Shapee VST Plug-in
User Guide

Introduction

This user guide describes the *Shapee* VST plug-in for PC. Information on how to install and run the plug-in and a description of what it does are all included in this document.

The Process

Shapee is a 'frequency shaping tool' conceived and described by Christopher Penrose in his papers "Frequency Shaping of Audio Signals" and "Extending Musical Mixing: Adaptive Composite Signal Processing". This plug-in offers *Shapee* in a real-time processing environment (i.e. any program you have on your PC that can host VST 2.0 specification plug-ins) with some additional controls. Examples of what this process does can be found at http://www.sfc.keio.ac.jp/~penrose/shapee/ and in the Plug-Ins section of http://www.mp3some.co.uk./

Frequency shaping uses short-time Fourier analysis and resynthesis to combine two sounds according to their spectral content. The 'spectral content' of a sound here is considered to be the magnitude and phase of a sound in different frequency bands and how they vary over time. *Shapee* treats one of the incoming sounds as an 'amplitude reference' and the other as a 'frequency reference'. The output sound combines the frequency distribution of the frequency reference with the spectral formant of the amplitude reference. In many cases this results in the timbre of the amplitude reference following the pitches in the frequency reference

If the last paragraph made sense, great! If not you have two options. The first is to go and read a good book (or article) on spectral processing (see the list later on this guide) and then read Christopher Penrose's papers. The second is to install the plug-in now, experiment and hear what it does.

Installation

This plug-in uses the Intel Signal Processing library (SPL) to perform parts of the algorithm (such as the Fast Fourier Transform). This means that, on Intel processors, the algorithm is extremely efficient. The plug-in will run on other Intel architecture processors (such as AMD) but is not optimised for these. The Intel SPL is rather clever in that it detects the type of processor (PI, PII, PIII or PIV) in your computer and loads program code which will give the best performance for that processor.

In order for this plug-in to run you must have additional Intel .dll files to detect and load the right code for your processor.

Make sure you follow these installations steps:
1. Download the Intel.exe and Shapee.exe files on to your hard drive.
2. Double click on Intel.exe and extract the files to the directory **c:\windows\system** (they should automatically extract to this directory – if

your windows system folder is not on your c: drive you will need to change the destination folder of the files int his archive).

3. Double click on Shapee.exe and extract the file to your VST plug-ins directory. Check the manual for your VST host application (e.g. Cubase, FruityLoops, Logic Audio) to see where on your system this folder should be. The default extraction destination is **c:\Program Files\Steinberg\Vstplugins**

How to use

This plug-in can only be used as an 'insert' style effect in a stereo signal path (track or buss). If it is used as a 'send & return' type effect it will not output any audio. For information on how to use insert effects in your host application see its user manual.

This plug-in must be used in a stereo signal path as it uses the left input for signal 1 and the right input for signal 2. The ouput is in mono (left and right outputs are identical). The plug-in takes two signals and combines them into one mono signal. Unfortunately, although 'side-chaining' has been an essential part of studio techniques for decades (used in keyed gating, de-essing etc.), it is not catered for in the Steinberg VST specification. This means that processes which combine two signals in some way (e.g. vocoders, side-chain compressors etc.) cannot be used as easily as they could be. In Cubase VST we have to treat the two separate signals as the left and right channels of a stereo signal. At the time of writing this document Emagic Logic 5.0 has just been released which offers side-chaining, however until I am able to test this version I do not if (or how) this side-chaining will work with VST plug-ins. A quick (-ish) way of combining the outpus of two mono soundfiles is as follows: Place one soundfile on Track 1 of your host application (e.g. Cubase) and the second soundfile on Track 2. Pan track 1 hard left and track 2 hard right and send these to a buss. Insert the plug-in across that buss – track 1 will be treated as signal 1, track 2 as signal 2 and the combined sound will be output on both sides of this buss. Otherwise use a stereo track and combine the two soundfiles beforehand.
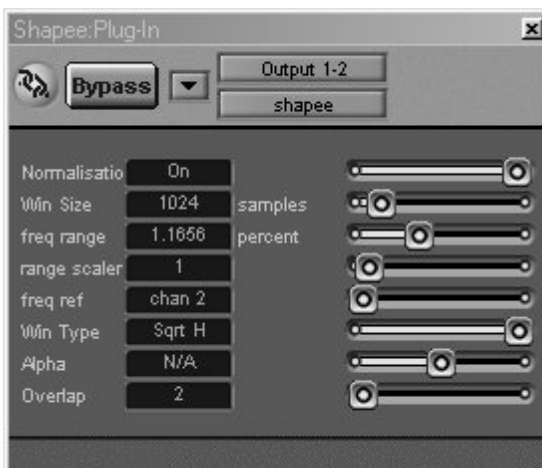
Screenshot: Shapee v1.0 (in Emagic Logic Audio 4.7)

Parameters

The plug-in offers eight user adjustable parameters, which can be altered whilst the audio is playing, and these are described below. Don't worry if you're not sure you understand what these parameters do – experiment and use your ears:

Normalisation: This normalises the ouput sound so that any reduction in the level of the frequency reference by the *Shapee* process is compensated for. Whilst this can prevent the ouput level varying it can produce a 'pumping' effect similar to when heavy compression is used on a signal. This can be set to 'on' or 'off'.

Window Size: This is the length (in samples) of the 'analysis' and 'resynthesis' windows applied to the audio. The longer the window, the finer the frequency precision but the lower the time resolution and vice versa. This can be adjusted in steps of 4 samples. The range of values available is 128 to 8192. At a sample rate of 44.1 KHz (CD quality) a window length of 8192 samples is equivalent to about a fifth of a second. The output of the process is delayed by the window size so you will need to compensate for this by setting a negative delay on any tracks you use this plug-in on in your sequencer.

Freq Range: Frequency range. The *Shapee* process works by shaping the signal across frequency bands. This parameter controls how wide these frequency bands are. For example, when the frequency range is 1% then the frequency spectrum is divided up into 100 separate bands for processing. When set to between 1 and 1.5 % the output sound combines the timbre of the amplitude reference with the frequencies of the frequency reference. As this value gets greater the timbre of the ouput becomes more and more like the timbre of the frequency reference. Much below 1% the frequencies of the two signals are heard together which can produce a very dissonant sound. At 0% the timbre and frequencies of the amplitude reference are heard.

Range Scaler: As stated earlier, to create a hybrid between two sounds, a frequency range between 1.0% and 1.5% is best. However it is possible to create morphing style effects between two sounds by varying the frequency range between 0% and 100%. To offer fine control of the frequency range parameter between 0% and 100% the range scaler allows the user to change the range of values over which freq range operates. When range scaler is set to 0, freq range operates between 0% and 0%. When range scaler is set to 32, freq range operates between 0% and 100%. A range scaler setting of 1 is useful for producing static hybrid sounds. Larger range scaler values are useful for creating morphs by adjusting the freq range control in real time.

Freq Ref: Frequency reference. As stated earlier one of the input sounds to this process is treated as the frequency reference and the other is treated as the amplitude\formant reference. This control allows you to select which input to the plug-in will be the frequency reference and which will be the amplitude reference. There are two settings: chan 1 and chan 2.

Window Type: This allows the user to choose between six different shapes of analysis window (Hann, Hamming, Kaiser, Bartlett, Blackman or square root Hann). Analysis windows are used to prevent parts of the signal spectrum 'contaminating' others. Unfortunately none are perfect and this selection offer a varying trade-off

between main lobe width and highest side band level. For a more detailed discussion see the papers by Harris and Nuttall cited below. Otherwise use your ears.

Alpha: The shape of two of the windows offered (Kaiser and Blackman) can be varied by the use of a variable commonly referred to in the literature as 'alpha' (hence the name of this control). For the Blackman window this can be varied between 0.0 and 1.0 and between 0.0 and 0.001 for the Kaiser windows. It is hoped that a future version of this plug-in will offer a more meaningful interaction with this value. For other window types this parameter has no effect (NA).

Overlap: In order to properly capture all the information in a signal analysis windows need to overlap in time. Some windows (such as the Hann window) need an overlap of 4 to capture the information correctly, others (such as the square root Hann) only need an overlap of 2 to operate correctly (but a higher setting may still give you better sound quality). The overlap can be set to 2 or 4. This parameter has a dramatic effect on the the processing power required by this plug-in. If you are having problems getting the plug-in to run in real time then set this value to 2 and use the square root Hann window (this should halve the processor overhead).

References
"Frequency Shaping of Audio Signals" – C. Penrose, 2001 Proceedings of The International Computer Music Conference

"Extending Musical Mixing: Adaptive Composite Signal Processing" – C. Penrose, 1999 Proceedings of The International Computer Music Conference

"Some Windows with Very Good Sidelobe Behvioour" – A. Nuttall, IEEE Transactions on Acoustics, Speech and Signal Processing [ASSP-29:1](1981)

"On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform", Proceedings of the IEEE [66:1] (1978)

Good introductions to Spectral Processing
The Computer Music Tutorial – C.Roads (chapter 13)

"The Phase Vocoder: A Tutorial" – M.Dolson, The Computer Music Journal [10:4](1986)

Known bugs
None at present!

Please send any bug reports, suggestions or questions to jez@mp3some.co.uk and I'll do my best to look into it and get back to you.

If you are using the plugins and you like them -- don't send any money -- please send us a thank you note: pvnation@leisuresonic.com (Chris Penrose – algorithm design) jez@mp3some.co.uk (Jez Wells – VST implementation). Ta.