that you lower the "Cut-off" parameter value in the "Effects" parameter section as it is by default set to it's maximum of 8000 Hz - try 4000 Hz instead [the value can't exceed 8000 Hz]).

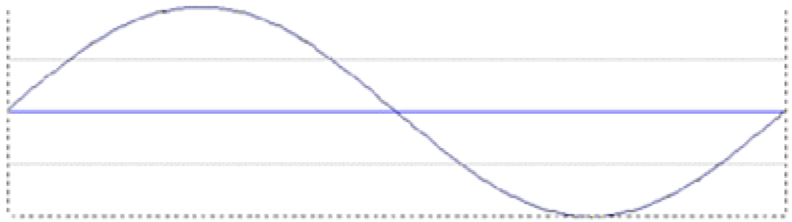
Experiment a bit with the filter envelope. The filter envelope can be used to create nice cass sounds of virtually any sample. It is also useful in combination with the "Filter Q" parameter from the "Effects" parameter section (described later) for making cool "Rubber" like sweeps for Dance and Techno music.

Dynamic Pitch Bending (To Pitch)
The pitch of the playing sound can also be changed dynamically as you play. The pitch will start off at the normal pitch +/- the value in the "To-Pitch" parameter. The value is measured in cents. 1 cent is exactly one 100th of a half-note. Thus, if you wanted the pitch to start out, let's say, an octave lower than normal, you would write -1200 cents (there are 12 half-notes to an octave). When the sound sets in, the Envelope Generator will change the pitch according to the Envelope parameters. This is quite useful for sound effects and certain instruments like, for example, a violin. A violin always plays a bit out of tune when the bow hits the string. The pitch bending scheme here can easily make that emulation by making a fast "slide" from the "out of tune" pitch to the "normal" pitch. Experiment a little with it.

As a test to see if you've got the idea of the Envelope generators, try and change the "extra" drum Instrument Zone you incorporated in the drum kit earlier. The drum-kit is lacking a closed hi-hat. This one could be created using the Volume Envelope on the extra copy of the open hi-hat.

The Generator Panel, Part II
Now what is an LFO? An LOFO stards for "Low Frequency Oscillator" and as in the

case of the Sound Blaster Live! and Sound Blaster Audigy series audio cards, a sine curve that changes very slowly over time (thus the word: Low fr..). A sine curve looks like this:



When you hit a key on the keyboard, the LFO starts to oscillate up and down according to the sine curve above. It starts out at zero, then goes positive, down to zero again, then negative, and back to zero. This procedure goes on and on.

This "Oscillating" can then be applied to the Filter, Pitch and Volume of an Instrument Zone. If the oscillating is applied to the Volume, you will get a Tremolo effect. If applied to Pitch, you get a Vibrato effect and lastly, if applied to the Filter, you get a WahWah effect.

The speed at which the oscillator swings up and down can be set by the LFO parameters and so can the level at which it oscillates (the "height" of the sine curve). To help you better understand each parameter, here is a thorough explanation of view of them:

Modulation LFO	Value	Unit
Delay	0.001	sec
Frequency	8.176	Hz
To Pitch	0	cents
To Filter Cutoff	0	cents
To Volume	0	dB

Vibrato LFO	Value	Unit
Delay	0.001	sec
Frequency	8.176	Hz
To Pitch	0	cents

As you can see, there are two LFOs available within the Sound Blaster Live! and Sound Blaster Audigy series audio cards. The Vibrato LFO (the lower one in the illustration above) is intended only for manipulating the pitch (Vibrato). The top one is meant for manipulating Pitch, Filter, and Volume. It is important to know that each of the three parameters assigned to the Modulation LFO will oscillate at the same speed (frequency). The three parameters can have their own "Level" or "Height," though.

Delay
This is like with the Envelope Generators. You set an initial delay before the oscillator will begin modulating anything.

Frequency
Defines the frequency or speed at which the LFO will oscillate. Frequency is measured in Hz which lets you decide how many times the sine curve will repeat itself within one second. This means that a frequency of 1 Hz will oscillate through the sine curve in one second. A frequency of 2 Hz will oscillate through the sine curve two times a second while a frequency of 44.000 Hz will oscillate through the sine curve 44.000 times per second! Just remember that it is a Low Frequency Oscillator. The frequency setting cannot exceed more than 10 Hz (which is a fair amount anyway).

To Pitch:
Here you set the level of the Oscillating Pitch (Vibrato). The level measures how "wild" the vibrato will be. Normally a setting of 10 to 30 is adequate for a nice instrument vibrato; higher settings will be too wild and are intended only for wild sound effects. The parameter is measured in cents like in the Pitch Envelope. Thus, you can calculate the "swing" of the pitch in half-notes by multiplying by 100. For example, if you want the pitch to swing an octave up and down, key in "1200" in the parameter (an octave is 12 keys or half-notes long).

If the parameter is set to a negative value, the sine curve is reversed. This means that the curve does not start positive but negative instead. You may think that this does not matter, but it does in some cases.

To Filter Cutoff:
This parameter sets the level of filter modulation as with the pitch. The higher the value, the more WahWah effect is produced. Negative values again reverse the sine curve.

To Volume:
This parameter is identical in function to the "To Pitch" and "To Filter Cut-off" parameters except that it changes the modulation depth of the volume instead, creating a Tremolo effect.

Try to make some wild sound effects using these LFO parameters in combination. Some rather strange results are actually possible.

The Generator Panel, Part III
The "Effects" and "Pitch" parameters in the Generator Parameter section let you add Tuning, Chorus, Reverb, Cut-off Filter, etc. to the Instrument Zone you are editing. The effects will be applied to the sound as it plays, allowing for rich and beautiful sound textures. The "Effects" and "Pitch" parameters look like this:

Pitch	Value	Unit
Coarse Tune	0	semitones
Fine Tune	0	cents
Scale Tune	100	cents

Effects	Value	Unit
Filter Q	0	dB
Filter Cutoff	20000	Hz
Reverb	0	%
Chorus	0	%
Pan	0	%

The Pitch Parameters:
Coarse Tune, Fine Tune and Scale Tune
The "Pitch" Parameters let you fine-tune the Instrument Zone. The "Coarse Tune" parameter tunes in half-notes, the "Fine Tune" in Cents (which is a 100th of a half-note) and the "Scale Tune" lets you decide how many cents a half-note actually is (100 by default).

The Effects Parameters:
Filter Q, Filter Cut-off, Reverb, Chorus and Pan
The "Effects" parameters consist of five parameters that enable you to add dynamic effects to the sound. The five parameters are explained in depth below:

Filter Q
The chosen frequency Cut-off point in the "Filter Cut-off" parameter (see below) can be enhanced with the "Filter Q" parameter. If, for example, the freqency Cut-off point is at 4000 Hz, then any frequencies in the sound at 4000 Hz will be enhanced (amplified) by the "Filter Q" parameter.

The "Filter Q" parameter can lead to interesting sounds if you dynamically change (real-time) the Filter Cut-off Frequency as the sound is playing (many Techno instruments use this feature to a great extent). This can be done by using either a Filter Envelope or a Filter LFO modulation, or even better, by changing MIDI controllers in your sequencer real-time.

Filter Cut-off
This parameter lets you set a frequency point at which every frequency higher than this point will be removed. For example, if this parameter is set to 4000 Hz, then any frequencies in the playing sound which are higher than 4000 Hz will be removed. The lower the value, the duller the sound gets. In combination with the Filter Q parameter that enhances frequencies around the Filter Cut-off point, interesting sounds can be made. This type of filter is called a "Low-pass Filter" in technical terms. It lets the low frequencies pass and removes the high ones.

Reverb:
The Sound Blaster Live! and Sound Blaster Audigy series audio cards incorporate a hardware effects engine that allows you to add Reverb and Chorus effects directly to your sounds. The Reverb simulates room reflections in Halls, Rooms, etc. The Reverb type is set using the Environmental Audio section of the Sound Blaster Live! Audio H.Q. application. Or for users of the Sound Blaster Audigy series audio cards, the Audigy EAX Control Panel can be used to adjust the reverb type. The amount of this

chosen reverb is selectable for each Instrument Zone in Vienna SoundFont Studio. The "Reverb" parameter is the one you use to set the Reverb Depth of the Instrument Zone you are editing. A mono sample will also become a "stereo" sample when Reverb is applied to the sound.

Chorus:
The "Chorus" parameter is set in the same fashion as the "Reverb" parameter. Chorus is an effect which simulates more voices playing together. Chorus adds "richness" and "warmth" to a sound. It also makes a "stereo" sound of a mono sample. Change the parameter and listen for yourself.

Pan:
The "Pan" parameter lets you place the Instrument Zone in the stereo perspective. A negative value places the sound more to the left while positive values will place the sound to the right. It does not matter if the sound is a mono sound; the sound will be placed in the stereo perspective anyway.

Try changing the Effects parameters yourself and see what happens.

Making the Final Preset
Having created the Insrument, the final step now is to make a "Preset." In the same way that instruments are made up of samples (all the samples in an Instrument make up an Instrument Zone), a Preset is made up of Instruments (all the Instruments in a Preset make up a Preset Zone).

If you right-click the "Melodic Pool" folder in the SoundFont Tree view, you get a context-sensitive menu which allows you to select a New Melodic Preset:

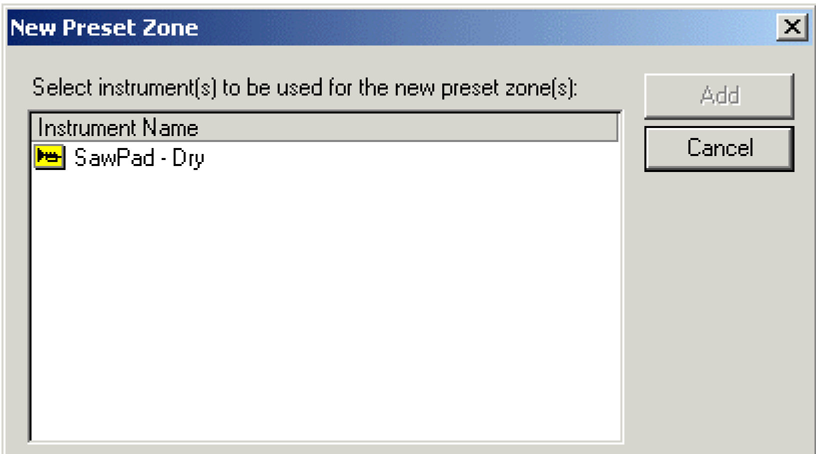
Select "New Melodic Preset" and a new screen appears:

Here you can key in the name of the Preset along with its Bank and Preset number. The Bank/Preset number can be anything between 0 and 127.

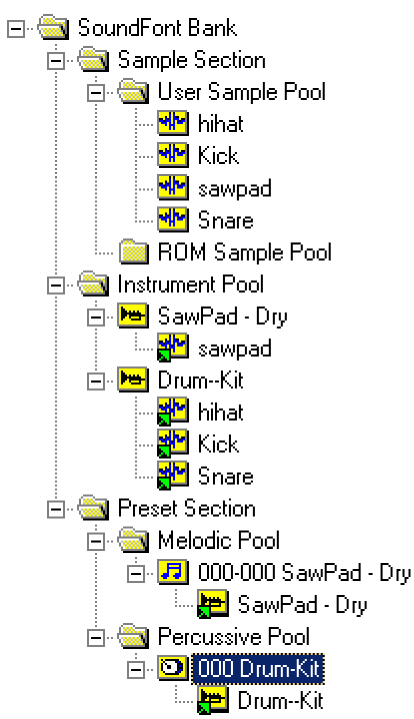
The standard MIDI language does not allow for more than 128 different instruments, so when more are needed, you usually apply to "Bank Change" message through MIDI to a synthesizer. This lets the synthesizer choose another "Bank" of 128 instruments, thus allowing 128 Banks or 128 Presets of different instruments, or 16,384 different Presets in the case of the Sound Blaster Live! or the Sound Blaster Audigy series audio cards.

When Vienna opens the above dialog box, it always does a search for empty space in the Bank and Preset positions (that is, a space not already occupied by another Preset). It then writes this location as the default Bank/Preset number, saving you from doing the search yourself (unless you want a specific Bank/Preset number for the Preset you are creating). Type in the name of the preset "SawPad - Dry" in the name field and click "OK."

Now you get a screen that allows you to choose what instruments to use in the Preset you are creating:



When you have chosen the "SawPad - Dry" Instrument, click "Add" to create the Preset. You now have the almost-finished Preset in the SoundFont Tree view, with all the used instruments directly beneath it (this is called a "Preset Zone"):



More Preset Zones can be added just like with the Instrument Zones - just right-click the Preset's name (the one with the musical note icon) and choose "New Zone." It works just like with the Instrument Zones explained earlier. You can also add a Global Zone if you want to.

Now that you have placed the Sawpad Preset, you only need to place the Drum-kit. The Drum-kit you created should not be put in the "Melodic Pool" folder but rather in the "Percussive Pool" folder instead. It is done in exactly the same way as with the "Melodic Pool" except that you do not choose a Bank number. You only select a Preset number (Vienna finds an empty preset for you). This gives you 128 possible drum-kits in one SoundFont file which should be more than enough.

You can save your finished SoundFont file now - it's done!

Using The SoundFont Bank
Now that you have mastered the creation of a SoundFont bank, all you have to do is learn how to use it with your sequencer. There are many different sequencers on the market today - to go through all of them would be a daring task.

Two prominent sequencers on the market today, Steinberg Cubase SX and Cakewalk Sonar both include SoundFont bank management tools within their respective applications. It’s best to refer to the user manual of your sequencer on how to load and manager SoundFont banks.

Conclusion
Well, is this it? Is this all there is to learn about designing sounds and instruments in Vienna SoundFont Studio?

No. Definitely not! There is still a lot to learn. Not all aspects of Vienna SoundFont Studio have been covered in this tutorial. It’s now up to you to explore the program further. You'll soon find that there's so many clever ways of using Vienna which can lead to interesting sounds and effects.

Another tip for any budding musician: get yourself a decent wave editor like Sony Sound Forge, Adobe Audition, or Steinberg Wavelab. One of these editors, when used with Vienna SoundFont Studio, can sometimes create miracles. Starting with the highest quality sampled materials is just as important as mastering Vienna SoundFont Studio. With just practice and determination, you will be amazed at what you can do with a combination of such programs.