Studio Necessities 2.5

User Guide

Acon Digital Media GmbH

Studio Necessities User Guide

Copyright © 2004-2006 Acon Digital Media GmbH.

Table of Contents

1   Introduction

Studio Necessities is a DirectX and VST compatible plug-in pack for use in host applications such as multi-track audio editing or processing software. It consists of six different plug-ins:

- **StudioVerb**, a high quality reverberation simulator
- **StudioModulator**, comprising a phaser effect and time-delay modulation effects such as flanger and chorus
- **StudioEQ**, a six-band full parametric equalizer with six different filter types including peak and shelving filters
- **StudioLimiter**, a brick wall limiter with adjustable look-ahead
- **StudioDynamics**, a versatile dynamic processor
- **StudioDelay**, a multi-tap delay effect

1.1   Requirements

Before you install Studio Necessities, please make sure your computer fulfills the following requirements:

- A Pentium or compatible processor with a CPU clock rate of 350 MHz or more
- A minimum of 64 megabytes RAM
- A Windows-compatible sound card

In addition, you will need the following software installed on your computer:

- Windows 98 SE, Windows Me, Windows 2000 or Windows XP
- DirectX 8.0 or later (the current version at the time this is written is DirectX 9.0c)
- A DirectX or VST plug-in compatible host application

2   The Studio Necessities Plug-Ins

2.1   StudioDynamics

About Dynamic Processing
A dynamic processor is used to alter the dynamic properties of the recording. To understand how a dynamic processor works, imagine a sound engineer trying to maintain as steady a volume level as possible while doing a recording. When the input level increases he pulls down the volume fader, and he pushes it up when the input level decreases. A dynamic processor does the same thing automatically according to its settings, only with a much faster reaction time.

Modern dynamic processors allow an arbitrary mapping of the input levels to the

corresponding output levels. The mapping is visualized as a curve where the horizontal axis represents the input level and the vertical axis represents the output level. A straight line as shown below represents a one-to-one mapping:
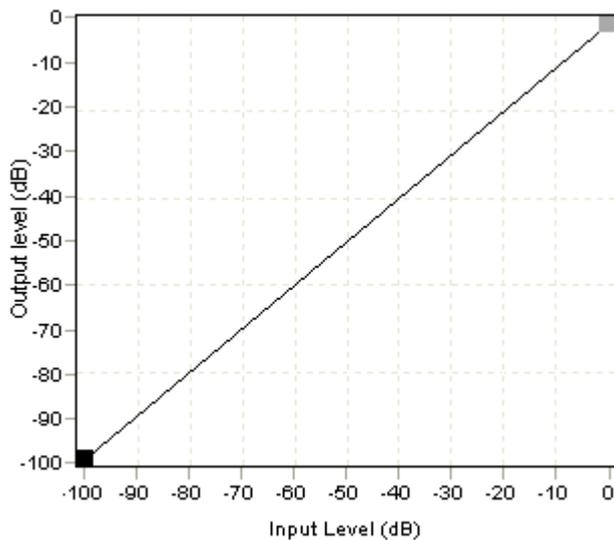
With such a setting, no change is made to the level as it is processed. Changes are made to the dynamics by adding more points to the curve. In the mapping curve below, all signal levels above -20 dB are attenuated so that the output level doesn't exceed -20 dB. This setting would be comparable to a piece of hardware known as a limiter. You can see from the graph below that once the input level reaches –20 dB, the output level isn't going any higher than -20 dB no matter how much higher the input level becomes.
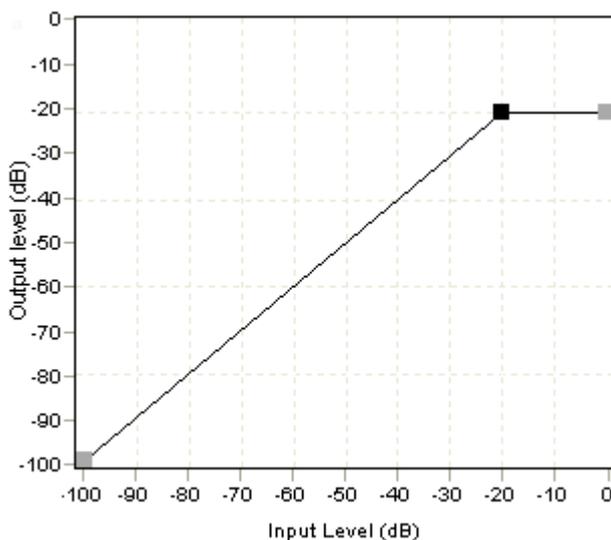
If the dynamic processor changes the level too fast, low frequency signal components will be distorted. How quickly the dynamic processor adapts to changes in the input level is called <u>response time</u>. The response time is divided into the amount of time when the input level rises (the attack time) and when it falls (the release time).

When applying mapping curves that leads to extreme changes in the dynamics, audible artifacts become noticeable (often referred to as "pumping and breathing"). Smoother mapping curves will generally reduce the artifacts of a dynamic processor. <u>Soft kneeing</u> automatically softens the curve to reduce such artifacts. A high level of soft kneeing was used in the mapping curve below.
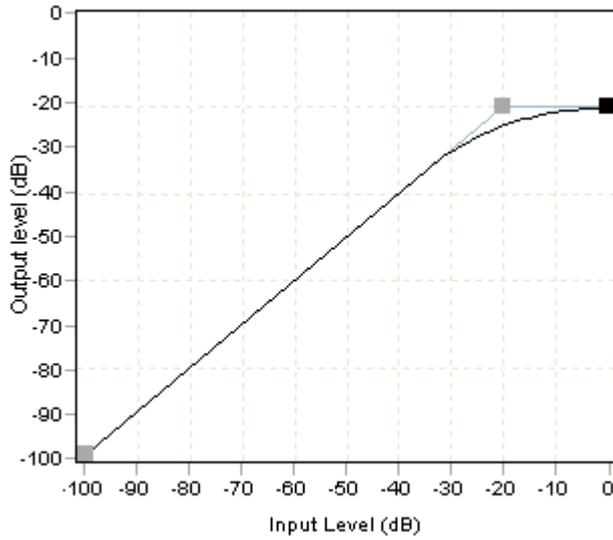
This type of curve would appear as below in the user interface:

User Interface
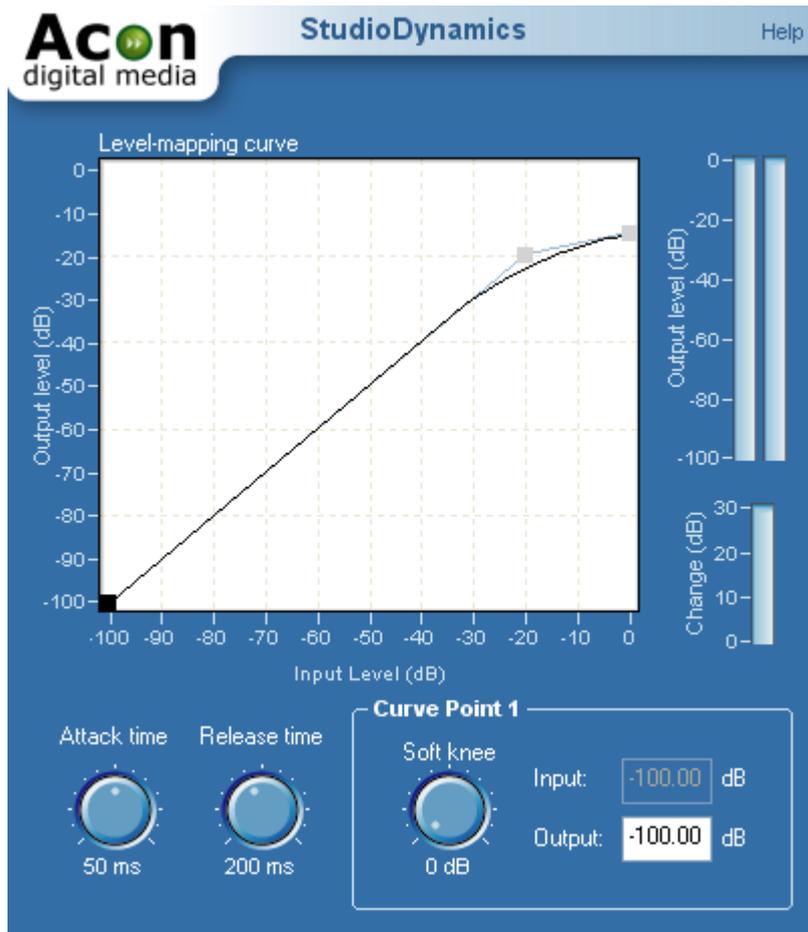
Parameter Settings

- **Level-mapping curve**
  You can add new points to the level-mapping curve by clicking the left mouse button where you want the new point to appear. To move a point, click an existing point and keep the mouse button down while moving the mouse pointer to the new location. You can remove an existing point by clicking the right mouse button on that point.

- **Attack time knob**
  You can use the attack time knob to adjust the response time when the input level of the source material increases. A longer attack time would prevent StudioDynamics from reacting as quickly to increased levels. It can also provide a more natural sound for the output. Clicking the mouse on them can control the knobs by dragging in the direction you want them to turn.

  Like most elements of the interface, the knobs can also be controlled using the arrow keys on your keyboard for greater precision. The *Page Up* and *Page Down* keys are

also effective for larger adjustments, and if you hold any key down it repeats for a continuous adjustment as long as it's held. If you prefer total keyboard control, the *Tab* key on the keyboard can be used to move from one control to the next, and the space bar is available to "click" on buttons in the interface.

- **Release time knob**
  You can use the <u>release time</u> knob to adjust the response time when the input level of the source material decreases. A longer release time can permit audible background noise if all other settings remain the same and the input level drops. To help avoid this, it's sometimes useful to set a threshold, a level below which no amplification is produced at all. A minimalist example is below:
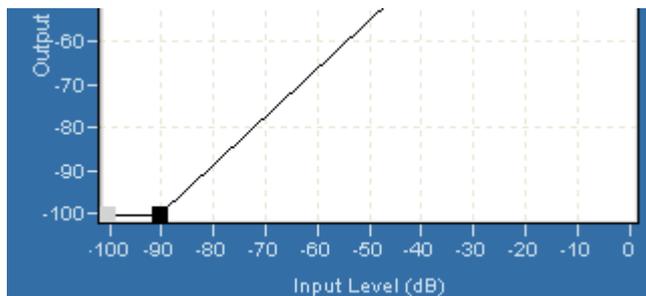
  In the example, you'll notice that the point that we're currently changing is highlighted. With this configuration, no audio below –90 dB would be sent to the output. You can see that no soft knee is used, since any artifacts would be inaudible for all practical purposes. However, this must be used with discretion; it is possible that the program material will sound very different when played back on a different system than the one being used for editing, and what seems to be a subtle modification at low levels can be quite noticeable during playback at high volume.

- **Soft knee knob**
  Each point in the level-mapping curve has adjustable soft kneeing. To adjust the soft kneeing of a point, highlight it by clicking on that point in the display before adjusting the soft knee knob.

- **Input and output edit boxes**
  You can define the input level and output level of an existing point numerically by highlighting the point in the level mapping curve and then entering the input and output levels in dB using the input and output text boxes at the bottom right of the interface.

2.2   StudioLimiter

The Purpose of a Limiter

Limiters belong to the dynamic processing tools and ensures that the signal level doesn't exceed a user selectable boundary while minimizing any possible distortions. To achieve this, limiters introduce a certain latency, called look-ahead. The look-ahead ensures that the limiter can respond in time when the signal level suddenly rises. Many CD producers also use limiters to push the perceived volume to a maximum without introducing audible distortions.

User Interface
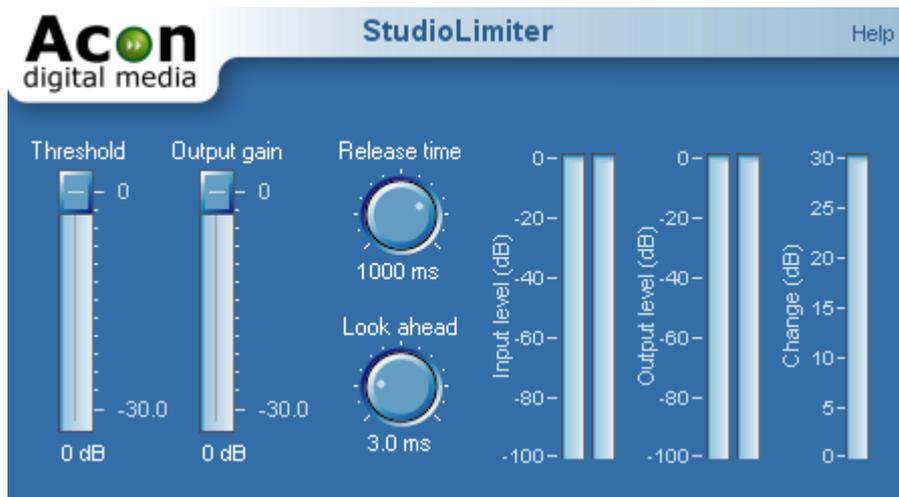
Parameter Settings

- **Threshold**
  All signals above the threshold level will be attenuated so that the threshold level is not exceeded.

- **Output Gain**
  The output gain of the limiter which corresponds to the highest possible output signal level.

- **Release time knob**
  You can use the release time knob to adjust the response time when the input level of the source material decreases. Longer release times will result in a smoother sound.

- **Look ahead knob**
  You can adjust the number of milliseconds the limiter uses to determine its internal

gain settings with the look-ahead knob. The limiter will have a latency equal to the look ahead time.

2.3   StudioEQ

About Equalizers

Equalizers are devices that allow frequency dependent volume changes. In home stereo gear they would be comparable to bass and treble controls, except that equalizers are more highly specialized for adjusting and fine-tuning the frequency spectrum amplitude in audio material. A parametric equalizer consists of filters with variable center frequencies and adjustable bandwidths, as opposed to graphic equalizers where the center frequencies and bandwidths are fixed.

StudioEQ would be comparable to a "paragraphic" equalizer, which combines the best features of both of the basic types of equalizers. You have total control over the frequency center point, bandwidth, and amount of change that a parametric equalizer provides, while being able to view the result in a graphic format.

User Interface
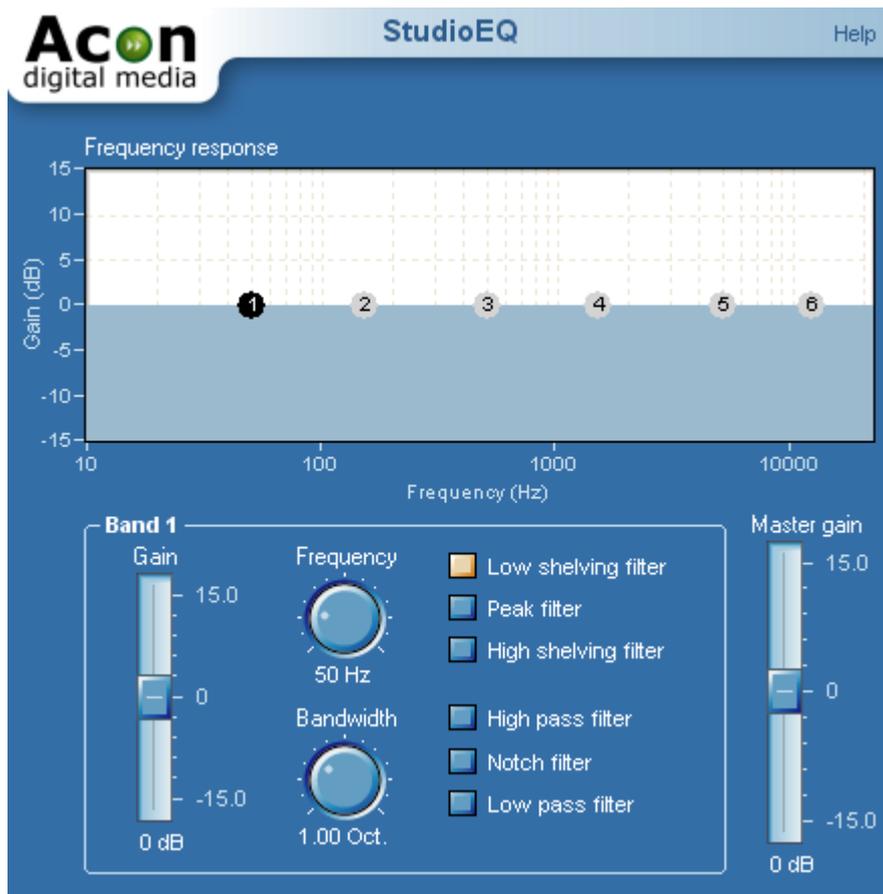
Parameter Settings

In StudioEQ each band can be set to one of the following filter types:

- **Peak filter**
  The peak filter increases or decreases the level of the frequency band surrounding the center frequency.

- **Low shelving filter**
  The low shelving filter increases or decreases the level of the frequencies below the cutoff frequency.

- **High shelving filter**
  The high shelving filter increases or decreases the level of the frequencies above the cutoff frequency.

- **High pass filter**
  The high pass filter removes frequencies below the cutoff frequency.

- **Notch filter**
  The notch filter removes frequencies surrounding the center frequency.

- **Low pass filter**
  The low pass filter removes frequencies above the cutoff frequency.

Each band has a gain and a bandwidth parameter in addition to the center frequency and the filter type. The gain specifies the volume or level of the frequency band. If the gain is set to 0 dB, there will be no change in level compared to the original input. By selecting a positive gain, the frequency band is boosted. A negative gain leads to an attenuation of the frequency band. The bandwidth parameter is of use only when the filter type is set to peak filter. The bandwidth is specified in <u>octaves</u>, whereas one octave band equals the frequency range of one octave on the piano keyboard.

- **Frequency response graph**
  The frequency response graph visualizes the frequency response of current equalizer settings. The visualization is calculated mathematically and updated as the parameters are changed.

  Each of six equalizer bands is visualized with a small circle containing the number of the band. You can select a band by clicking its corresponding circle. By keeping the mouse button down while moving the mouse cursor, you can change the gain and the center frequency of the bands. The numbered circle affected by any changes to the settings is highlighted as shown above. In the example, band three is the one affected by any changes made.

- **Gain slider**
  You can also use the gain slider to adjust the gain of the currently selected band. The gain can be set to a value between -15 and +15 dB.

- **Frequency knob**
  The frequency knob lets you set the center frequency (for peak filters) or cutoff frequency (for shelving filters) of the selected band. To enter the frequency numerically, click the frequency knob and tap the space bar.

- **Bandwidth knob**
  The bandwidth knob lets you set the bandwidth of the selected band. The bandwidth determines the width of the frequency band surrounding the center frequency in a peak filter. The bandwidth has no effect on shelving filters.

- **Filter type (Low shelving, peak, high shelving, high pass, notch and low pass)**
  You can set the desired filter type for the selected band with these radio buttons. As with other standard radio buttons, selecting one deselects any others.

- **Master gain**
  The master gain is the master volume for the audio being equalized and is frequency independent. You can adjust the volume using the master gain slider if the overall output level is too high or too low. This lets you compensate for the changes in level that can result from large changes made in the frequency response curve using the equalizer, particularly in the lower frequencies.

2.4   StudioVerb

About Reverberation
Reverberation occurs when sound is produced in an enclosed acoustical environment. Even outdoors, there is likely to be some level of reverberation, however subtle. The sound propagates through the air before it arrives at the listener, but the sound is also reflected when it hits the walls. Due to the propagation time, these reflections arrive at the listener later than the sound from the direct path. There are usually so many reflections that no distinct echoes are distinguishable, but rather a smoothly decaying sound.

The first reflections, usually called <u>early reflections</u>, are important cues for our perception of an acoustical environment. For that reason, most digital reverberation units differentiate between early reflections and the <u>dense reverberation</u>. The early reflections are simulated by a small number of non-repeating echoes, whereas a complex network of delays simulates the dense reverberation, with feedback coefficients carefully calculated to match the reverberation time (acoustic size) of a room.

An important tool when analyzing the reverberation of real rooms is the <u>echogram</u>, also known as impulse response. Playing a short impulsive sound and recording the resulting reverberation creates an echogram. Below is an example of an echogram with early reflections and dense reverberation.
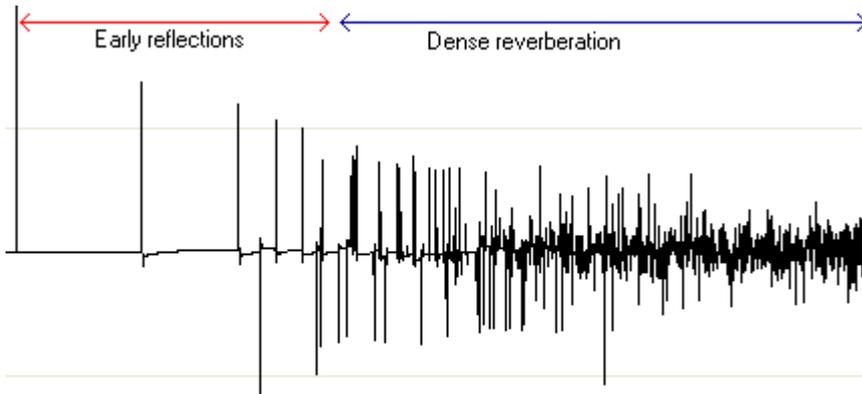
User Interface

Parameter Settings

- **Reverberation time slider**

  You can use the <u>reverberation time</u> slider to adjust the reverberation time of the dense reverberator. The reverberation time is specified by the number of seconds before the reverb tail (fall-off) drops below 1/1000 (or 0.001) of its initial amplitude.

- **Pre-delay slider**

The pre-delay slider allows you to adjust the time before the dense reverberation begins.

- **Input filter graph**
  You can use the input filter graph to apply filtering on the input signal of the early reflections simulator and the reverberation module. The frequency response of the input filter is visualized by the blue curve. The filter consists of a low and a high shelving filter. To adjust the filter cutoffs and gains, click the circle labeled "1" for the high pass filter or "2" for the low pass filter and keep the mouse button down while moving the mouse cursor.

- **High frequency damping knob**
  High frequencies decay faster than low frequencies when sound propagates through air. You can adjust the level of high frequency damping using the high frequency damping knob.

- **Room size knob**
  Use the room size knob to adjust the size of the simulated acoustic environment. It is important that the room size matches the reverberation time. In open air, small rooms have shorter reverberation times than large rooms. If the two parameters are mismatched, the reverberation will not sound natural. However, this unnatural sound may be an effect that's desirable for your particular program content.

- **Dry level slider**
  The amount of unprocessed signal in the output mix.

- **Early reflections slider**
  The amount of early reflections in the output mix.

- **Reverb level slider**
  The amount of dense reverberation in the output mix.

2.5   StudioModulator

The StudioModulator plug-in consists of three different effects:

- **Chorus**
  Chorus adds warmth to recordings by dividing the audio source into several voices with a random vibrato. These artificially created voices are then mixed together. The result is a digital approximation of a literal chorus of people or instruments.

- **Flanger**
  The flanging effect occurs when two sources playing the exactly same recording with a short time delay are mixed together. The result is that some frequencies are cancelled, while others are amplified. Changing the time delay between the two sources will result in other frequencies being cancelled or amplified. (The name comes from years ago when studio engineers using open-reel tape would manually turn or hold one reel or "flange" to create this effect, distorting the tape speed past the playback heads.)

- **Phaser**
  The phaser effect uses several notch filters with a time-varying center frequency to create a sweeping or electronic "whooshing" effect.

User Interface
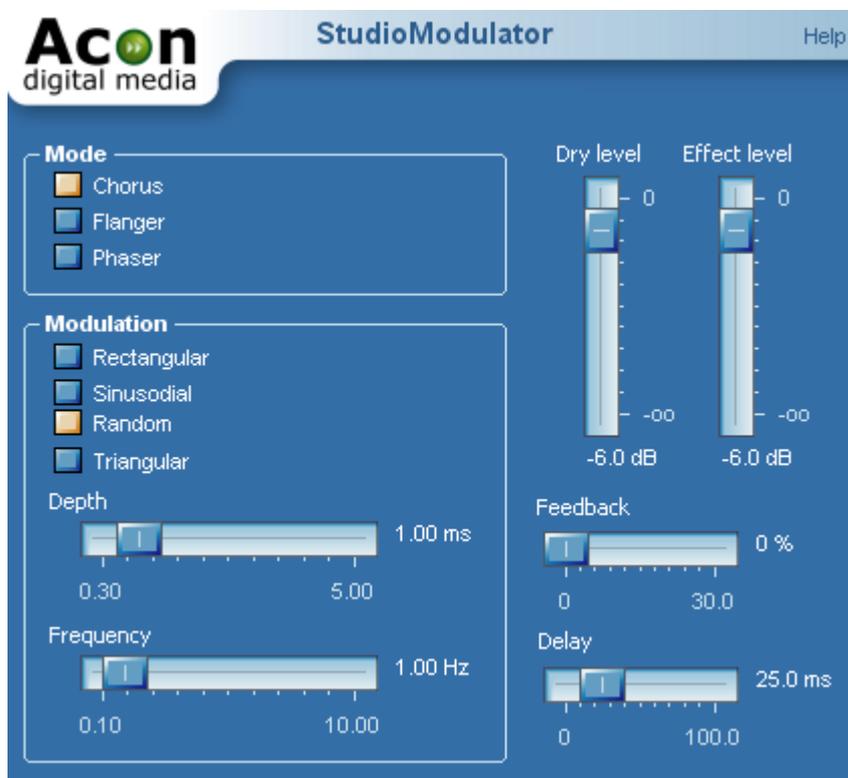
Parameter Settings

- **Mode selection**
  You can select the modulator mode by clicking Chorus, Flanger or Phaser from the list.

- **Modulation selection**

The delay (in the chorus and the flanger mode) or the notch center frequency (in phaser mode) can be modulated by one of four different modulation sources. To change the modulation source, select sinusoidal, triangular, square, or random from the list.

- **Modulation depth slider**
  Use the modulation depth slider to set the depth or amount of the modulation.

- **Modulation frequency slider**
  The modulation frequency slider allows you to set the frequency or the speed of the modulation function. A higher modulation frequency will result in faster changes in the tone quality.

- **Feedback slider**
  The feedback slider controls the attenuation in the internal delayed feedback loop.

- **Delay slider**
  You can add an additional fixed delay to the chorus or the flanger effect by adjusting the delay slider.

- **Dry Level slider**
  Use the dry level slider to adjust the amount of unprocessed signal in the output mix.

- **Effect Level slider**
  Use the effect level slider to adjust the amount of processed signal in the output mix.

2.6   StudioDelay

StudioDelay is a <u>multi-tap</u> delay effect. Multi-tap means that you can add several delays (up to eight of them in StudioDelay) with arbitrary delay times and gains. StudioDelay offers two different timing modes, the BPM (Beats per minute) mode or the milliseconds mode. In the BPM mode, the time delay of each tap is specified in beats.

User Interface

Parameter Settings

- **Echogram**
  This depicts the gain versus time for the configured taps. You can add new delay taps by clicking the left mouse button where you want the new delay tap to appear. To move a delay tap, click an existing point and keep the mouse button down while moving the mouse pointer to the new location. You can remove an existing delay tap by clicking the right mouse button on that point.

- **Range**
  This represents the amount of time shown in the echogram.

- **Delay**
  This lets you specify the amount of time audio is delayed before it is sent to the output. In the echogram, it affects the horizontal position of the tap.

- **Gain**
  The individual tap is present by this amount in the output. In the echogram, it affects the vertical position of the tap.

- **Feedback**
  The feedback percentage specifies the amount of attenuation since the last delay interval.

- **Low pass**

  There is a low pass filter in the feedback loop for each tap. You can change the cut-off frequency of the low pass filter using the low pass knob. The low pass can be enabled or disabled by clicking the low pass check box above the low pass knob.

- **Channel bouncing**

  This provides for a bouncing stereo delay. If the input audio has too small a stereo image (the acoustic "width" and "depth" of the output) you can add more sensation of space to the recording through channel bouncing. This also adds an indicator to the echogram:

  The blue mark shown above depicts the delay time for the left channel output, whereas the red mark depicts the delay time for the right channel output.

- **Dry Level**

  The amount of unprocessed signal in the output mix.

- **Effect Level**

  The amount of processed signal in the output mix.

- **Time Mode**

  Select either beats per minute for the BPM mode or milliseconds.

- **Tempo**

  Here you can specify the tempo by entering the number of beats per minute.

- **Snap**

  To make it easier to adjust the delay of each tap without loosing the alignment to the tempo, you can optionally turn on the snap mode. You can snap to quarters, eights, sixteens or their triplet counterparts by clicking the arrow down and clicking the desired mode from the drop-down list.

2.7   In Conclusion

Acon Digital Media has spent many long hours creating the Studio Necessities plug-ins. Properly configured, these plug-ins permit you to dramatically shape or alter your audio, customizing it in spectacular ways. It's our hope that you will find a multitude of uses for them.

Happy editing!

Index