SPL Analog Code® Plug-in
Manual

SPL

DrumXchanger

Optimize and replace drum sounds

Manual

DrumXchanger Analog Code Plug-in
Model number 1060
Manual version 2.0 – 12/2011

This user's guide contains a description of the product. It in no way represents a guarantee of particular characteristics or results of use. The information in this document has been carefully compiled and verified and, unless otherwise stated or agreed upon, correctly describes the product at the time of packaging with this document. Sound Performance Lab (SPL) continuously strives to improve its products and reserves the right to modify the product described in this manual at any time without prior notice.

Content

Content

Installation

Plugin Alliance Activation

Your Analog Code plug-in must be activated in your Plugin Alliance account. You can set it up and log into your account anytime at **http://www.plugin-alliance.com**

For details about the activation process, read the Plugin Alliance Activation Manual. The PDF file is stored in the same folder of your computer like this product manual file.

Alternatively, the following web page provides the same information: **http://www.plugin-alliance.com/activation**

System Requirements and Compatibility

For details about system requirements and supported platforms or formats visit **http://www.plugin-alliance.com/compatibility**

MAC and Windows Installation

1. Check for the latest plug-in software version before installation:
   http://software.spl.info/download
2. Execute the installer file and follow the instructions.

Glossary

Transient:	first impulse of a waveform
Host Program:	program on which the plug-in is running (Pro Tools, Cubase, Logic, etc.).
Sample:	portion of a sound recording, here: a drum sound
Trigger:	module that controls sound recognition
Loudness:	perceived volume as opposed to amplitude

SPL Analog Code® Plug-ins

While SPL hardware products have been fascinating audio professionals from home studio owners to mastering engineers in the world's most renowned facilities for years, the need for this technology in the form of plug-ins has also been an ever-growing demand. With the Analog Code® plug-ins we have finally accomplished our much desired goal: to transfer to the digital domain the high quality we have striven to achieve with our analog processors throughout several decades.

The first time we ever heard a software that fulfilled our expectations, one of our hardware developers said to the programmers: "you have cracked the Analog Code" — thus was coined the name of our digital products.

Introduction

The DrumXchanger

The name of the DrumXchanger is self-explanatory: it replaces drum sounds. But when it comes to processing it relies on a unique approach: it uses the Transient Designer technology for sound recognition. That way the DrumXchanger can faithfully recognize even the faintest ghost notes — regardless of their dynamic level.

To clearly understand the DrumXchanger's true potential it is important to understand that it is not merely a sound replacer. It is, in fact, a powerful tool to optimize drums tracks. The processing possibilities to improve and shape sound — without the need to replace any sounds — are almost limitless, while the actual sound replacement itself can be done gradually (0 to 100 percent) in order to mix both the original and the sampled sound.

Better simply — simply better

If the DrumXchanger stands out due to its extremely reliable sound recognition, its simplicity is nothing but surprising. The principle behind the DrumXchanger concept was to focus on the essentials: the snare, the toms and the bass drum.

All functions are clearly displayed and processing is done at the highest standards of precision and quality. When you need to create a quality sounding snare from a more than dubious drum sound in a matter of seconds, the DrumXchanger offers lots of possibilities.

DrumXchanger Analog Code® Plug-in

Dual Threshold Technology

Sound replacement technologies are usually based on level recognition, so you can only set the threshold for a given level. This results in softer sounds not being recognized due to the fact that, with a threshold set too low, other instruments mask the drum sounds, making an accurate recognition impossible.

On the DrumXchanger you can set two different thresholds for a more accurate recognition of drum hits: a level threshold and a transient threshold. The sample plays back as soon as both of these values are reached. Simple operation — sound recognition.

Two Transient Designers included

The Transient Designer isn't only active on the background. Apart from the basic functions, the DrumXchanger also includes two Transient Designer processors, one for the original and one for the sampled sound.

1 GB Sample Library included

The DrumXchanger includes a 1 GB sound library with high quality 24 bit/96 kHz samples. The library, created exclusively with SPL's preamps and processors, consists of three drum kits with one snare (including rim shots), one bass drum and up to four toms per kit. Every multisample is made up of up to 80 sounds with sixteen dynamic levels and five variations. With the SPLX File Editor you can even create your own multisamples and import external samples in WAV and AIFF format.

Introduction

Features

- Phase-accurate processing
- Free sample mixing
- Ducking function for the original signal
- High and low pass filter on each module
- Transient Designer on both the original and the sample module
- Dynamic control to adjust the sample's loudness
- Real-time +/- one octave pitch sifting
- Delay control for precise replacement

Applications

The DrumXchanger can be used in mixing applications to process single drum tracks. You can optimize drum elements that are typically recorded with close microphone techniques: snare, bass drum and toms.

The DrumXchanger was not conceived to replace hi-hats and cymbals, nor drum sounds in a finished mix down.

Mouse wheel control for all rotary knobs

All SPL Analog Code plug-ins support mouse wheel control for rotary controls and faders. Place the mouse cursor over a rotary control and move the scroll wheel of your mouse to adjust the setting. Hold the CTRL (Windows) or COMMAND (Apple) key while moving the scroll wheel to make fine adjustments; the resolution of the mouse wheel is increased, making fine-tuning easier.

Keyboard Shortcuts

All SPL Analog Code plug-ins support format and OS specific functions for value reset, fine adjustment and mouse control. For more detailed information please refer to the host program's documentation.

Mono, stereo or multi-channel operation

The DrumXchanger plug-in can be used either for mono or stereo operation. All samples of the SPL library are mono. In stereo operation, each control element governs both channels simultaneously and you can load your own stereo samples.

The DrumXchanger is divided in five vertical sections: Editor, Settings & Input, Original, Trigger, Sample, and Output. We call these sections Modules. Here you will find a short overview of the different Modules. To learn more about the functions refer to „Control Elements" on page 13 et seq.

(1) Editor, Settings & Input

Here you can access the EDITOR to create and import samples, and you can also open the quick HELP guide.
Use buttons A-D to save your settings in a preset. COPY and PASTE allow you to copy the settings from one preset to another.
The INPUT GAIN control determines the input level.

(2) Original

Here you can process the original signal with a high pass filter, a low pass filter, a Transient Designer, and a ducker. The sound changes you make in this module have no influence on the actual sound recognition and replacement.

③ **Trigger**

The original signal is doubled in order to feed a second path leading to the Trigger module. It is at this point that sound recognition takes place. You have different options to optimize recognition. As soon as the recognition criteria are met, the sample is triggered.

④ **Sample (two columns)**

In the Sample module is where you choose the SPL Kit, i.e. a drum set, whose samples are loaded in the plug-in. The dynamic and frequency filters allow you to process the samples. Use the Delay control to adjust the exact position of the sample, and the Tune control to adjust the pitch.

⑤ **Output**

In the Output module is where everything merges. Here you set the ratio between the original and the sample signals, as well as the filter and volume settings for the output.

First Steps

1. Insert the DrumXchanger as an effect in a drum track.
2. Choose a sample.
3. Make any necessary fine adjustments in the Trigger module.
4. Get to know all other settings in the other modules.

On step 1.
There is no sample loaded in the DrumXchanger when you load the plug-in in your host program. However, the Trigger is already working, as you can see in the corresponding meters. Insert the DrumXchanger in a snare, tom or bass drum track. Loop a couple of bars and play them back — you can also do it in solo listening mode.

On step 2.
Choose a sample from the SPL Kits or load an SPLX, WAV or AIFF file. Click on LOAD to point out the folder in which you have saved the SPL Kits (see "LOAD" on page 24).

On step 3.
In the Trigger module, adjust sound recognition with the meters' green arrows. These arrows determine the signal's level and transient thresholds. Set the arrows according to your needs. The samples are only played back when both arrows blink.
Rim shots: use the red arrow to set the level from which a snare or tom hit ought to include a rim shot. Use the RIM control above the meters to set the ratio between the rim shot and the sample.

On step 4.
The DRY/WET function in the Output module is of utmost importance: use it to determine how much of the sample signal is to be mixed with the original signal.
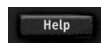
EDITOR, SETTINGS & INPUT MODULE

Editor

Click on this button to access the EDITOR window to create your own multisamples in SPLX format. For more information refer to "EDITOR" on page 30.

Help

Click on HELP if you want to recall the "First Steps" section on page 12 of this manual. To close the HELP window simply click on it.

Settings A/B/C/D, Copy and Paste

The four settings buttons (A-D) allow you to save all your settings with a simple mouse click (settings = presets. We use the term "settings" here to avoid confusion with host presets). The DrumXchanger saves the current settings permanently. As soon as you click on another settings button, the current settings are saved under the previously active setting. For example: if setting B is on and you were to click on another settings button, all parameter settings would be saved under setting B.

Any previously saved setting can be recalled with a simple mouse click on the corresponding button; you can then use or edit the settings.

Right above the settings buttons (A to D) are the COPY and PASTE buttons. Use them to copy your adjustments in one of the settings ("A", for example) into another one ("B", for example). Select setting A and make any necessary adjustments; if you wish to transfer those adjustments to another setting click on COPY. Select setting B and click on PASTE. The COPY and PASTE functions make it easier to compare settings that differ just slightly. To do so, COPY the adjustments in one of the settings (C, for example), change to another setting (D, for example), PASTE the previously copied settings, and then modify the adjustments in that setting (D). Now you can do a simple and direct comparison between the adjustments saved in settings C and D.   ⋯⋯⟩

If the host program allows it, the settings can also be automated so you can use different settings at different points. The automation parameters in the host program's list is called "Settings".

As long as you work within a single session of the host program and the plug-in is installed, the settings are saved and can be recalled afterwards. When opened, the plug-in loads the active setting adjustments instead of the default settings. If you remove the plug-in from the host program all settings are lost.

There is an exception though: when you save all settings of a plug-in instance as a preset in your host program. A host preset includes all internal settings.

To erase all settings at once you can reinstall the plug-in or reset it in the Preset Manager of your host program.

Input Gain

Use the INPUT GAIN control to determine the input level of the original sound and adjust the signal's volume. The level meters in your host program ought to give you a good idea of the original signal's input level. You can use the INPUT GAIN control to compensate low and high level signals.

These adjustments have no influence whatsoever on the recognition and trigger level. The level ranges from -15 dB to +15 dB.

The OVL LED left of the INPUT GAIN control indicates if the processor is overloading. The OVL LED is post-INPUT GAIN, so it indicates if an input signal is too hot, but also if the overloading is taking place after the INPUT GAIN, which means that the adjustment of the latter might be too high.

ORIGINAL MODULE

High and low pass filters

The first controls are the high and low pass filters. A high pass filter (HPF) only allows frequencies above a certain frequency to pass through, while a low pass filter (LPF) only allows frequencies below a certain frequency to pass through. These filters allow you to cut the low or high end of the original signal to focus on a narrower frequency range. Typical uses of these filters include the filtering of rumble and low frequency interferences with an HPF, and the suppression of hi-hat crosstalk with an LPF. These settings have no influence whatsoever on the sound recognition or the trigger module, nor do they affect the sound of the sample.

The high pass filter goes from 20 Hz to 11 kHz. The low pass filter goes from 20 Hz to 22 kHz.

Solo

When you activate SOLO on the Original module you will listen only to the original signal. This will allow you to concentrate on that specific signal and make all necessary adjustments.

Do note that the DUCKING function works independently from the SOLO button. To avoid any influence on solo listening, deactivate the ON button next to the DUCKING control.

Transient Designer Processor

A transient can be defined as the first impulse of a waveform. What the Transient Designer does is shape the waveform according to the transient, and it only needs two controls to achieve that, either by amplifying or attenuating the ATTACK and/or SUSTAIN. Processing is not level dependent so it acts equally on all transients, regardless of whether they are loud or soft. Among its most common uses are the sound shaping of percussion instruments. ⋯⋯⋯⋯⟶

The Transient Designer in the DrumXchanger's Original module allows you to process the original signal. You have a fully operational Transient Designer at your disposal to shape the sound of the signal as you wish. The processing of the original signal will depend on whether you plan to mix it or replace it with the sample.

You can either process it so that it fits better with the sampled signal when mixed together, or you can optimize it so the Trigger module can recognize it better. Do note that these adjustments only influence the original signal. The Trigger module is always fed the same signal as the Original module, except when you depress the TRIGG. button on the Original module. In which case, the processing you carry out with the Transient Designer on the Original module also affects the Trigger module's signal.

Transient Designer Processor/Attack

The ATTACK control cuts or boosts a transient attack up to 15 dB. Positive values enhance the attack while negative values reduce the attack.

Transient Designer Processor/Sustain

The SUSTAIN control cuts or boosts a transient sustain up to 24 dB. Positive values extend the sustain while negative values shorten the decay.

Trigg.

When the TRIGG. button is activated, the Transient Designer processor of the Original module also affects the Trigger module.

This can be very useful when drum sounds are too resonant and therefore not easily recognizable or distinguishable. This usually affects toms and bass drums rather than snares. If that were the case, activate the Transient Designer in the Trigger module for a more precise processing. If the sound is sharper, recognition can be more easily achieved. This processing has no influence on the sound of the sample.

Ducking

The DUCKING function reduces the level of the original signal momentarily. It can be set from 0 dB to -40 dB. Time constants are preset and cannot be modified: attack is set to 1 ms and release to 10 ms, so level reduction takes place for a very short time.

The higher the DUCKING value, the more the original signal is overshadowed by the sample. The short time constants guarantee that all room acoustics information in the original signal is kept intact. Thus, sample replacement is seamless and sounds more natural.

Use the ON button next to the DUCKING button to activate or deactivate this function. This way you can also compare the DUCKING processing easily. Do note that the DUCKING function works independently from the SOLO button in the Original module. To avoid any problems with the DUCKING function, deactivate it with the corresponding ON button.

TRIGGER MODULE
Frequency and bandwidth controls

The first two controls on the Trigger module are the frequency and bandwidth (Q) controls. Sound recognition can thus be focused on a more specific frequency or bandwidth. You can actually set these controls by ear most of the time. The values are displayed underneath the control so that you can relate to these values when processing the signal. The frequency control ranges from 20 Hz to 22 kHz. The bandwidth (Q) control ranges from 0.5 to 50. High Q values result in a narrow bandwidth and viceversa. The maximum Q value defines a very narrow and therefore more precise processing range.
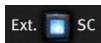
IMPORTANT: The precision and speed at which sound is recognized depends a lot on these two controls. Set the frequency as close as possible to the drum's fundamental frequency and define the bandwidth as precisely as possible. To make fine adjustments use the SOLO function.

DrumXchanger Analog Code® Plug-in   **17**

Solo

When you activate SOLO on the Trigger module you will listen only to the trigger signal. This will allow you to set the frequency and bandwidth parameters as precisely as possible, but also to check all other module parameters and optimize them.

Ext. SC

Ext SC. stands for "External Side Chain". The EXT. SC button activates the plug-in's side chain in order to feed it with, for example, the signal of a microphoned drum trigger.

To learn how to feed the side chain of a plug-in, please refer to your host program's documentation. Not every host program offers this function.

Gain controls

Use the the two gain controls above the trigger meters to adjust the level for the dual threshold recognition. Frequency and bandwidth settings in the Trigger module can often result in signal level changes. If this were the case, adjust the level with the GAIN control.

TRANS Gain control

The left-hand GAIN control determines the level for transients. If the signal does not have a wide dynamic range, you can improve the resolution of the parameter by increasing the GAIN value for transients. That way you will be able to make a better reading of the TRANS trigger meters and set the green arrows more precisely. Always try to keep a balance between the GAIN control and the position of the green arrows in the TRANS trigger meters to guarantee a better sound recognition. As a general rule, it is recommended to take the highest transient as a reference in order to take full advantage of the meters, and set the green arrows just underneath the faintest transient. The meters display peak values with horizontal lines that are held for a short period of time(peak hold). See also TRANS trigger meter on page 20.

LEVEL Gain control

The Trigger module's right GAIN control determines the level for signal recognition by the trigger. If the signals are too low, increase the level; if the signals are too high, reduce the level.

The trigger reacts to volume changes with the corresponding selection of louder or softer samples. If you want to be sure that the sample played back is always the loudest, increase the GAIN value until the peak signal reaches the upper region of the trigger meters and the red mark lights up.

Always try to keep a balance between the GAIN control and the position of the green arrows on the LEVEL trigger meters to guarantee a better sound recognition. For a better sound replacement, it is recommended to take the highest level as a reference in order to take full advantage of the meters, and set the green arrows just underneath the lowest level. The meters display peak values with horizontal lines that are held for a short period of time (peak hold). See also "LEVEL Trigger Meter" on page 21.

Trigger Meter

In contrast to common sound replacement tools, the DrumXchanger's Trigger module processes signals at two different levels. The Dual-Threshold technique takes transient and level values in consideration and therefore guarantees an extremely effective recognition.

This recognition method relies on the Trigger module's central tools for graphic visualization and control, i.e. the trigger meters, which give information about the replacement activity. The left-hand TRANS trigger meter indicates the signal's transient level, while the right-hand LEVEL trigger meter displays the level of the trigger signals.

The green arrows indicate the threshold for samples to be played back. Both thresholds must be reached at the same time (both arrows blink) in order for the sample to be played back. The blue LED in the PLAY button illuminates when a sample is triggered.

The red arrow on the right, next to the LEVEL trigger meter, determines the threshold for RIM shots to be triggered. If the red arrow is set accordingly, the rim shot will also be played back when reached.

TRANS Trigger Meter

The left-hand trigger meter indicates the transients' level. Use the GAIN control directly above it to set the amplitude level (see Transient GAIN Control on page 19).

The green arrow corresponds to the threshold, in other words, it determines when a sample is to be played back (arrow blinks). Given that recognition relies on both parameters (TRANS and LEVEL), both arrows must blink at the same time in order for the sample to be played back.

Normally, it is recommended to take the highest transient as a reference in order to take full advantage of the meters, and set the green arrows just underneath the faintest transient. The meters display peak values with horizontal lines that are held for a short period of time (peak hold).

LEVEL Trigger Meter

The right-hand trigger meter indicates the level of the trigger signal. Use the GAIN control directly above it to set the amplitude level (see Level GAIN Control on page 19). If the level is too low, pull the GAIN up; if the level is too high, bring it down. GAIN is set perfectly if the level range is fully used. If the level reaches the top, a red mark blinks to show that the loudest sample is triggered. The green arrow corresponds to the threshold, in other words, it determines when a sample is to be played back (arrow blinks). Given that recognition relies on both parameters (TRANS and LEVEL), both arrows must blink at the same time in order for the sample to be played back (PLAY button blinks).

For a better sound replacement, it is recommended to take the highest level as a reference in order to take full advantage of the meters, and set the green arrows just underneath the lowest level. The meters display peak values with horizontal lines that are held for a short period of time (peak hold).

The red arrow on the right, next to the LEVEL trigger meter, determines the threshold for RIM shots to be triggered. As long as the red arrow is set accordingly, the rim shot will also be played back when this threshold is reached (see also "Rim" on page 25).

SAMPLE MODULE

The Sample module is the only one divided in two columns. We will describe the functions starting from top to bottom of the left-hand column.

The SPL Kits

The DrumXchanger includes four SPL drum kits (SPL Kits). All SPL Kits include one snare, four toms and a bass drum. Every sample is actually a multisample made up of five variations with sixteen dynamic levels, which result in 80 different sounds for every drum. The dynamic level of the sample depends on the original signal, i.e. loud sounds are matched to loud samples. Sound variations are very slight and are assigned randomly, so when a drum hit is replaced it sounds more natural, considering that a drummer doesn't always play the exact same sound every time.

You can also add a rim shot sound to every snare and tom sample. All samples were recorded by Chris Marr at Prinzipal Studios in Münster, Germany. We used SPL channel strips and preamps and four Transient Designers. All drums were recorded with all relevant drum playing techniques. Drum hits were recorded in one-dB steps from 0 dB to -28 dB and recorded at 24 bits/96 kHz.

SPL Kit Selectors

The first four buttons in the Sample module's left-hand column are meant to select one of the SPL Kits.

Visual SPL Kit

The image below the kit selector buttons symbolizes a drum kit. The following descriptions apply only if a multi-sample from an SPL Kit is loaded. If you load other samples, the image turns gray and has no function.

The large circle on the bottom left symbolizes the snare; the four size-increasing circles above it symbolizes the toms; and the large colored area underneath these circles symbolizes the bass drum.

Click on any of these symbols and the corresponding sample is loaded. For example, if you wish to load a snare sample from one of the SPL Kits, click on the snare symbol. To compare all the snares from the different SPL Kits, click on the snare symbol and then click on the four kit selector buttons above it. The same applies to all SPL drum kit elements.

Every time you change a kit, via the kit selector buttons, the color of the image changes: Kit 1 is red, Kit 2 is blue, Kit 3 is green, and Kit 4 is yellow. It is nice and colorful, but it is also very useful: the association of different concepts is one of the best learning methods. After working with the various SPL Kits for some time, it will be easier for you to associate the sounds of the different kits to a color.

File Name Button

Directly underneath the Visual SPL Kit is a button where you can read the file name of the selected sample. That way you always have a reference of the active selection. When it reads "SPL K3 Snare.splx" you will know that it is the snare from SPL Kit 3 that is being used; if it reads "MajorTom.wav" then it is a WAV file from another source that is active. Clicking on this button has the exact same function as clicking on the LOAD button described below — it opens the "Select Drum Sample" dialog.

Control Elements SAMPLE Module

`Load` ## Load

When you click on the LOAD button the "Select Drum Sample" dialog appears on screen. Choose an SPLX file or a WAV or AIFF sample and load it into the DrumXchanger's Sample module.

IMPORTANT: The first time you use the DrumXchanger you must specify, via the LOAD function, the folder in which you have saved the SPL Kits. Click on LOAD and select the folder in which you have saved SPL Kits 1 - 4. This path will be permanently stored so that you always have a direct access to the SPL Kits.

`Play` ## Play

When you click on PLAY, the sample is played back once. If you only wish to listen to the sample, activate the SOLO function in the Sample module's right-hand column.

The PLAY button lights on every time a sample is played back by the Trigger module. Thus, it can be very useful for setting the trigger meters, since it serves as an additional visual indicator for when a sample is being played.

`Prev` `Next` ## Prev and Next

These two buttons allow you to select the PREV(ious) or NEXT sample available. The next sample is always the next sample file available in the same folder. For example, if you load an SPL Kit and the currently active sample is the snare sample, when you click on PREV, the sample selected will be the bass drum, while the NEXT one will be the first tom.

However unspectacular at first sight, this unique function can be very practical, considering that it can be applied during a running session. When you have a folder with lots of different samples of the same type — snares, for example —, this function becomes very powerful, since you can easily listen to all the samples available and apply them directly within the project. You will be able to listen to the drum sound in context, making the selection for you and other people involved easier.

24 *DrumXchanger Analog Code® Plug-in*

Rim

The red arrow on the right side of the LEVEL trigger meter determines the level from which an additional rim shot sound is to be played back. Use the RIM control to set the rim shot volume in relation to the drum sample.

All SPL Kits include rim shot sounds for all snare and toms multi-samples.

Phase

This button reverses the polarity of the sample signal. If there were any phase cancellation problems with the original signal while using a sample, use this function to correct them. These cases ought to be very rare, so always choose the setting that sounds better.

Delay

The DrumXchanger always matches the original and sampled drum hits automatically. However, the possibility of a delay is always present (if the sample was not cleanly edited, for example). A decrease in low-end power or the appearance of flams (double hits) are typical signs of a mismatch between the original and the sampled signal. Use the DELAY control to correct these problems. The delay time ranges from +3.5 ms to -3.5 ms.

Dynamics

This control determines how the DrumXchanger reacts to level variations in the original signal and plays back loud or soft samples accordingly.

In the center position the relation is linear, i.e. the relation of the original sounds to the loudness of the samples follows a linear pattern. This linearity is indicated by the straight line above the center position of the knob. The more you turn the knob to the left, the faster the volume of the played back samples increases. This relationship, where the dynamic differences are very small, is indicated by the rapidly increasing curve at the left end of the knob.

On the contrary, if you turn the knob to the right, the volume of the samples increases more slowly. This relationship, where the dynamic differences are larger, is indicated by the slowly increasing curve at the right end of the knob.

Two application examples: to even out fluctuating drum sounds turn the knob to the left; the further you turn, the higher chances are that samples with the same volume are played back (= lower dynamic range).

When you wish to replace ghost notes but the sounds are too loud, turn the DYNAMICS knob to the right to increase the dynamic range. By doing so, the loudest samples will be played only with the loudest drum hits.

High and low pass filters

The first controls in the Sample module's right-hand column are a high and a low pass filter, just like on the Original module. A high pass filter (HPF) only allows frequencies above a certain frequency to pass through, while a low pass filter (LPF) only allows frequencies below a certain frequency to pass through. These filters allow you to cut the low or high end of the sample signal to focus on a narrower frequency range. These settings have no influence whatsoever on the sound recognition or the Trigger module, nor do they affect the sound of the original signal. The high pass filter goes from 20 Hz to 11 kHz. The low pass filter goes from 20 Hz to 22 kHz.

Solo

When you activate SOLO on the Sample module you will listen only to the sample signal. This will allow you to concentrate on that specific signal and make all necessary adjustments.

Transient Designer Processor

A transient can be defined as the first impulse of a waveform. What the Transient Designer does is shape the waveform according to the transient, and it only needs two controls to achieve that, either by amplifying or attenuating the ATTACK and/or SUSTAIN. Processing is not level dependent so it acts equally on all transients, regardless of whether they are loud or soft.

The Transient Designer in the DrumXchanger's Sample module allows you to process the drum sample. The Transient Designer's potential is often underestimated given that it only has two adjustable parameters. Nevertheless, the acoustic interaction of the four essential characteristics affected by the control options provides everything to process a dynamic signal curve. It is a very useful tool that can be easily set "by ear" and whose real potential is only evident in the outcome. And it is precisely this sound shaping based on the signal's dynamic pattern that is very useful when embedding sounds. So, when you use the Transient Designer to optimize a sample for replacement, the results are amazingly perfect.

Transient Designer Processor/Attack

The ATTACK control cuts or boosts a transient up to 15 dB. Positive values enhance the attack while negative values reduce or eliminate the attack.

Transient Designer Processor/Sustain

The SUSTAIN control cuts or boosts a transient's sustain up to 24 dB. Positive values extend the sustain while negative values shorten the decay.

Tune

The DrumXchanger even allows you to tune the drums in your sample library. The Tune control changes the pitch of the samples in +/- one-octave steps. The scale is divided in twelve semitone steps (ST = semitone) starting from the neutral position at the center of the knob.

Use the TUNE control to match a sample to the tuning of your drum kit or a particular song.

OUTPUT MODULE

High and low pass filter

The first two controls on the Output module are a high pass and a low pass filter, just like on the Original and Sample modules. The high pass filter goes from 20 Hz to 11 kHz. The low pass filter goes from 20 Hz to 22 kHz.

A high pass filter (HPF) only allows frequencies above a certain frequency to pass through, while a low pass filter (LPF) only allows frequencies below a certain frequency to pass through. These filters allow you to cut the low or high end of the output signal. If the filters in the Original and Sample modules serve more of a corrective and creative purpose, the filters in this module are meant to help you shape the overall sound. The combination of all module filters results in a very large flexibility and a greater effectiveness.

Dry/Wet

Use the DRY/WET control to determine the ratio between the sample and the original signal. The ratio ranges from 0% to 100%, where 0% means only the original signal is played back (DRY) and 100% means only the sample is played back (WET). When monitoring, the Sample module's SOLO function has the same effect as turning the DRY/WET control to 100%.

Output

The OUTPUT control reduces or increases the output level. The level ranges from -15 dB to +15 dB.

The 0 dB mark is set at the center position (12 o'clock).

Generally speaking, you will need to reduce the level, considering that the DrumXchanger processing tends to increase it. Supposing you mix a sample with an already hot snare drum recording, or that it might be necessary to emphasize the ATTACK in the Original and Sample modules, you can almost take for granted that the output signal will overload (OVL LED blinks). If this were the case, reduce the OUTPUT level.

IMPORTANT: Use the OUTPUT control to take full advantage of the dynamic range available, but try to avoid distortion. The OVL (Overload) LED next to the OUTPUT control can be a very useful reference. This LED indicates overloading is occurring at the plug-in's output, which might result in distortion. The OVL LED should never turn on. If it does, turn the OUTPUT level control down until it turns off again.

EDITOR

The purpose of the Editor is to allow you to create your own multisamples in SPLX format. You can create three sound variations with eight different dynamic levels. Snare and tom samples can also include RIM shot sounds.

If you are interested in commercial sample production for the DrumXchanger, write an e-mail to software@spl.info. Development partners get the Pro Editor which allows to manage up to 128 velocities and 10 variations.

Dynamic Level   Variation 1   **Variation 1-3**

Load three different drum hits with the same volume. The DrumXchanger plays the variations randomly to achieve a natural sounding performance.
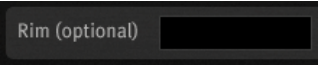
(loud) 1   **Slot 1-8 (loud ... soft)**

Volume variations are loaded into the slots, from top (louder) to bottom (softer), according to their loudness. You do not have to load each slot. Dynamic is scaled automatically with less than eight samples.

Rim (optional)

Load rim shot sounds into the RIM slot.

Rim shot variations are handled the same way as standard sample variations.

Slot Buttons

Behind every slot are three buttons.

1. The button with three dots on it opens the dialog window to load samples. It has the same function as simply clicking on the slot.
2. The button with the right-pointing arrow is used to play back the sample.
3. The X button empties the slot.

Export

When you have finished editing, export the end result as a multisample in SPLX format. Choose the location where it should be saved. We recommend to create and maintain a tidy collection of DrumXchanger samples, so that they are always readily accessible. You can not save or recall an editing status. You can only export an SPLX multisample, and each multisample must be created as a new SPLX file.

Close

To go back to the DrumXchanger's graphic interface close the Editor window by clicking on CLOSE. Note that previous editing is lost after pressing CLOSE.

DrumXchanger
SPL Analog Code® Plug-in

Manual