



## The Wersi Sound Factory Master Classes

### Master Class 2 : December 2012 : The Samples Tab Part 2

We have already looked at some parts of the Samples screen: the list of Layers on the left; parts of the tables headed Volume and Pitch; and we're leaving the Play-Mode table until much later. We now need to complete our overview of this screen by looking at the remaining parts of the Volume and Pitch tables as well as have a first look at the Sample Edit and the Sample Delay tables.

#### The Volume Table

'Pan Key' in  
the Volume  
table

The Pan Key function is the only part of this table which now needs an explanation. The default value is 0 %: whether it displays as -0 % or +0 % doesn't matter; it's the same result. So what is that result?

It's an extension of the Panorama feature but instead of applying to the whole sound it enables individual samples to produce a stereo sound spread according to the keys played. The values can range from -100 % through to +100 %.

Positive values assign low pitched sounds to the left and high pitched sounds to the right, just as you'd expect to hear from playing a piano, for example. The higher the % number, the further apart will be the spread.

Needless to say, negative values reverse that effect: low pitched sounds are heard from the right and high pitched sounds from the left. The best way to audition this feature is to use headphones.

I think it stands to reason that this feature could conflict with a Selectors Panorama setting of anything other than <C>. It's worth experimenting, though.

#### The Pitch Table

'Cent' in the  
Pitch table

The Cent function provides a range of values from +50 to -50. +50 raises the pitch by a quarter tone (half of a semitone) and -50 lowers the pitch by a quarter tone (half of a semitone). Subtle adjustments here can remove any beating which may occur between the sounds of different sample layers should those layers not be precisely in tune, as well a help fatten up two or more samples as very few orchestral instruments are ever really in tune with each other.

'Key' in the  
Pitch table

The Key function provides a range of values from -100.00 to +300.00. Here is what it does at certain reference values:

-100.00 ; the pitch sequence of the keyboard is reversed. Middle C is normal pitch but the keys below rise and the keys above fall in pitch, exactly the opposite of 'normal'.

-100.00 to -2.00 ; various tunings occur, in reverse note order to 'normal'.

-0.00 ; all the keys play the same note. This is useful for a non-tuned percussion sound.

+2.00 to +98.00 ; various tunings occur in 'normal' note order.

+100.00 ; all the keys play as 'normal'. It is best to use this value for all 'normal' use.

+102.00 to +300.00 ; various tunings occur up to +300.00, giving some strange effects.

It is possible to compress or stretch the sample(s) tuning across the keyboard. This gives us access to some 'micro-tonal tunings', for example Ethnic instruments that don't normally have a 'western' style tuning.

## The Sample Edit Table

Wave Start  
in the  
Sample Edit  
table

When you play any sample, the sound starts playing immediately you press a key, right from the start of its wave. Now imagine we could make that sound begin from further into its wave so the start of the wave was missing. It would be like coming in on a conversation after the first word had been spoken. This is what the Wave Start function does. What's the point of this, then?

To demonstrate this we're going to use the Bagpipe sound [091-000-066]. Ian has designed a modification to this sound, one part of which we're also going to look at when we concentrate on the LFO/Matrix section. For now, open Sound Factory and load this Bagpipe sound. Your Sample screen should look like this.

The Sample  
screen for  
the Bagpipe  
sound



Notice there is just one sample, called Bag Pipe. This is actually a composite layer containing more than one (hidden) sample and we cannot separate them. One is set to sound below D3 and the other to sound above Eb3. The lower sound provides the drone of the bagpipe and the upper sound provides the chanter which is used for the melody line. A feature of this particular chanter sample is a pronounced turn at the start of each note. There are two ways of adjusting this turn:

- To reduce the amount of the turn: this is what we're going to do here using the Wave Start function.
- To enable the turn to occur according to the amount of dynamic attack given to the key being pressed, thus enabling the player to control the feature. This can be achieved here using the Velocity function in the Sample Edit table. It can also be a function of the Matrix but that will have to wait until we're looking at the LFO/Matrix section.

In the audio demonstration of the final result of this editing, both of the above changes can be heard.

Using the  
Wave Start  
function to  
remove the  
turn at the  
start of  
each Bagpipe  
chanter note

Let's remove the turn completely then, using the Wave Start function. Remember, we're removing it from being heard, not cutting it out of the sample totally: it's still there so we can revive it under dynamic control.

The values of Wave Start range from 0 (no effect) to 100 (maximum effect). As you raise the value above zero, so the starting point at which you hear the sample is moved away from the sample beginning into the rest of the sound. The sound still starts as soon as you press a key but the sound you first hear is a little way into the sample, thus removing the beginning from being heard: and it's at the beginning of the bagpipe chanter that the turn occurs. Imagine the sample as an LP record. 'Wave Start' is the arm and stylus of the record player. You can play any record of your choice at any time, from start to finish. Or by moving the arm (changing 'Wave Start'), you can play your favourite bits or tracks from that record.

**Using the Velocity function to place the turn under Dynamic control**

So, by trial and error, adjusting the Wave Start value and playing a key enables you to hear the turn gradually being removed from the start of the sound. A value of 74 % just about 'removes' the turn. Once you're satisfied you can no longer hear the turn, save this new sound as Bagpipe SF1, for example. You can, of course, call it what you like, but I tend to identify my User Sounds made with Sound Factory by using the letters 'SF' in their name.

➔ Now load your new Bagpipe sound into Sound Factory and we'll use the Velocity function of the Sample Edit table to bring the turn back under dynamic control. Now there is an interaction between Wave Start and Velocity. A Wave Start value of +100 % and a Velocity value of +50% provides a workable dynamic. Negative Velocity values reverse the overall effect with a dynamic playing removing the turn and legato playing allowing it to be heard.

**The Random function**

The final function in the Wave Start table is Random. This has a range of values from 0 % to +100 %. It randomly selects a different Wave Start value with each key pressed, the higher its value the wider the range of Wave Start values it uses. It's better to leave Dynamic unticked in Selectors to allow Random to work, otherwise it can be overpowered by dynamic playing. Still set Wave Start at +100% and Velocity at +50%.

**Further examples of using the Wave Start function**

You can try all this out on other sounds which have a pronounced effect at the start, such as picked bass or electronic organ with key click (though be aware that Sound Factory does not work on the OX7 module, nor the B4 VST). The Appendix on page 6 has an expanded explanation of the Wave Start, Velocity and Random functions, provided by Ian.

One aspect of sound editing we shall cover later is the relationship between what we do in Sound Factory and the six parameters available in Sound Controls. Here's an example. Using Quickload, load the sound Metal Tyne [081-000-039] into Selector 1 of the Upper Manual of a Standard Preset. Look in Sound Controls and you should see Wave Start listed to adjust. As you increase the value of Wave Start and play the keyboard the sound is assigned to, the Tyne will gradually be (re)moved from what you are hearing at each key stroke. Now you know how this is done and the meaning of the Wave Start function in Sound Controls. You could achieve the same effect by editing the Wave Start value in SF.

**The Sample Delay Table**

**The Sample Delay Table**

The first 'teaser' audio demonstration placed on the website to show some of the results which can be achieved with Sound Factory was two Harp Arpeggios, one ascending and the other descending, each arpeggio consisting of nine notes triggered by playing just one key. The Sample Delay Table contains the functions which allow that to be done. To show the basic process, there is a Wersi sound which has this built in and it is the 1 Finger Plucker [082-000-004]. This is the one with the ascending notes. In Sound Controls, by making the Major/Minor value 63 or below the arpeggio produced is major: values of 64 and above (default) produce a minor arpeggio. Load this sound into Sound Factory.

Here is a screenshot of the Sample screen of 1 Finger Plucker.

**1 Finger Plucker Sample Screen**



**Explanation of how the 1 Finger Plucker Sound works**

The list of layers contains six copies of the sample Nylon Finger 1 Hard followed by two Fret Noise samples and one Effect, inaccessible to the user. Touch each of the Nylon Finger samples in turn and look at what changes occur in the Octave and Semi values in the Pitch table and what changes occur in the Sample Delay table bottom right.

Looking at the Pitch table first, notice each layer increases in pitch shown by the Semi or Octave values changing. The Major/Minor feature seems to be built in to that Sound Controls function only.

Looking next at the Sample Delay table, there is a delay to the start of the sound in all except Layer 1. The Mode indicates Tempo and the Delay ms is not in milliseconds but is a fraction of a whole beat (crochet), eg 1/16. Layer 4 has a Delay of 1/8D which is 1/8 of a dotted crotchet so is a longer delay than 1/8, the delay of the preceding Layer 3. That ensures Layer 4 plays after Layer 3. Try muting various Layers, play a note on the Upper Manual and notice what is happening. Notice Layers 5 and 6 have the same Delay value and so play as a two-note chord.

If you wish to make a similar arpeggio sound for another instrument, then you can replace all the Nylon Finger samples with another instrument and delete the Fret Noise samples. Then you would also have the Major/Minor feature of Sound Controls available to use. Alternatively you could start by loading the actual sound you wished to use and copy the samples into additional Layers for as many notes in the arpeggio as you required, up to a maximum of ten. That way you would retain the Sound Control features appropriate to the instrument of that sound. Major/Minor can then be achieved according to the Semi value for the sample which plays the third interval of the arpeggio as this is always one Semitone lower for a minor arpeggio.

**Explanation of how to convert the 1 Finger Plucker Sound into the Harp Ascending Arpeggio**

Here is how the 1 Finger Plucker Sound was converted into the Harp Ascending Arpeggio. Just use the Filter-Map, Envelope and LFO/Matrix Screen tables as 'recipes' for now.

Samples 1-6 : Change to Harp (Wersi 1 – Harp). Delete Fret Noise Samples. Copy 6 to 7, 8 and 9.

First Screen : Vol +3.2 dB, Reverb 1 -0.2 dB, Remainder at Off. Octave 0.

Sound Controls : MajorMinor 0%; Width 50%; Tone 100%; EQ 60%; Frets Level 0%; Release 53%

Sample Screen (for all Layers)

Volume		Pitch		Sample Edit		Sample Delay	
Volume dB	Varies*	Octave	Varies*	Wave Start	0 %	Mode	Varies*
Panorama	Varies*	Semi	Varies*	Velocity	+53 %	Delay ms	Varies*
Pan Key	0 %	Cent	0.0	Random	0 %		
		Key	+100.00				

Filter-Map Screen (for all Layers)

Filter		Key Switch		Velocity Switch	
CutOff	100 %	High Key	G6	High Vel	127
Resonance	0 %	Low Key	E0	Low Vel	0
Velocity	+ 75 %	X-Fade	0	X-Fade	0
Key	-0 %				

Envelope Screen (for all Layers)

Envelope		Velocity to:	
Attack ms	0 ms	Attack	+39%
Decay	1.34 s	Volume	+ 97 %
Sustain	0 %		
Fade	0 %		
Release ms	1.34 s		

LFO/Matrix Screen: Sample Layers 1 to 5 - As below Row 2 only. Sample Layers 6 to 9 - As below Rows 1 & 2. All other Rows, select Off.

Matrix					
Row No.	Source	Modifier	Modify	Destination	Depth
1	Pitch	Off	0 %	Cutoff	-96 %
2	Constant	Off	0 %	Pitch	-100 %

\*Volumes, Panorama, Pitch Settings and Sample Delay Settings per Sample.

Sample Layer	Volume	Panorama	Pitch		Sample Delay	
			Octave	Semi	Mode	Delay ms
1	-5.2 dB	L64	0	0	Off	0 ms
2	-5.2 dB	L48	0	+7	Tempo	1/32
3	-7.2 dB	L32	+1	0	Tempo	1/32D
4	-7.8 dB	L16	+1	+7	Tempo	1/16
5	-8.3 dB	<C>	+2	0	Tempo	1/16D
6	-8.9 dB	R16	+2	+7	Tempo	1/8
7	-10.1 dB	R32	+3	0	Tempo	1/4T
8	-11.4 dB	R48	+3	+7	Tempo	1/8D
9	-14.3 dB	R64	+4	0	Tempo	1/4

The arpeggio speed depends upon the tempo of the style in the Total Preset in which the sound is being used. It also synchronises with that Style tempo if the Style is running.

The inaccessible effect layer seems to vanish but it's not needed here anyway.

The notes used are only octaves and fifths, so it works with both major and minor chords played at the same time by other sounds. There are many variations possible – you could adjust the Octave and Semi values so a major or minor arpeggio is played. Do not then raise the Sound Controls Major/Minor value to above 50% though. You could start with the original Harp sound and use the above values to produce the result.

The Descending Arpeggio can be achieved by reversing the Sample Layers above, so the values (except Sample Delay values which remain as for the Ascending Arpeggio) for Sample Layer 9 are applied to Sample Layer 1, 8 to 2, etc.

The third teaser audio demo on-site was a modification of the above to provide a five-note major seventh arpeggio from an acoustic guitar, suitable for playing at the end of a piece of music by just holding down a root key. Here's the set-up for that. Again, treat sections we've not yet covered as mere 'recipes'.

**Explanation of how to produce the Acoustic Guitar Major 7 Arpeggio**

Acoustic Guitar Major 7 Arpeggio [Load "Harp" : 090-000-124]

Samples 2 & 3 : Delete. Using the drop-down lists, change Layer 1 to Acoustic Guitars > Nylon Finger 1 Soft. Then adjust Layer 1 as below then copy Layer 1 to Layers 2 to 5, leaving the effect at Layer 6.

First Screen : Vol +6.0 dB, Octave +1. Reverb 1 -0.2 dB; Reverb 2 -5.1 dB; Remainder at Off.

Sound Controls : Attack Shape 0%; Color 50%; Tone 83%; EQ 50%; Perc Length 50%; Release 60%

Sample Screen

Volume		Pitch		Sample Edit		Sample Delay	
Volume dB	Varies*	Octave	Varies*	Wave Start	0 %	Mode	Varies*
Panorama	Varies*	Semi	Varies*	Velocity	+53 %	Delay ms	Varies*
Pan Key	-0 %	Cent	0.0	Random	0 %		
		Key	+100.00				

Filter-Map Screen (for all Layers) – blanks are unchanged originals

Filter		Key Switch		Velocity Switch	
CutOff		High Key	G6	High Vel	
Resonance		Low Key	A0	Low Vel	
Velocity		X-Fade		X-Fade	
Key					

Envelope Screen (for all Layers) – leave unchanged as original

LFO/Matrix Screen: Rows 2 to 6 OFF.

Matrix					
Row No.	Source	Modifier	Modify	Destination	Depth
1	Constant	Off	-0.0 %	Pitch	-100 %

\*Volumes, Panorama, Pitch Settings and Sample Delay Settings per Sample.

Sample Layer	Volume	Panorama	Pitch		Sample Delay	
			Octave	Semi	Mode	Delay ms
1	-0.0 dB	L32	0	0	Off	0 ms
2	+0.3 dB	L16	0	+4**	Time	490 ms
3	+0.7 dB	<C>	0	+7	Time	1.10 s
4	+1.0 dB	R16	0	+11	Time	1.76 s
5	+1.3 dB	R32	+1	0	Time	2.70 s

\*\*Make this value +3 and you have an Acoustic Guitar Minor 7 Arpeggio.

This sound is most suitable for the end of a piece, in the lower part of an Upper Manual split. Stretch the index finger from the left hand while holding a LM chord down and touch and hold the root of the tonic chord in the UM for the duration of the arpeggio. Four variations in the sound are possible from the "Color" Sound Control parameter.

Alternative Sounds: Using the "Sample" drop-down lists on the Sample screen, a large number of different sounds could replace each of the samples. Percussive instruments like guitars and pianos work particularly well, but anything could be used if it works.

You could start instead by loading a different sound from the Harp, but your Sound Control functions would then be different from the above and you would lose the four 'Color' variations in the sound.

Wave Start, Velocity and Random - expanded explanation

\*See page 7 for Ian's explanation of looping

Appendix

Ian's Expanded Explanation of Wave Start, Velocity and Random

If we imagine that the database has a sample of somebody saying the words or phrase: "One, Two, Three, Four". 'One' is at the beginning of the sample and 'Four' is at the end. There is a looped\* section which turns out to be around the word 'Four', so it will play back repeating the word 'Four, Four, Four,' etc until we release the key triggering the sample. Wave Start will move the start point when we trigger a note. For example, if I set Wave Start to 33%, in this example it would start with the word 'Two' ('One' has been passed by or skipped over), and then continue with the rest of the sentence ending with the looped section once again around the word 'Four'. So using Wave Start we could play back the sample from any word of our choice. We also can 'cut' into a word. If I changed Wave Start to 37% our phrase would then start with '...wo' and continue as before with 'Three, Four'.

**\*See page 8  
for Ian's  
experiment  
to  
demonstrate  
Velocity  
control**

Now let's add Velocity\* into the equation...

Set Wave Start back to 0%, so our phrase will now play back all the words in order, and use Velocity to dynamically change the position of where the sample will start to play from. Use a soft Velocity value and we will get all of the phrase as if nothing has altered. A hard Velocity will start the playback from the word 'Three' and continue as before. So we can sort of control (by dynamically playing) the start point by using 'Velocity to Wave Start'. Remove Velocity (set it to 0%) for now and go onto the next parameter...

Now let's add Random to the phrase. When we trigger a note, the phrase can start at any point the computer chooses at that given time. We are able to control the range of this with the Random parameter value. In this case a value of 25% gives us '...e, Two, Three, Four'. If I play that note again (with the same random value), the computer will choose another start point for the sample to play from governed by its percentage parameter, perhaps '...Two, Three, Four' this time. Trigger it again and we may get '...ne, Two, Three, Four'. If I use a much larger value, for example 100%, in the Random parameter and then play a note again, it's possible that the computer will pick the end of the Wave Start range, just before the loop point, so all we would get is the word 'Four'.

The thing to notice here is that, even with the maximum possible values, the Wave Start does not include the entire length of the sample. 'Four' is out of bounds, most likely because it contains the looped section. All of the samples that are looped in the data base have an area in which Wave Start cannot go beyond.

So what we have got is a way to change the start point value without using Velocity or Wave Start. Every time a trigger is received, the computer uses a number supplied by a Random Number Generator as the parameter's current value.

If you replace the phrase with any of the database samples, eg Bagpipes, the Sample Page can be set up so that, by using only these three parameters (Wave Start, Velocity, Random) on the two (hidden) samples (a Drone/Chanter split performed by the key map and not SF) that make up this sound, we can play just the 'Chanter' without the 'mordent/turn' (Wave Start only) or the 'Chanter' with the 'mordent/turn' (Wave Start + Velocity). By adding a small amount of Random with Wave Start and Velocity, we can produce no 'mordent/turn', or a single note from the 'mordent/turn' and then the loop or all of the 'mordent/turn' plus the loop.

If you use just Wave Start and Random, depending on the settings of each you will get a different sample start point each time you trigger a note; remember you can control the range of 'randomness'.

I've found that I use a small amount of Random (between 3% and 17% depending on the sample in question and the effect I'm after), in a collaboration with Velocity and sometimes Wave Start as well, to change how my notes sound. If you try this with a String Section, I think it (with the right values) adds more life to the sound. This happens because we are varying the attack phase of where the sample plays from on each and every key trigger. This also avoids the dreaded 'machine gun' effect with any sample in the data base.

**Explanation  
of looping**

### Ian's Explanation of Looping

With the 'Bagpipe' sample, it's a single layer, divided/split into two by using two individual samples, 'Drone' and 'Chanter'. The 'Drone' (on the lower part of the split) loops almost immediately from a key trigger. The 'Chanter' (on the right part of the split) plays a portion of the sample before it then loops continuously. It's that portion before the loop that contains the 'mordent/turn' that we hear, and are able to jump/skip over by using Wave Start and/or its associated parameters.

There are many possibilities with this type of technology. It all depends on the developer/manufacturer as to what sort of control and how much control is given to the end user. As far as looping goes, anything you can hear could be sampled, looped and then played from a keyboard or sequencer.

There are only three loop functions (that I know about):

Forwards: the sample will only loop in one direction. Going back to the example I gave you, 'Four, Four, Four' etc., the data is played back in the correct direction.

Backwards or Reversed: our sample will play back as 'ruoF, ruoF, ruoF'. The loop starts at the end of the sample loop and not the beginning, reading the data out in a reversed order.

Forwards/Backwards: this version is used for more difficult samples which are to be looped. The result of using this is 'Four ruoF, Four ruoF'. As you can see, it is a combination of both of the above modes.

**Demonstration  
Experiment to  
show how  
Velocity  
works**

Ian's Demonstration Experiment to show how Velocity works

First load a sound into a Total Preset to be then edited in Sound Factory, and for this I had a little look into the database to find a sound that makes any changes using the parameters in question fairly obvious to us, the player/programmer.

For this I have picked MusicBox \* 090-000-114. Load this into your Total Preset, make sure that in Selectors > Advanced > Dynamic is ticked and that its Dynamic Curve is set to 4.

Enter Sound Factory and then Expert Edit.

On the Sample page we can see that two layers are used. Mute the second layer (Sine) so we can only hear the first layer (Musicbox) and any edits made to it. (Multiple layers and effects running can make things confusing and hard to pin down as to what's going on!)

We are now going to bypass the unwanted 'Dynamic/Velocity' routings in other pages that are active in this first layer as they can interfere with our test results. They can always be added back at the original value or a value which suits you later after the initial programming has been completed.

Go to the Filter-Map page. In the Filter panel is Velocity +41%. Change this value to +0%.

Go on to the Envelope Page. In the Velocity to panel, change Volume +100% to +0%. Also set Sustain to 100% and Fade to +0%.

We don't need to touch the Matrix-LFO page as it has nothing that will interact with what we are about to do. So we have now neutralized any hardwired 'Dynamic/Velocity' routings apart from the one we will be using.

Go Back to the Sample page.

You might have noticed on Layer one (Musicbox) that, in the Sample Edit panel, Velocity already has the value +20%. Set this to +0% for now, as it is a good idea to listen to the sample unmodulated. This should help make any changes in parameter values clearer.

When we play this sound it has a 'bright tine' type attack which then fades away leaving something that sounds like three sine waves at approximately 8' + 4' + 1' (to my ear!). We can confirm this by changing the Wave Start value. A small value (0% to 10%) seems to have very little effect on the start of this sample. Values between 10% and 20% appear to progressively remove the 'tine' impact. Then, at 20% upwards, more and more of the 'tine' is removed until we get to 40% where it has then been bypassed/skipped over altogether. From 40% onwards is just the sine waves sound I mentioned before.

We have discovered that the 'tine' occupies only the first 40% range of the Wave Start parameter.

Set Wave Start to 0% and we get the full sample playing back.

Now we will add Velocity to Wave Start.

Change Velocity to +100%. Play the keyboard with a hard touch and you get the full sample playing back. Play the keyboard with a light (and I do mean light!) touch, the 'tine' is then bypassed and you get the rest of the sound that is left to loop on its own.

Now change the Velocity value to -100%. As expected this now works in reverse. Play hard on the keyboard and you get only the looped part of the sample. Play light (as above) and you get the start of the sample again with the 'tine' on the front.

Now set Velocity back to +0%.

Try adding a value of +100% to Random, (still with Wave Start at +0%). Playing notes at different velocities will have no effect now, but the sound still changes on each key stroke. Its Wave Start position is being moved/modulated. Sometimes we get the 'tine', sometimes only part of it and other times, none of it. It is random as to which start place/point the computer decides to play the sample from.

We can further refine this programming by changing the Wave Start parameter from +0% to +20%. Remember I mentioned that the first 40% of Wave Start deals with only the 'tine' in this samples case. By setting it to +20% you have now 'biased' the random function. You can find the particular value to suit your taste. With Wave Start at 20% the Random parameter now is more obvious with each note played.

So I hope we have now established that all the parameters found in Sample Edit not only work but you also now have a good idea of how they function and are interacting with each other.

Try to remember that 'Dynamic/Velocity' can be used in so many places in SF to control all sorts of parameters that it can get in the way sometimes of what we are listening for. If you can, disable it from anywhere you are not trying to change (as we did above) so you can focus on what happens in the area you are trying to change. This is most important.

**Using Wave Start to produce a Comb Filter**

Ian's Explanation of Using Wave Start to Produce a Comb Filter

Here is a new programming concept and its results will vary in value because it depends on the harmonic content of the sample selected to be used. I know this method as Comb Synthesis.

It's quite simple to implement on any sample based system. All that happens is we take the sample in question, copy and paste it to another layer (usually the one directly below for ease of editing), and adjust the Wave Start parameters to taste. Sounds simple, but in fact when we adjust the start point of the copy all sorts of things happen as the two samples interact with each other. By doing this we create a Comb Filter and one of its products is that it will accentuate certain harmonics and suppress others all at the same time.

Now before we go too deep there are two versions of this, and both are valid on this system.

Firstly, the Simple Comb. This is easy, as all you need to change is any of the Wave Start parameters on the copy. I would suggest you change only the Wave Start on its own first. Work from zero upwards with your data wheel to hear all the possible variations. Once you go beyond a certain point you will become unaware of any changes as the two samples then take on their own identities again. So the Comb range will in some cases be a small window of values, in others it will be greater. Remember to alter only the copy layer for now. Depending on the source sample, this should give a few variations of sound. Also try to remember that this is very dependent on the harmonic content of the sample. Usually the ones with a lot of harmonics (think brighter sounding, therefore more interaction of 'Phase' between harmonics on each layer) work well. This version depends on all the other parameter values being identical so that the only difference is the start point of the second sample. It is also easy to compare the results as all you need to do is mute the copy layer to hear the original untouched sample.

Now for the second version, called Advanced Comb. Basically you copy the sample as before but this time the one thing that must be changed is the Pan value. Usually the two samples are panned left and right to some degree. For the most pronounced effect, hard left and right will give you the biggest change once we alter Wave Start. Here are some examples for you to try.

Load Strings 1 Soft into a Total Preset. Enter Sound Factory and touch Expert Edit. We are then presented with two Layers, the Sample Layer called Strings Warm and a Factory Effect Layer which is a Basic EQ.

Let's do the Simple Comb version then, by selecting the first Layer and copying it to the clipboard. Then Paste Behind with the first Layer still selected. This adds a second Layer to the first duplicating all of the original sound programming. At this point the only difference is that the two samples play louder together than each one singly. This is to be expected as they are currently Phase Aligned, and therefore all the harmonics combine to reinforce their combined amplitudes. But this is the start of the comb process. You might want to mute the Effect Layer, (now Layer 3), so it does not interfere with what we are listening for.

Go to Layer Two, on the Sample Page.

It's time to start playing with the Wave Start parameter. Whilst repeatedly playing a note (G2 for example), change Wave Start from 0% to 2%. You should hear a change in the harmonic content of the sound. If you don't re-trigger the note for each value you will not hear the effect as we are dealing with the start point of the sample. Try 4%. Once again the sound changes its harmonic content. Another value to try is 7%. If you go beyond 12% (in this case), the samples are too far apart from each other and they no longer act as a Comb Filter, thus appearing to return to their original sound(s)/states. There is no doubt that 'combing' is taking place here with the larger values on this sample but it is well beyond the range of hearing!

So, small amounts of Wave Start values are more noticeable in this Simple Comb version. Let's turn this example into version two, the Advanced Comb.

Return Wave Start to 0%. All we then need to do is change the Pan value of the first layer to L64 and the value of Pan in the second layer to R64. Now this is odd: if you now change the Wave Start parameter as before, there appears to be no comb effect! Small or large values appear to do nothing to the sound. There is, however, a slight 'shift' in the sound image at the values stated above, but nothing as drastic as when the two sounds were, in fact, aligned to the same Pan position/or space.

So how do we produce the Advanced version? In order to demonstrate that we need to start with a different sample altogether.

Delete Layers two and three leaving Layer one. In the Sample Page change the Sample Bank to read Analog-Sounds 2 and from the Samples list, select W2 ZR Perc 1 Fast. This is a sample of an old Helios drawbar sound. We do need a couple of fixes before we go any further. On Filter-Map Page, change Cutoff to 100%. On Envelope Page, set Release to 40 ms. Okay, with that all in place the thing I noticed, (having owed a Helios), is that this sound lacks a 'richness' that the 'Analog range' had; it's not 'Stereo'. We are going to use the Advanced Comb to help to try to put some of that richness back.

First, change Layer one Pan to L64. Now, as before, make a copy and Paste Behind. Change Layer two Pan to R64. At this point we have taken a mono sample, duplicated it, then split the signals to go to different channels. This is also called a Mono Split, but it is not true 'Stereo'! In order to do that, you need to change the value of Wave Start. Once again, re-trigger the keyboard whilst changing the Wave Start parameter. As Wave Start increases, the image changes and produces a 'pseudo' stereo. There are several values that work well; 28% in this case is a favourite of mine, as are 4% (appears to open the image apart), 15% (more of the same).

So, from all of this are some conclusions.

First, use the Simple Comb as a method to create variations of a sample.

Secondly, use the Advanced Comb technique to help create a larger sounding image (pseudo stereo) than just a mono sample or mono split can produce on its/their own. This works a lot better with mono samples than stereo samples because a 'true' stereo sample will already contain several time/phase differences between the two channels that make up the sample, which is why it didn't work so well with the string sound. It is the interaction and difference of these two parts that give our ears and then our brains the idea of stereo. The Advanced Comb technique gives us the chance to change the time/phase relationship of one of the channels to the other. This isn't the same as 'Delaying' the signal and 'Panning' it to a different position in the stereo field. Delaying takes all of the source and reproduces it later. Advanced Comb takes only part of the signal and combines it at the same time with the original, with no delay.

Thirdly, you can combine both methods/techniques to change a mono sample into a variation of the original and then convert that into a pseudo-stereo image. Remember that if you start with one Layer, you will require the addition of a second Layer to perform the Simple Comb part. Then you will need to copy both parts to convert it to an Advanced Comb, a total of four Layers to complete the effect. If you do this you will need to make sure that Layers three and four (which will create the 'Advanced' part), will need a Wave Start value larger than their original counterparts. If it's less, you will most likely destroy the Advanced Comb and/or the variation as well.

Q1. The results obtained seems to depend upon the sample having a section which responds to Wave Start.

A1. All samples will respond to a change in Wave Start. Its effect is just more dependent on the information contained at the start of the sample. If the sample has a pronounced transient at or near its start (like the Music Box in the previous section of this Appendix), using Wave Start is more apparent. If the sample does not change that much from start to finish, then Wave Start will appear to do little or nothing.

Q2. I've also noticed a difference according to the pitch of the key being played, lower pitched keys usually exhibiting a more noticeable effect than higher pitched ones.

A2. Another use for Wave Start is to 'chop off' the 'dead air' at the beginning of a lot of samples. If you start with a sample of a sound that plays back at 'Middle C' as Middle C, then stretch that sample across the keyboard range to play the entire compass of the instrument, a couple of things happen.

First, when you go up an octave, the sample plays back at twice the speed of the original. This raises the pitch but also the play back rate, going from start to finish twice as quickly.

Second, if you go down an octave the sample play back speed is then half of the original. This lowers the pitch and play back rate taking twice as long from start to finish. The lower end of the keyboard (because of this slowing), introduces a lag or delay from when the note is pressed to the time it takes for us to hear anything. This space between the key down and the production of sound is called 'Dead Air'. Using Wave Start will enable us to skip over the 'dead air' triggering the sound to start when the key is depressed. The trouble is that when you do this from the Sample Page alone, it affects the whole range of the keyboard and not just the low end.

Fortunately the Matrix can help here because we can map the keyboard's current note value to Wave Start. Different values from the keyboard (different notes) will then supply Wave Start with a custom value for each note played. This can be used to trim off the 'Dead Air' in low samples which also has the positive side effect of playing them more in time with each other across the key range. But we shall have to wait a bit before investigating the Matrix.

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