Mix & Mastering Suite

User Guide

Acon Digital AS

Mix & Mastering Suite User Guide

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# Table of Contents

1   Introduction

Thanks for using *Acon Digital Max & Mastering Suite*. The suite consists of six plug-ins for audio mixing and mastering that were designed for ultimate signal transparency:

- **Equalize 2** is a parametric equalizer. Unlike other equalizers, you can freely adjust not only center frequency, gain and bandwidths, but also the filter slope for each band. The filter slope can be set anywhere from 3 dB to ultra-sharp 120 dB per octave. You can operate Equalize as a zero latency plug-in when using the minimum phase mode, choose to preserve the phase relationships in the linear phase mode or use the unique mixed phase mode. The mixed phase mode allows you to set the latency freely in the range 5 to 120 milliseconds while preserving the phase relationships as far as possible. That gives a unique control over potential pre-ringing artifacts which is a common problem with linear phase filtering.
- **Dynamics** is a combined compressor, expander or gate. The look-ahead can be set anywhere from 0 to 30 milliseconds and the intelligent algorithm ensures a band-limited gain signal which reduces potential distortion to a minimum even with minimal attack and release times. You can use up to 4 times oversampling to further reduce distortion. The side chain signal can be either internal or external and there's a full blown side chain equalizer integrated with low cut, low shelf, peak, high shelf and high cut filters. The complexity of algorithms is not reflected in the intuitive user interface which provides graphical visualizations of all aspects of the processing.
- **Multiband Dynamics** is the multiband extension of Dynamics which allows dynamic processing in up to four separate bands. You can freely set the cross-over frequencies and even the filter slopes can be set freely in the range 6 and 48 dB per octave. *Multiband Dynamics* operates in linear or mixed phase modes to avoid phase issues between the processed bands. The unique mixed phase mode reduces pre-ringing that can degrade transients after linear phase processing.
- **Limit** is a transparent limiter with intelligent look-ahead in the range 0 to 30 milliseconds that reduces distortion to a minimum. The limiting is performed in a two step process with a pre-compressor follower by the actual peak limiting algorithm. You can use up to 4 times oversampling for true peak limiting and to further reduce distortion.
- **Dither** is a dithering plug-in with flexible perceptual noise shaping. You can adjust the maximum gain and the filter length of the frequency response of the noise shaping filter is visualized graphically.
- **Verberate 2** is an algorithmic reverb that simulates real acoustical spaces with a new degree of realism.

1.1 Purchase and Authorization

The plug-ins in the *Acon Digital Mix & Mastering Suite* will run in demo mode the first time you open it from your audio editor or digital audio workstation (DAW). The demo mode is fully functional with exception of short passages with muted audio output at irregular intervals. The demo version can be unlocked by purchasing a license key from Acon Digital (see Acon Digital online shop). When a plug-in is opened in the demo mode, a dialog box appears where you can choose to continue with the demo version or enter a license key. There is also a purchase button which directs you to the Acon Digital online shop.

If you have purchased a license key from Acon Digital, please enter you name, company name if applicable and the license key you obtained. The Authorize button will remain deactivated until the license key has been accepted. After having successfully authorized *Acon Digital Mix & Mastering Suite*, the demo dialog box will not appear again.
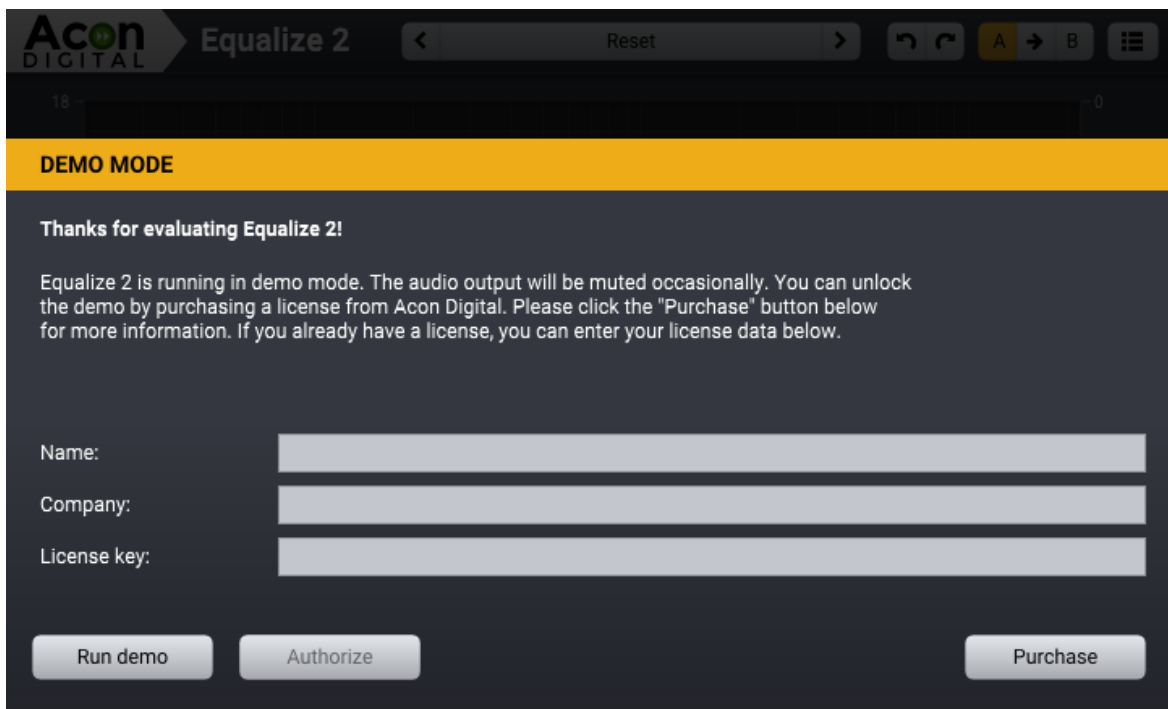
The demo dialog box appears when starting any of the plug-ins in the Acon Digital Mix & Mastering Suite prior to authorization.

1.2 Requirements

Before you install *Acon Digital Mix & Mastering Suite*, please make sure your computer fulfills the following requirements:

PC Version (Windows)

- Windows 10 / 8 / 7
- Intel Core i3 or AMD multi-core processor (Intel Core i5 or faster recommended)
- 1366 x 768 display resolution (1920 x 1080 or higher recommended)
- 1 GB RAM (4 GB or more recommended)
- 1 GB free HD space
- A host application that is compatible with VST (32 bit or 64 bit), VST3 or AAX (Pro Tools 10.3.5 or higher)

Macintosh Version (OS X)

- OS X 10.8 or later
- 1 GB RAM (4 GB or more recommended)
- 1 GB free HD space
- A host application that is compatible with AU, VST, VST3 or AAX (Pro Tools 10.3.5 or higher)

2 Host Integration

You can access the plug-ins in the *Acon Digital Mix & Mastering Suite* from your host application (audio editor or DAW) of choice. The plug-ins are available as a 32 or 64 bit VST, VST3 or AAX plug-in on the Windows platform or as 64 bit VST, VST3, AU or AAX plug-ins on Mac OS X. Some host applications will require a rescan and possibly adding the *Acon Digital Mix & Mastering Suite* installation directory to the list of VST directories. Please consult the manual for your host application for further details.

3 Using the Plug-ins

The plug-ins in *Acon Digital Mix & Mastering Suite* have some properties in common such as preset management, A/B comparisons and bypassing. You can find a common set of buttons and controls in the header section of the included plug-ins and these are described below.

Preset Manager
Acon Digital Mix & Mastering Suite is shipped with a set of factory presets that serve as a starting point for further adjustments. You can browse through preset categories and presets as well as create and manage your own presets using the preset management section:
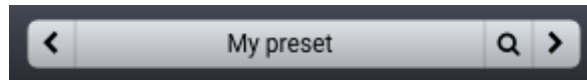
The preset management section available in all the integrated processors.

You can browse through the presets using the arrow buttons. Alternatively, you can click the current preset name and a drop-down menu appears. Click the button with the magnifying glass icon to search for a preset. Matching presets appear in a drop-down list as you type.

You can also save your own presets by choosing "Save user preset file..." from the menu. A file chooser dialog box appears where you can enter the name of the preset you wish to save. You can create sub folders and place your preset files inside, and these will appear as categories in the user presets.
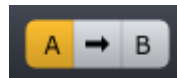
Undo and Redo
You can undo (or redo) changes to the parameter settings by clicking the circular arrow back or forward buttons:

Undo and redo buttons

A / B Comparisons
It is frequently useful to be able to quickly compare different parameter settings. You can do this using the A / B comparison buttons:

The A / B comparison buttons allows you to quickly compare different settings

You can keep two independent sets of parameter settings, the A and B settings, and switch between them using the corresponding buttons. The arrow button copies the settings from A to B or the other way around depending on which parameter set that is currently active.

The Processor Menu
The last button in the header section displays the processor menu:

You can click the processor menu button for the processor specific menu

The processor menu allows you to access the processor help topic directly along with other processor specific features.

Using Sliders to Adjust Parameters

Horizontal, vertical and rotary sliders (or knobs) are frequently used in the plug-in user interfaces. Clicking with the mouse on a knob and moving up or down will allow you to modify the setting. You can also use the scroll wheel of your mouse. For more precise control, you can hold down the Ctrl key on your keyboard at the same time as you use your mouse to modify the setting.

3.1   Equalize

Introduction

Equalize 2 is a versatile, user friendly and great sounding equalizer with several unique features. Unlike other equalizers, you can freely adjust not only center frequency, gain and bandwidth, but also the filter slope for each band. The filter slope can be set anywhere from 3 to ultra sharp 120 dB per octave. Needless to say, you can operate Equalize as a zero latency processor when using the minimum phase mode or choose to preserve the phase relationships in the linear phase mode. Equalize goes one step further, though, and introduces the unique mixed phase mode that allows you to set the latency freely in the range 5 to 120 milliseconds while preserving the phase relationships as far as possible. That gives a unique control over potential pre-ringing artifacts which is a common problem with linear phase filtering.

Latency values below 20 milliseconds ensure that any pre-ringing is masked by the temporal masking of the human hearing while preserving the time-alignment across the audible frequency range. Great care has been taken to provide a user interface that is straight forward to use. Band parameters can be adjusted using handles directly in the graphical representation of the frequency response, including bandwidths and filter slopes. A flexible real-time analyzer lets you monitor every aspect of the processing. You can easily switch between full, mid, side, left or right channel processing for each band and Equalize automatically routes the audio signal internally to ensure the best results and lowest possible latency.

User Interface

The graphical user interface of *Acon Digital Equalize 2* is designed to hide the complexity of the plug-in and provide an efficient workflow as well as intuitive control over the plug-in parameters.

Adding and Removing Bands

You can add up to 12 individual bands in Equalize. Each equalizer band represents a filter type, which can be either low cut, low shelf, bell, notch, high shelf or high cut. To add an additional band, click the + button below the curve display or double click in the curve display. A bullet shaped handle appears in the curve that you can move around using the mouse. The currently selected band also shows additional handles that you can use to manipulate the bandwidth (if applicable) and filter slope. You can remove the band by clicking the x button or by double clicking the handle.

Soloing and Bypassing Bands

You can solo or temporarily bypass an equalizer band in order to monitor its effect on the audio signal. You can enable the solo mode by pressing the Ctrl key while moving an equalizer band handle. The Shift key enables the bypass band mode. Alternatively, you can use the buttons to enable or disable the solo and bypass modes:

🎧 : Solo mode

🚫 : Bypass mode

Band Parameters

The parameters that are related to a specific band are placed within the band group and the header indicates which band is currently active. The parameters apply to the currently active band only. To change the currently active band you can either click its bullet handle in the frequency response visualization or use the arrow buttons to browse through the active bands.

Frequency (Hz)

The center (for peak and notch filters) or threshold frequency of the currently active equalizer band in Hertz.

Band gain (dB)

The gain of the currently active equalizer band in decibels. This parameter is not available for high or low cuts, or the notch filter.

Bandwidth (oct.)

The bandwidth of the currently active equalizer band in octaves. This parameter is only available for peak and notch filters.

Slope (dB/oct.)

The filter slope of the currently active equalizer band in decibels per octave. This controls the steepness of the filter.

Gain to Bandwidth Link

The perceived bandwidth of an equalizer band is dependent on the gain setting. You can link the bandwidth to the gain setting, so that the bandwidth is automatically adjusted when you change the gain to preserve the perceived bandwidth.

⚙ : Click this button to activate or deactivate the gain to bandwidth link

Filter Type Buttons

You can choose between eight different filter types:

⌐ : Low cut, removes frequency content below the band frequency
➤ : Low shelf, boosts or attenuates frequency content below the band frequency
⊶ : Peak filter, boosts or attenuates frequency content around the band frequency
✕ : Tilt filter, tilts the frequency content around the band frequency so that one side get boosted while the other side gets attenuated
⊓ : Bandpass filter, removes all frequency content except for the band surrounding the band frequency
⋎ : Notch filter, removes frequency content around the band frequency
◄ : High shelf, boosts or attenuates frequency content above the band frequency

$\diagdown$: High cut, removes frequency content above the band frequency

Channel Mode Buttons

You can choose any of the following channel modes for each equalizer band individually:

L : Apply to left channel only

M : Apply to mid channel only. The mid channel is the sum of the input channels multiplied by a scaling factor.

$\textbf{ⓞⓞ}$: Apply to both channels

S : Apply to side channel only. The side channel is the difference of the input channels multiplied by a scaling factor.

R : Apply to right channel only

Global Parameters

Master gain (dB)

The master gain controls the overall gain of the equalizer in decibels. Equalize can automatically adjust the overall gain to compensate for any gain changes in the equalizing process. This is based on an average distribution typically found in music and cannot be completely accurate. To activate the automatic gain compensation, click the button labelled A to the left of the master gain knob.

Phase mode

You can choose between three different phase modes in Equalize. Since Equalize is a 'clean' equalizer in the sense that it doesn't apply any saturation or similar effects to simulate analog artifacts, the resulting filter applied by Equalize is fully described by the amplitude response (which is the visualization of the filter curve response that is shown in the upper part of the plug-in window) and the phase response.

Minimum phase: This is the way analog equalizers process audio and has the great advantage of zero latency and no pre-ringing. Sharp filters can cause time alignment problems and crosstalk between tracks can cause unpredictable results when tracks are processed separately.

Linear phase: The advantage of the linear phase mode is that you can process track independently even if there is crosstalk between them. Linear phase processing introduces high latencies and can cause pre-ringing effects that aren't desirable.

Mixed phase: The unique mixed phase mode combines the best of the two phase distributions. You can adjust the latency and hence the maximum

pre-ringing time so that it is masked by the temporal masking effect in the human auditory system.

You can read more about the phase modes in the [Phase Modes chapter](#) 16.

Latency (ms)

If the mixed phase mode is activated, you can choose the latency and maximum pre-ringing time freely in the range 5 ms to 120 ms. Low latencies will result in a phase distribution more like minimum phase while higher latencies result in a more linear phase distribution. We recommend to keep this value around 20 ms to ensure that the pre-ringing is masked by the temporal masking effect of the human auditory system while still preserving the time alignment of the signal component in the audible frequency range.

Note: The latency is doubled if you combine mid or side bands with left or right channel bands and use the mixed phase mode.

Analyzers and the Frequency Response Visualization

Equalize visualizes the frequency response of the current equalizer settings and contains two separate spectrum analyzers that you can set up to monitor the effects of the processing.

Analyzer 1 & 2

You can choose to analyze different signal sources using the drop-down lists under the Analyzer 1 and Analyzer 2 headers. Either input or output signals can be analyzed and you can also choose if the left, right, mid or side signal should be analyzed.

Range Buttons

You can change the range of the frequency response visualization or the spectrum analyzers by clicking the buttons at the lower end of the level axes. A drop down list appears with the alternatives.

Adjusting the Analyzer Frequency Range

The frequency range of the equalizer curve can be altered conveniently thus making it possible to zoom in on specific frequency ranges for more precise control. Move the mouse cursor to either the left or right edge of the frequency axis below the frequency curve editor. The mouse cursor turns into a left-to-right arrow. Now click and keep the left mouse button pressed while adjusting either the start or end frequency of the visible frequency range. You can also move the visible frequency range up or down moving the mouse to the center of the frequency axis. The mouse cursor turns into a hand. Click and keep the left mouse button pressed while moving the visible frequency range up or down in frequency.

Piano Keyboard Display

You optionally let Equalize display a piano keyboard in the bottom of the equalizer curve editor and you can find this setting in the <u>Preferences</u> 14.

Acon Digital Equalize 2 with the piano keyboard display activated.

The piano keyboard makes it easy to map the band frequencies to the frequency of a specific piano key. Click a key on the piano keyboard to assign the frequency of the currently active band to the frequency of the clicked key.

Adding and Removing Bands

You can add up to 12 individual bands in Equalize. Each equalizer band represents a filter type, which can be either low cut, low shelf, bell, notch, high shelf or high cut. To add an additional band, click the + button below the curve display or double click in the curve display. A bullet shaped handle appears in the curve that you can move around using the mouse. The currently selected band also shows additional handles that you can use to manipulate the bandwidth (if applicable) and filter slope. You can remove the band by clicking the x button or by double clicking the handle.

Soloing and Bypassing Bands

You can solo or temporarily bypass an equalizer band in order to monitor its effect on the audio signal. You can enable the solo mode by pressing the Ctrl key while moving an equalizer band handle. The Shift key enables the bypass band mode. Alternatively, you can use the buttons to enable or disable the solo and bypass modes:

🎧 : Solo mode

🚫 : Bypass mode

Band Parameter Settings

The parameters that are related to a specific band are placed within the band group and the header indicates which band is currently active. The parameters apply to the currently active band only. To change the currently active band you can either click its bullet handle

in the frequency response visualization or use the arrow buttons to browse through the active bands.

- **Frequency (Hz)**
  The center (for peak and notch filters) or threshold frequency of the currently active equalizer band in Hertz.

- **Band gain (dB)**
  The gain of the currently active equalizer band in decibels. This parameter is not available for high or low cuts, or the notch filter.

- **Bandwidth (oct.)**
  The bandwidth of the currently active equalizer band in octaves. This parameter is only available for peak and notch filters.

- **Resonance (dB)**
  Controls the resonance in decibels at the threshold frequency for cut and shelving filters. Resonance occurs in analog filters and results in a boost around the threshold frequency.

- **Slope (dB/oct.)**
  The filter slope of the currently active equalizer band in decibels per octave. This controls the steepness of the filter.

- **Gain to Bandwidth Link**
  The perceived bandwidth of an equalizer band is dependent on the gain setting. You can link the bandwidth to the gain setting, so that the bandwidth is automatically adjusted when you change the gain to preserve the perceived bandwidth.

  ⚙ : Click this button to activate or deactivate the gain to bandwidth link

- **Filter Type Buttons**
  You can choose between six different filter types:

  ⌐ : Low cut, removes frequency content below the band frequency
  ⊃ : Low shelf, boosts or attenuates frequency content below the band frequency
  ⊖ : Peak filter, boosts or attenuates frequency content around the band frequency
  ⋁ : Notch filter, removes frequency content around the band frequency
  ⊏ : High shelf, boosts or attenuates frequency content above the band frequency
  ⌐ : High cut, removes frequency content above the band frequency

- **Channel Mode Buttons**

You can choose any of the following channel modes for each equalizer band individually:

L   : Apply to left channel only

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⫴   : Apply to both channels

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R   : Apply to right channel only

Global Parameters

• **Master gain (dB)**
The master gain controls the overall gain of the equalizer in decibels. Equalize can automatically adjust the overall gain to compensate for any gain changes in the equalizing process. This is based on an average distribution typically found in music and cannot be completely accurate. To activate the automatic gain compensation, click the button labelled A to the left of the master gain knob.

• **Phase mode**
You can choose between three different phase modes in Equalize. Since Equalize is a 'clean' equalizer in the sense that it doesn't apply any saturation or similar effects to simulate analog artifacts, the resulting filter applied by Equalize is fully described by the amplitude response (which is the visualization of the filter curve response that is shown in the upper part of the plug-in window) and the phase response.

Minimum phase:
This is the way analog equalizers process audio and has the great advantage of zero latency and no pre-ringing. Sharp filters can cause time alignment problems and crosstalk between tracks can cause unpredictable results when tracks are processed separately.

Linear phase:
The advantage of the linear phase mode is that you can process track independently even if there is crosstalk between them. Linear phase processing introduces high latencies and can cause pre-ringing effects that aren't desirable.

Mixed phase:
The unique mixed phase mode combines the best of the two phase distributions. You can adjust the latency and hence the maximum pre-ringing time so that it is masked by the temporal masking effect in the human auditory system.

- **Latency (ms)**

  If the mixed phase mode is activated, you can choose the latency and maximum pre-ringing time freely in the range 5 ms to 120 ms. Low latencies will result in a phase distribution more like minimum phase while higher latencies result in a more linear phase distribution. We recommend to keep this value around 20 ms to ensure that the pre-ringing is masked by the temporal masking effect of the human auditory system while still preserving the time alignment of the signal component in the audible frequency range.

  Note: The latency is doubled if you combine mid or side bands with left or right channel bands and use the mixed phase mode.

Analyzers and the Frequency Response Visualization

Equalize visualizes the frequency response of the current equalizer settings and contains two separate spectrum analyzers that you can set up to monitor the effects of the processing.

- **Analyzer 1 & 2**

  You can choose to analyze different signal sources using the drop-down lists under the Analyzer 1 and Analyzer 2 headers. Either input or output signals can be analyzed and you can also choose if the left, right, mid or side signal should be analyzed.

- **Drop (dB/s)**

  The drop parameter sets how quickly the analyzer adapts to lower signal levels and is specified in decibels per second.

- **Range Buttons**

  You can change the range of the frequency response visualization or the spectrum analyzers by clicking the buttons at the lower end of the level axes. A drop down list appears with the alternatives.

3.1.1   Equalize Preferences

Equalize 2 introduces a new *Preferences* page where you can adjust settings like language, tooltips and analyzer resolution. Click the menu button in the upper right corner and choose **Show Preferences...** to show the *Preferences* page:
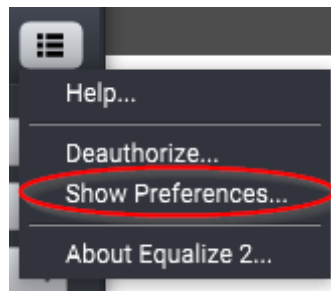
Select the Show Preferences command as indicated above to show the Preferences page.

The following *Preferences* page appears:
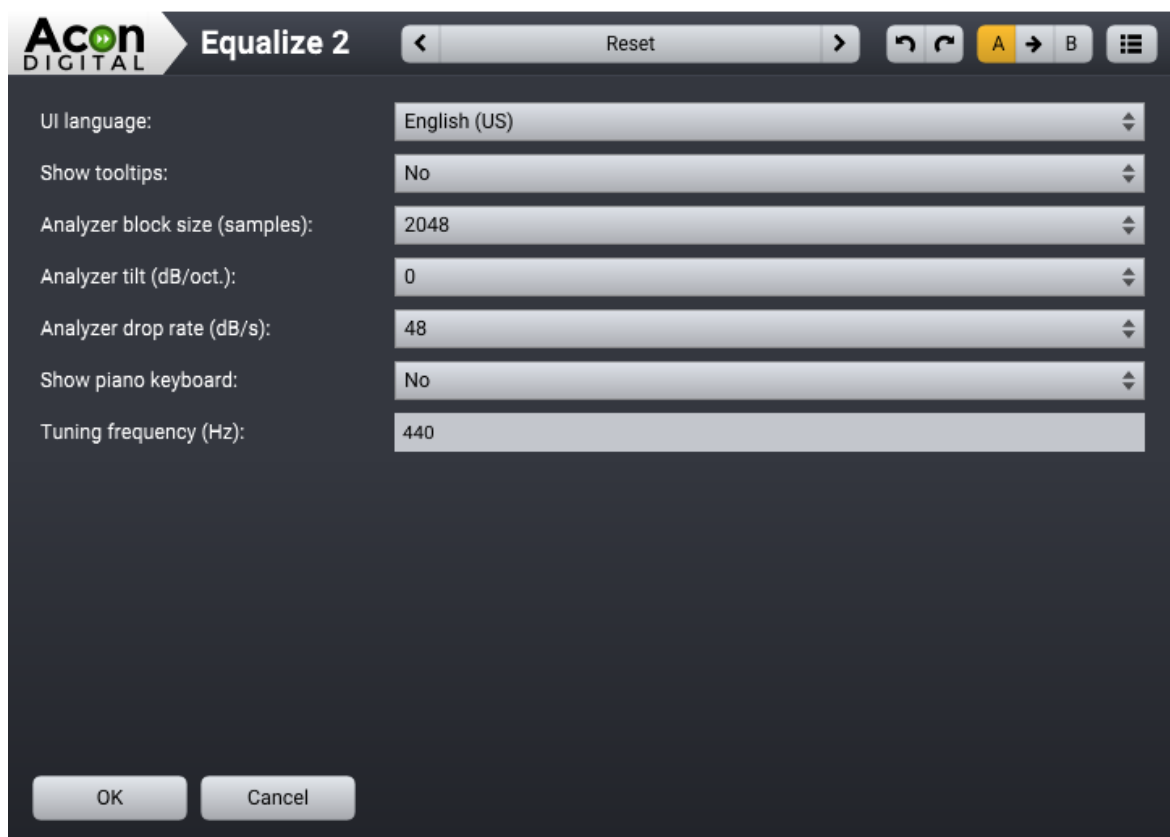
The new Preferences page shows language, tooltips, analyzer and piano keyboard settings.

Settings available from the Preferences Page

UI language: Allows you to switch the user interface language. You will need to reopen the plug-in before the change has an effect.

Show tooltips: Allows you to enable or disable the tooltips that appear when you rest the mouse cursor over a user interface control for a while.

|	
Analyzer block size (samples):	Affects the frequency versus time resolution in the analyzer. Larger block sizes will result in a better frequency resolution, but will at the same time reduce the time resolution.
Analyzer tilt (db/oct.):	You can tilt the frequency spectrum with an adjustable attenuation per octave. White noise will appear like a flat horizontal line when the tilt is set to 0 dB/octave. Pink noise can be set to appear like a flat horizontal line with the tilt set to 3 dB/octave.
Analyzer drop rate (dB/s):	Defines how quickly the analyzer adapts to lower signal levels and is specified in decibels per second.
Show piano keyboard:	Shows or hides the new piano keyboard display.
Tuning frequency (Hz):	The tuning frequency for the piano keyboard display. You can enter the center frequency of the note A4 in Hertz (440 is the most common tuning).

3.1.2   Equalize Phase Modes

The mixed phase mode in *Acon Digital Equalize 2* is a new and unique feature and the difference between the available phase modes are described in detail in this chapter. The effect of a clean equalizer that doesn't emulate analog artifacts like saturation or noise can be fully described by the amplitude and the phase response. The amplitude response is what Equalize visualizes in the curve editor in the upper part of the plug-in window. The phase response is not visualized, however, because its impact on the processed signal wouldn't be visualized intuitively in a spectrum analyzer. However, the importance of the phase response is revealed when we measure the so called impulse response of the equalizer. The impulse response is measured by feeding the equalizer with a Dirac impulse. In the digital world with sampled time series, a Dirac impulse is nothing more than a zero signal consisting of one single full scale sample. The output is what we describe as the impulse response.
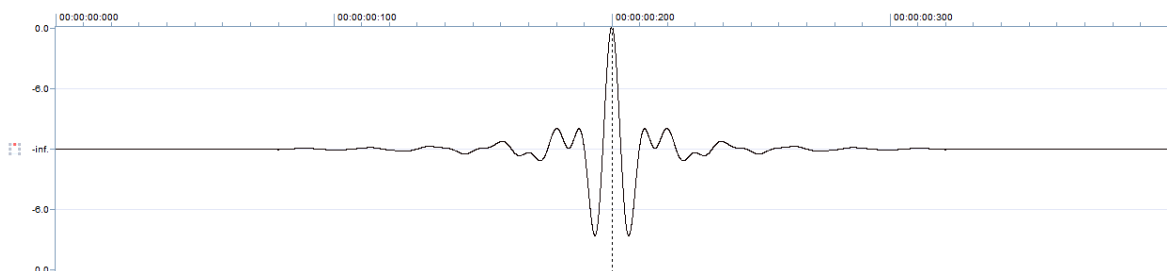
The illustrations below visualize the impulse responses obtained from a steep band pass filter (100 - 200 Hz) using the three phase modes that are available in Equalize along with the input signal at the top:
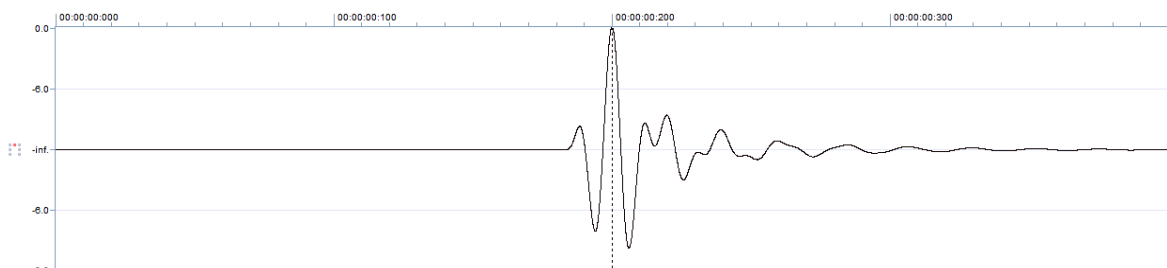
Input test signal consisting of a single pulse at 200 ms (Dirac impulse).
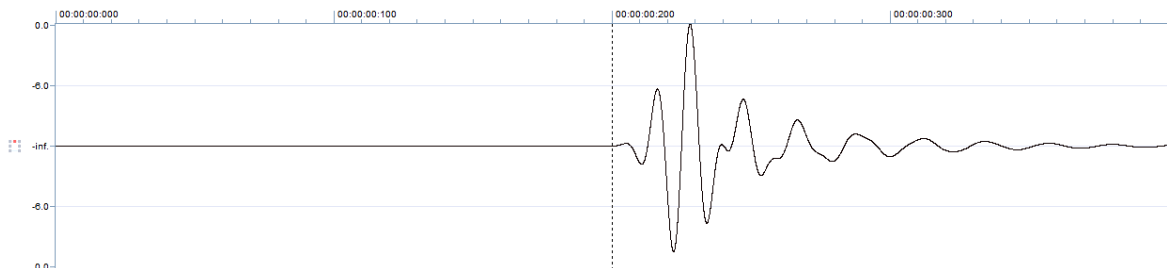
***Minimum phase mode**. Notice that the peak amplitude is shifted backwards in time.*

***Linear phase mode**. The extensive non-zero signal prior to the peak is noticeable and disturbing. Another disadvantage is the long latency introduced by the filtering.*

***Mixed phase mode**. The pre-ringing is limited to fit within the temporal masking of the human auditory system and the latency introduced is short. The amplitude peak is perfectly time aligned.*

Traditionally, only the minimum and the linear phase modes have been available. The minimum phase mode is the only mode that ensures zero latency and is the way analog equalizers work. The downside is that the time alignment between different frequency components is lost.

The linear phase mode was introduced with the advent of digital processing, but the pre-ringing effect can distort transient rich material severely. Linear phase filters show advantages when processing tracks separately and there is crosstalk between the tracks. In such cases, linear phase processing has predictable results whereas minimum phase processing can alter the frequency response in unintentional ways.

The new mixed phase mode is a unique feature in Equalize and allows you to set the latency and hence the maximum pre-ringing time. The human auditory system masks signals before and after an impulsive noise, something which is called temporal masking. The masking is stronger after the impulsive noise than before, which is why extensive pre-ringing is undesirable. By choosing latencies around 20 ms in Equalize, the pre-ringing is masked by our hearing. At the same time, the maximum peak is perfectly time aligned.

3.2   Dynamics

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 you to set a ratio between the input levels and the output levels. This ratio 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 1:1 ratio.
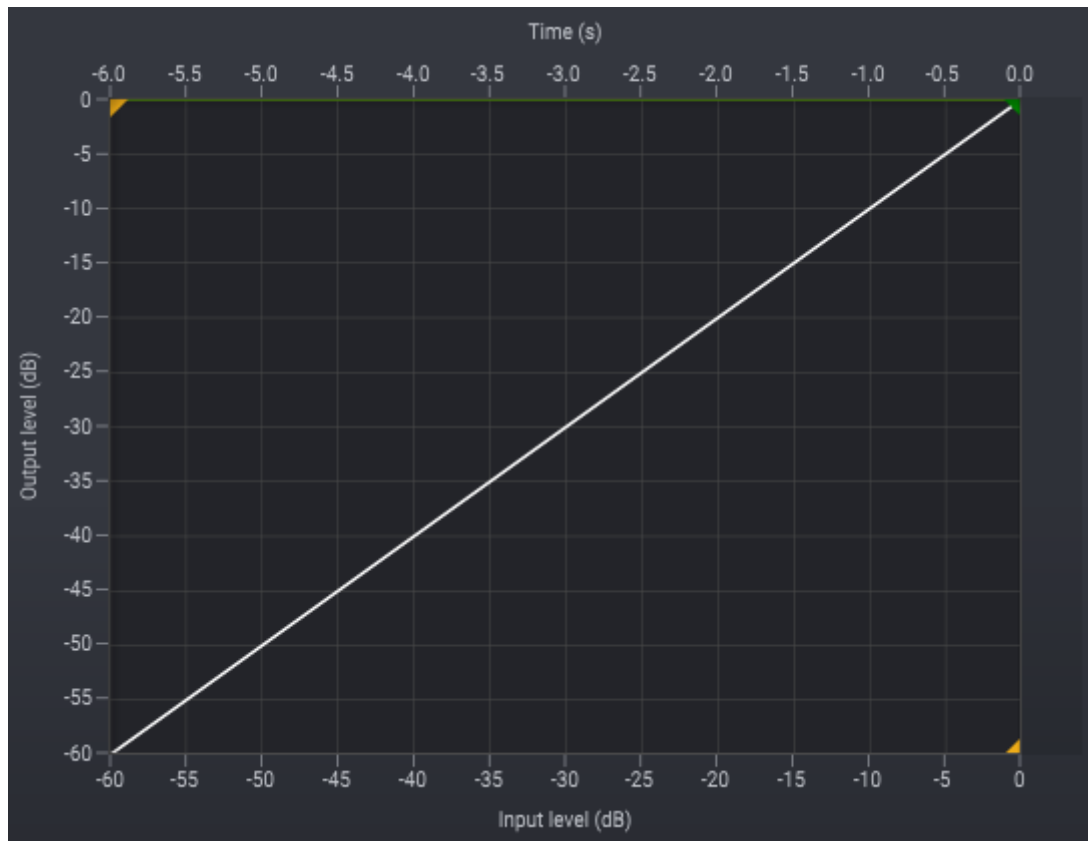
With such a setting, no change is made to the level as it is processed. Changes are made to the dynamics by altering the ratio and the threshold. In the example below, all signal levels above -20 dB are attenuated with a 100:1 ratio, so that the output level only raises 1dB when the input raises 100dB above the threshold. 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 is barely going any higher than -20 dB, even when the input level increases.
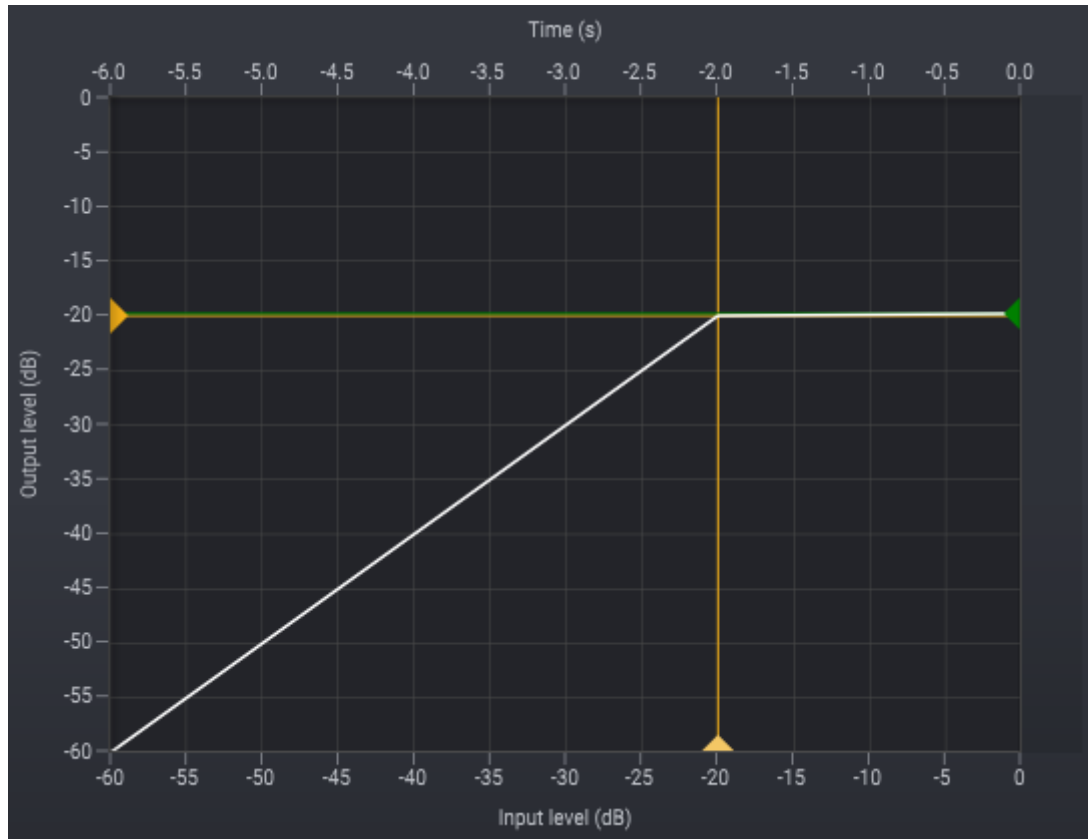
If the dynamic processor changes the level too fast, low frequency signal components might become distorted. How quickly the dynamic processor adapts to changes in the input level is called response time. The response time is divided into the amount of time when the input level rises (the attack time) and when it falls (the release time).

When applying compressor ratios that lead to extreme changes in the dynamics, audible artifacts might become noticeable (often referred to as "pumping and breathing"). Smoother compression curves will generally reduce the artifacts of a dynamic processor. Soft kneeing automatically softens the curve to reduce such artifacts. A high level of soft kneeing was used in the image below.
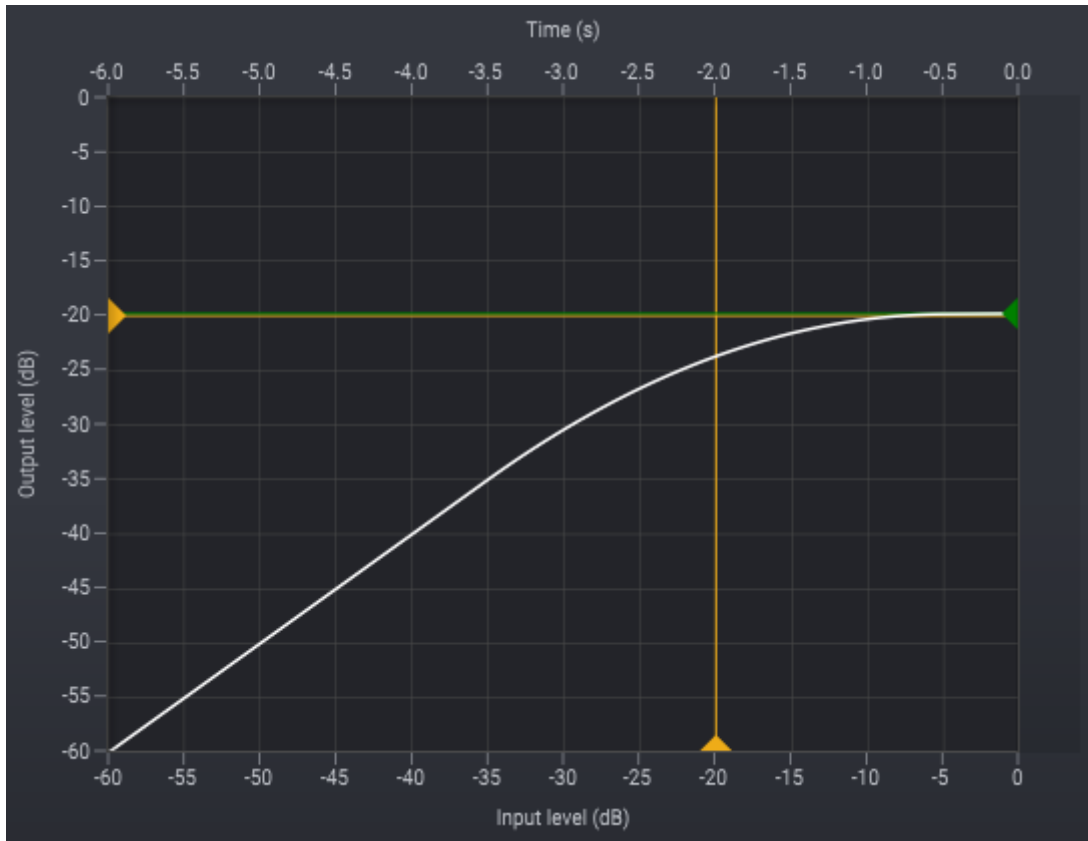
User Interface

Parameter Settings

- **Threshold level (-60 dB to 0 dB)**
  When the input level exceeds the set threshold, the compressor starts to apply compression. For example, if the threshold is set at -12 dB, it will only respond to incoming audio that exceeds -12 dB. However, as soon as you activate the Soft knee, it's possible that compression takes place even before the input level exceeds the set threshold, due to the nature of the soft knee.

- **Ratio (0.01:1 to 100:1)**
  By setting up a ratio, you decide how much gain reduction should take place for every dB the input level exceeds above the set threshold. For example, if the threshold is set on 2.00:1, there will only be a 1 dB output increase for every 2 dB the input level exceeds above the set threshold. You could also say that the compressor will do 1 dB of gain reduction for every 2 dB the input level exceeds the set threshold.

It's also possible to set a ratio that is below 1.00:1. This basically turns the compressor into a gate or expander. Audio levels below the threshold will now be reduced.

- **Range (0 to 60 dB)**
  The range control sets the maximum gain reduction of the dynamic processor.

- **Soft knee (0 dB to 24 dB)**
  A soft knee allows you to create a smoother compression curve, which can help to prevent or reduce audible artifacts. With the soft knee set to 0 dB, there is no soft knee applied to the compression curve. When the soft knee is set to 8 dB for example, the compression curve will be smoothly interpolated for input levels in the range -8 dB to +8 dB relative to the threshold level.

- **Attack time (0.01 ms to 200 ms)**
  You can use the attack time knob to adjust how quickly the compressor will react to incoming audio that is above the threshold. The faster the attack time, the faster the compressor will react. If compression is applied on a snare for example which has a very loud attack, setting up a very fast attack time might help to reduce the loud attack of the snare, as the compressor can almost instantly apply gain reduction. However, in many scenarios a slightly slower attack time might provide you with a more natural sounding compression behavior, for example during mastering.

- **Release time (ms)**
  You can use the release time knob to adjust how quickly the compressor will stop processing audio that is no longer above the threshold. A fast release time can greatly enhance the sustain/decay of certain softer elements of the audio, but it can also introduce a "pumping" effect. Slow release times (>200ms) often result in a more smoother/natural compression behavior.

- **Auto-Release button**
  You can enable auto-release mode by clicking the button labeled "A" on the left side of the numerical entry field for the release time. In auto-release mode, the release time specifies the maximum release time and the dynamic processor shortens the release time for sharp transients.

- **Hold time (0 ms to 500 ms)**
  As soon as audio is below the set threshold, the release time kicks in. However, the hold time allows the compressor to postpone the activation of the release time for up to 500ms, which might provide you with a smoother compression behavior in certain audio material. Although you are free to experiment with the hold function, we would like to suggest to use a hold time that is significantly shorter than the release time.

- **Make-up (0 dB to 32 dB)**
  You can use the make-up knob to compensate for the level reduction caused by the compressor. It's possible to set the make-up in auto mode, by pressing the A button in front of the knob. When the A button is yellow, the auto make-up is activated and the make-up knob is disabled.

- **Latency (0 ms to 15 ms)**
  It is possible to reduce harmonic distortion of low frequency content even with short attack and release times if the dynamic processor is allowed to examine the signal ahead of time. The downside is a slightly increased latency. You can adjust the maximum allowed latency in milliseconds.

- **Channel Linking (0% to 100%)**
  With the channel linking set to 100%, the amount of gain reduction will be the same for the left and right channels, even if there is a difference in level between these channels. The more you dial the knob away from 100%, the more the channels will be independently treated, with complete independence between the channels once you have a 0% channel linking. Channel linking values below 100% can result in shifts in the stereo image. During mastering, the best results will be most likely achieved with the channel linking set between 70% and 100%.

- **Mix (0% to 100%)**
  The mix parameter controls the unprocessed and processed signal mix. Normally, this will be set to 100%, lower settings allow for convenient parallel compression.

- **Oversampling (off, 2x, 4x)**
  The internal sample rate of the dynamic processor can be multiplied by 2 or 4, depending on the sample rate of the project. For example, if you have a project with a sample rate of 48 kHz, setting the oversampling to 4x will result in an internal sample rate of 192 kHz.

Side Chain Filtering

The signal that is used to calculate the gain signal in dynamic processing is referred to as the *side chain* signal. You can filter the side chain signal in *Dynamics* so that the dynamic processing is more sensitive in certain frequency regions. This works similar to an equalizer. Click the button labeled *Side chain filter* in the bottom left corner of the Dynamics user interface to open (or close) the side chain filtering settings:

The side chain filter in Dynamics allows you to filter the side chain signal similar to the way an equalizer works.

You can click one of the filter buttons to toggle the filter and the following filter types are available:

- ⌐ : Low cut, removes frequency content below the band frequency
- ⊃ : Low shelf, boosts or attenuates frequency content below the band frequency
- ⊸ : Peak filter, boosts or attenuates frequency content around the band frequency
- ⊂ : High shelf, boosts or attenuates frequency content above the band frequency
- ⌐ : High cut, removes frequency content above the band frequency

The *Side chain filter* button turns orange as soon as filtering is active.

3.3   Multiband Dynamics
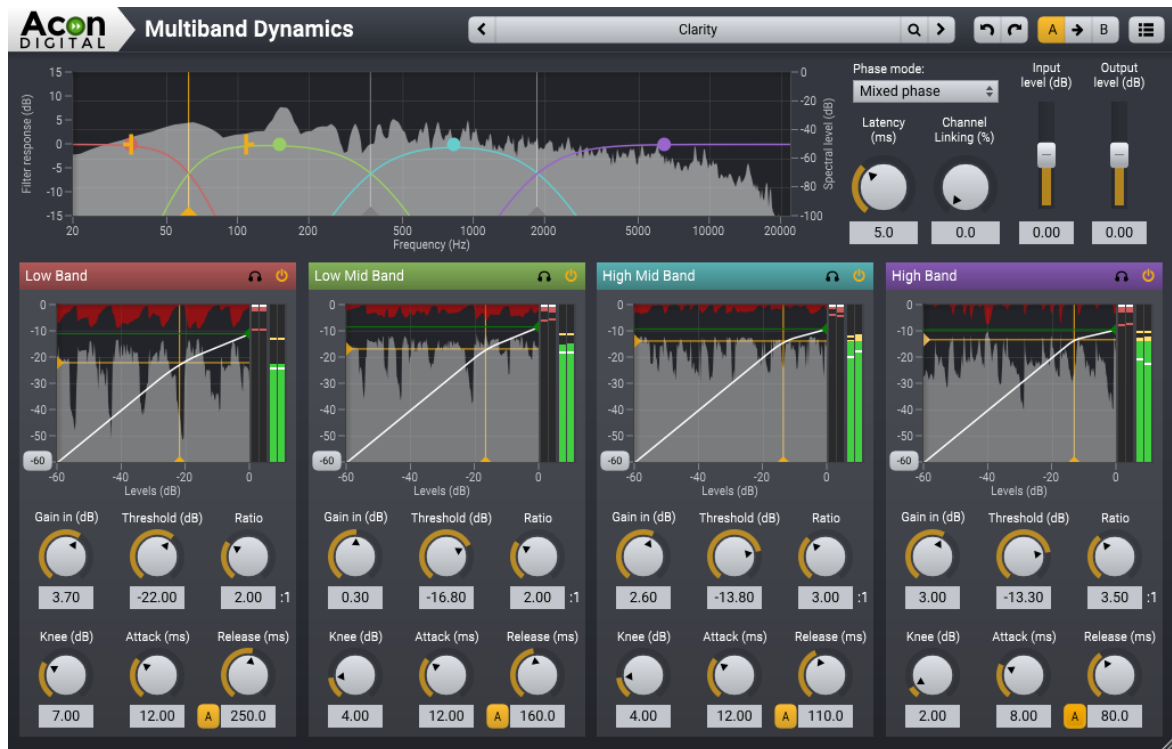
About Multiband Dynamics

Multiband Dynamics features four independent dynamic processors in parallel, each working on a different frequency band of the audible spectrum. This allows you to apply dynamic processing with different settings for each of the four bands, making it a perfect tool for addressing certain problem frequencies in your mix, like low frequency rumble or sibilance for example.

There are four bands in *Multiband Dynamics,* which means that there are three cross-over frequencies. These are between the four frequency bands called *Low Band*, *Low Mid Band*, *High Mid Band* and *High Band*. Each band can be set between 20 Hz and 20 kHz, but a cross-over point can never cross another cross-over point. In the example below you see the three cross-overs set at 100 Hz, 400 Hz and 4 kHz.
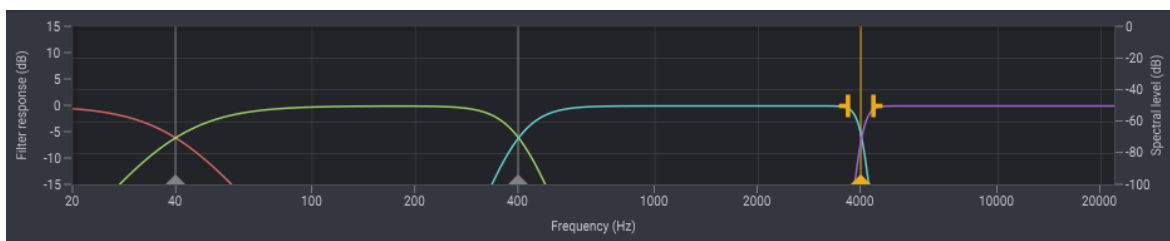
In the same example you also see that each cross-over uses a different slope setting. The first cross-over uses a slope of 6 dB/octave, it's the slowest slope that you can set for each band. The second cross-over uses a pretty common 24 dB/octave slope, while the last cross-over uses the steepest slope that you can set for each band, which is 96 dB/octave. To set your preferred slope, select the desired cross-over marker to make the slope setting visible. You can drag either one of the slope markers to modify the slope to your liking. A steeper slope provides you with more precision at the cost of potential pre-ringing artifacts, which can distort transient rich material. Therefore, we encourage you to start with a medium slope and modify to taste.

User Interface

Global Parameter Settings

- **Phase mode**

  You can choose between two different phase modes in *Multiband Dynamics*:

Linear phase:	The advantage of the linear phase mode is that it completely avoids phase issues when the bands are recombined after processing. However, linear phase processing introduces pre-ringing effects that distorts transients and also increases latency.
Mixed phase:	The unique mixed phase mode combines the best of minimum and linear phase processing. You can adjust the latency and hence the maximum pre-ringing time so that it is masked by the temporal masking effect in the human auditory system. At the same time, phase issues that plague multiband processing with minimum phase filters are reduced to a minimum.

  You can read more about the phase modes in the <u>Phase Modes chapter</u> 16 .

- **Latency (ms)**

  If the mixed phase mode is activated, you can choose the latency and maximum pre-ringing time freely in the range 3 ms to 20 ms. Low latencies will result in a phase distribution more like minimum phase while higher latencies result in a more linear

phase distribution. We recommend a settings around 5 ms which is sufficiently short to mask pre-ringing.

- **Channel Linking (%)**
  With the channel linking set to 100%, the amount of gain reduction will be the same for the left and right channels, even if there is a difference in level between these channels. The more you dial the knob away from 100%, the more the channels will be independently treated, with complete independence between the channels once you have a 0% channel linking. Channel linking values below 100% can result in shifts in the stereo image.

- **Input level (-16 dB to 16 dB)**
  You can use the input level to adjust the level prior to all processing.

- **Output level (-16 dB to 16 dB)**
  You can use the output level to adjust the level after all processing.

Band Parameter Settings

The following parameters are available for each of the four bands. Press and hold the *Shift* key while using the sliders to modify a parameter for all four bands.

- **Band solo**
  You can solo the output from a band by clicking the solo button ( )

- **Band on / off toggle**
  Bands can be enabled or disabled with the power on / off button

- **Input gain (-16 dB to 16 dB)**
  You can use the input gain to increase of decrease the signal level at the band's audio input.

- **Threshold level (-60 dB to 0 dB)**
  When the input level exceeds the set threshold, the compressor starts to apply compression. For example, if the threshold is set at -12 dB, it will only respond to incoming audio that exceeds -12 dB. However, as soon as you activate the Soft knee, it's possible that compression takes place even before the input level exceeds the set threshold, due to the nature of the soft knee.

- **Ratio (0.01:1 to 100:1)**
  By setting up a ratio, you decide how much gain reduction should take place for every dB the input level exceeds above the set threshold. For example, if the threshold is set on 2.00:1, there will only be a 1 dB output increase for every 2 dB the input level

exceeds the set threshold. You could also say that the compressor will do 1 dB of gain reduction for every 2 dB the input level exceeds the set threshold.

It's also possible to set a ratio that is below 1.00:1. This basically turns the compressor into a gate or expander. Audio levels below the threshold will now be reduced.

- **Soft knee (0 dB to 24 dB)**
  A soft knee allows you to create a smoother compression curve, which can help to prevent or reduce audible artifacts. With the soft knee set to 0 dB, there is no soft knee applied to the compression curve. When the soft knee is set at 8 dB for example, the compression curve will be smoothly interpolated for input levels in the range -8 dB to +8 dB relative to the threshold level.

- **Attack time (0.01 ms to 200 ms)**
  You can use the attack time knob to adjust how quickly the compressor will react to incoming audio that is above the threshold. The faster the attack time, the faster the compressor will react. If compression is applied on a snare for example which has a very loud attack, setting up a very fast attack time might help to reduce the loud attack of the snare, as the compressor can almost instantly apply gain reduction. However, in many scenarios a slightly slower attack time might provide you with a more natural sounding compression behavior, for example during mastering.

- **Release time (ms)**
  You can use the release time knob to adjust how quickly the compressor will stop processing audio that is no longer above the threshold. A fast release time can greatly enhance the sustain/decay of certain softer elements of the audio, but it can also introduce a "pumping" effect. Slow release times (>200ms) often result in a more smoother/natural compression behavior.

- **Auto-Release button**
  You can enable auto-release mode by clicking the button labeled "A" on the left side of the numerical entry field for the release time. In auto-release mode, the release time specifies the maximum release time and the dynamic processor shortens the release time for sharp transients.

- **Output Level (-16 dB to 16 dB)**
  You can use the level handle (colored circle) in the filter response display to compensate for the level increase or decrease trough the dynamic processing.

3.4   Limit

The Purpose of a Limiter

Limit is a two-stage dynamic processor, designed to transparently increase the perceived loudness of audio, while at the same time making sure no audio clipping occurs. First, there is a compressor stage (Pre-Compressor) that helps to keep the number of exceeding peaks within an acceptable range, which helps to prevent or reduce audible distortion. The second stage is the actual peak limiting stage (Peak Suppression) which makes sure that no audio will pass through above the threshold, also known as *brickwall* limiting. In this stage, peaks are reduced with a very quickly responding algorithm in the most transparent way as possible.

User Interface
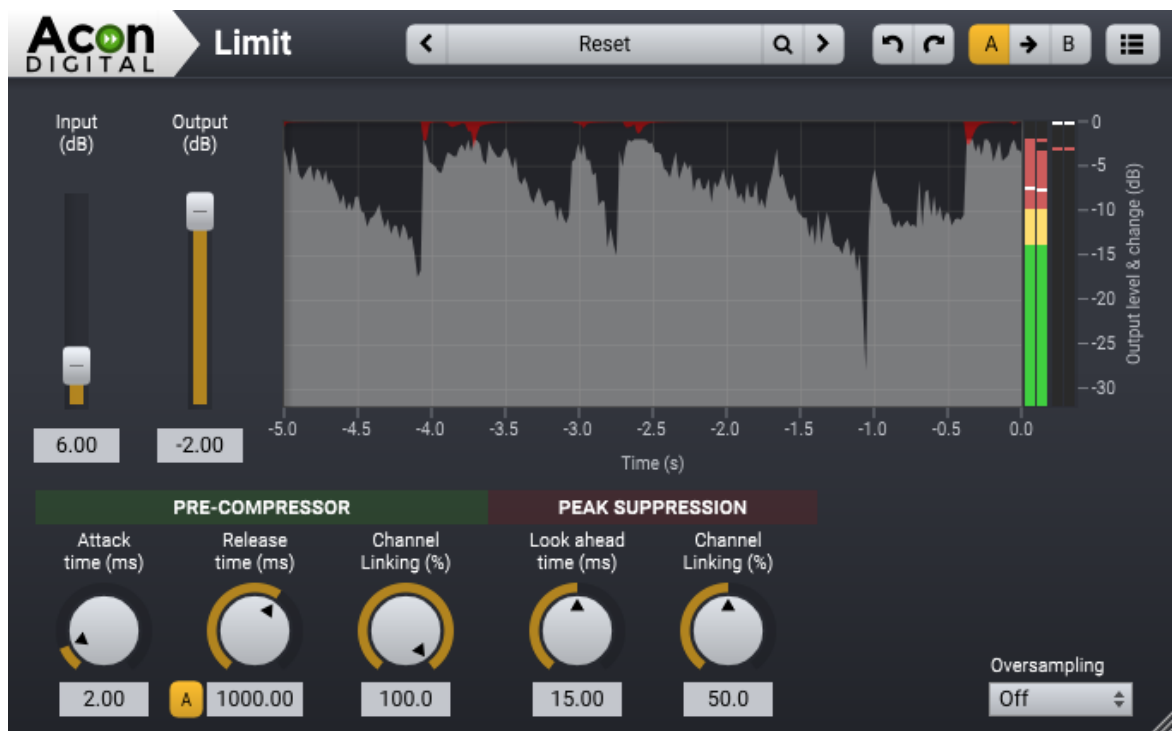
Parameter Settings

- **Input (0 dB to +32 dB)**
  The internal threshold in Limit is always 0 dB. In order to increase the perceived loudness, you need to increase the input level of the audio. As soon as the input audio will exceed the internal threshold, gain reduction will be applied in order to keep the audio from clipping. Excessive amounts of gain reduction can lead to audible distortion

and/or a pumping effect. For mastering duties, we recommend to keep the amount of gain reduction to a minimum.

- **Output (0 dB to -32 dB)**
  The output of Limit is set to 0 dB by default, but can be used to compensate for the increased level when the input is boosted.

- **Attack time (0.01 ms to 500 ms)**
  You can use the attack time knob to adjust how quickly the Pre-Compressor stage will react to incoming audio that is above the internal threshold. The faster the attack time, the faster the compressor will react. A fast attack time will be more transparent, at the cost of a slightly lower increase in perceived loudness. A slow attack time will increase the perceived loudness, at the risk of introducing audible distortion or a pumping effect. For mastering duties, we recommend to start with an attack time of 40 ms and adjust it to taste.

- **Release time (1 ms to 5000 ms)**
  You can use the release time knob to adjust how quickly the compressor will stop processing audio that is no longer above the internal threshold. A fast release time will greatly increase the perceived loudness at the risk of audible distortion. A slow release time will be more transparent, at the cost of a lower increase of perceived loudness. For mastering duties, we recommend to start with a release time of 150 ms and adjust it to taste.

- **Auto-Release button**
  You can enable auto-release mode by clicking the button labeled "A" on the left side of the numerical entry field for the release time. In auto-release mode, the release time specifies the maximum release time and the pre-compressor shortens the release time for sharp transients.

- **Pre-Compressor Channel Linking (0% to 100%)**
  With the channel linking set to 100%, the amount of gain reduction will be the same for the left and right channels, even if there is a difference in level between these channels. The more you dial the knob away from 100%, the more the channels will be independently treated, with complete independence between the channels once you have a 0% channel linking. Channel linking values below 100% can result in shifts in the stereo image. During mastering, the best results will be most likely achieved with the channel linking set between 70% and 100%.

- **Look ahead time (0 ms to 15 ms)**
  Limit uses a sophisticated look ahead algorithm to ensure the highest possible transparency. The higher this value, the more transparent the processing will be. Please

keep in mind that look ahead time will also introduce latency, which will normally be automatically compensated for by your host application.

- **Peak Suppression Channel Linking (0% to 100%)**
  With the channel linking set to 100%, the amount of limiting will be the same for the left and right channels, even if there is a difference in level between these channels. The more you dial the knob away from 100%, the more the channels will be independently treated, with complete independence between the channels once you have a 0% channel linking. Although channel linking values below 100% can result in shifts in the stereo image, it is not as evident as during the pre-compressor stage. During mastering, the best results will be most likely achieved with the channel linking set between 50% and 100%.

- **Oversampling (off, 2x, 4x)**
  The internal sample rate of Limit can be multiplied by 2 or 4, depending on the sample rate of the project. For example, if you have a project with a sample rate of 48 kHz, setting the oversampling to 4x will result in an internal sample rate of 192 kHz.

3.5   Dither

Introduction to Dither

Whenever you reduce the resolution (bit depth) of an audio signal you will introduce truncation errors, which can be a very unpleasant artifact if audible. The quantization noise is correlated to the audio signal which we perceive as more disturbing that uncorrelated noise. Dither introduces low level random noise to de-correlate the noise. Additionally, it is possible to alter the frequency distribution of the noise signal. The human ear has a frequency dependent sensitivity and it makes sense to "move" the noise to frequency regions where the ear is less sensitive. This process is called noise shaping and the *Dither* plug-ins offers detailed control over the noise shaping process.

User Interface

Parameter Settings

- **Target resolution**

  Here you have to set the target resolution. If you want to change the resolution from 24 bit to 16 bit for example, you need to select 16.

- **Enable noise shaping**

  Click to enable or disable the noise shaping filter

- **Dither level (%)**

  Dither amplitude in percent of 1 LSB (least significant bit). This should be at 100% for complete de-correlation of quantization noise.

- **Filter length (ms)**

  This allows you to set the length of the noise shaping filter in milliseconds. Longer filters allow more accurate shaping, but comes at CPU cost.

- **Max. noise shaping (dB)**

  This sets he maximum gain allowed in the noise shaping filter. The higher the setting, the less audible the added noise will be, by moving the noise up in the frequency spectrum. However, too much energy concentration in the high frequency bands should be avoided as this can cause issues with certain digital to analog convertors.

3.6   Verberate

The graphical user interface of *Acon Digital Verberate 2* is designed to give quick and intuitive access to all the parameter settings defining the sonic quality of the reverb.

The graphical user interface of the Verberate 2 plug-in in action.

Reverb Parameters

- **Dry level (dB)**
  The amount of dry (unprocessed) signal to send to the output specified in decibel. You can use the toggle button to exclude the dry signal completely. You can lock the relation between the dry and reverb levels by clicking the lock button between the dry and reverb level sliders.

- **Reverb level (dB)**
  The amount of reverberation signal to send to the output specified in decibel. You can use the toggle button to exclude the late reverberation completely.

- **ER boost (dB)**

The amount of early reflections to send to the output specified in decibel relative to the reverb level. The early reflections are important for our perception of distance to the sound source. Increasing the early reflection level gives the impression of getting closer to the source. You can use the toggle button to exclude the early reflections completely.

- **Algorithm**
  Verberate 2 offers two late reverb algorithms. We recommend the new *Vivid Hall* algorithm, but you can also choose the *Legacy Hall* algorithm from version 1.

- **Early reflections mode**
  You can choose the early reflection (ER) mode from a set of predefined modes, such as rooms, chambers, halls and plates. The modes with names that correspond to natural acoustical surroundings aims at realism. Please note that the room size should match the ER mode. Realistic room simulations should have small room sizes whereas halls require larger room sizes to sound natural. The *Wide* ER programs are designed to be as smooth as possible while at the same time enhancing the stereo image. These are ideal for music production.

- **Reverb time (s)**
  The reverb time specifies the duration of the reverberation and is specified by the number of seconds before the reverb tail level drops below -60 dB or 1/1000 of its initial amplitude. The freeze button on the left hand side of the numerical reverb time entry can be toggled to "freeze" the reverb. The reverb time is infinite and no input is added when the reverb is frozen.

- **Room size (%)**
  The room size defines the size of the simulated acoustical environment.

- **Pre-delay (ms)**
  The pre-delay slider allows you to adjust the time in milliseconds before the reverberation signal arrives.

- **Stereo spread (%)**
  You can use the stereo spread parameter to control the stereo width of the reverberation signal. If the stereo spread is set to 0%, the reverberation signal will be mono and at 100% the full stereo width is achieved. The *Vivid Hall* algorithm allows stereo spread values above 100%. *Vivid Hall* uses M/S processing when the stereo spread is above 100% and the reverb gradually decays more into the side channel with higher values. This ensures mono compatibility while adding a touch of extra width.

- **Dispersion (%)**

Frequency dispersion is an effect that occurs in mechanical plate reverbs. High frequency content travels faster in metal than low frequency content. You can read more about this effect in the [About Reverberation] 39 chapter. You can set the amount of dispersion in percent and 0% disables the effect completely.

- **VIVID HALL ONLY** **– Swirl (%)**
The swirl effect smooths the frequency spectrum as the reverb tail fades out, thus giving it a vintage "swirley" or "cloudy" effect. You can read more about this effect in the [About Reverberation] 39 chapter. The effect strength is specified in percent. Positive values of Swirl produce a more modulated reverb tail, whereas negative values calm down the fluctuations in the vivid hall algorithm. 0% disables the effect completely.

- **VIVID HALL ONLY** **– Bloom (%)**
The bloom parameter changes build-up time of the reverb and is specified in percent relative to the room size. The late reverb build-up gets quicker with *Bloom* values less than 100% and slower with values above.

-

- **LEGACY HALL ONLY** **– Modulation rate (Hz)**
Modulation can be applied to the late reverberation to simulate fluctuations in the arrival times of the sound waves and you can control how rapid the fluctuations are using the modulation rate.

- **LEGACY HALL ONLY** **– Modulation depth (%)**
The modulation depth controls the depth of the modulation ranging from no modulation (0%) to full modulation (100%).

- **Decay Editor**
The decay editor allows you to set up reverberation times that are dependent of the frequency. Air absorbs high frequencies to a larger degree than lower frequencies, so the low frequencies will in general have a longer reverberation time in natural spaces. This effect is easily modeled in the decay editor with a high cut or a high shelving filter. The *Vivid Hall* algorithm offers high and low cut, high and low shelving as well as bell filters to sculpt decay behavior with great flexibility. These filter types all have variable filter slopes. The *Legacy Hall* algorithm offers high and a low shelving filters with a fixed slope. You can toggle the filter bands using the buttons in the upper right corner:
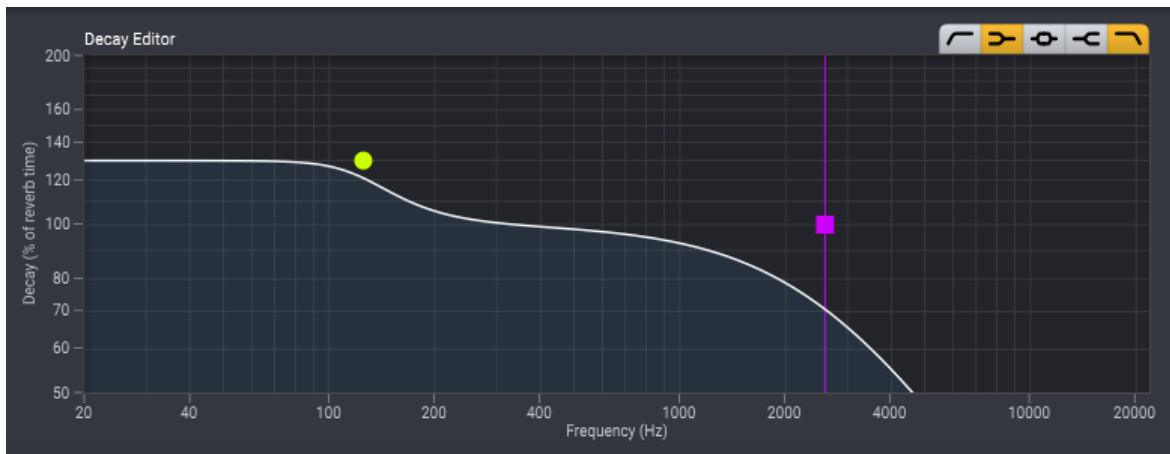
The decay editor lets you define frequency dependent reverberation times and the curve displays the relative reverberation time as a function of frequency.

- **Output Equalizer**

  You can use the output equalizer to apply filtering to the reverb signal. Both the early reflections and the dense reverberation are filtered using the output equalizer. The *Output Equalizer* is operated in the same way as the *Decay Editor* and you can use the button in the upper right corner to toggle filter bands:
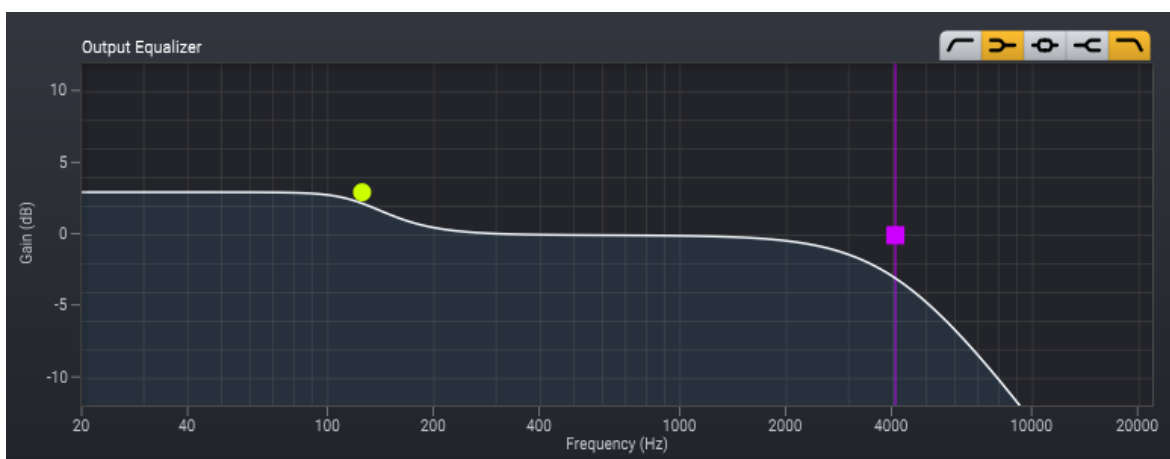
The output equalizer allows detailed control over the frequency content of the reverberation signal.

Preset Manager

Acon Digital Verberate 2 is shipped with an extensive set of factory presets. You can browse through preset categories and presets as well as create and manage your own presets using the preset management section:
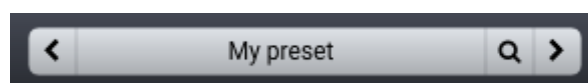
The preset management section available in Verberate 2.

You can browse through the presets using the arrow buttons. Alternatively, you can click the current preset name and a drop-down menu appears. Click the button with the magnifying glass icon to search for a preset. Matching presets appear in a drop-down list as you type.

You can also save your own presets by choosing "Save user preset file..." from the menu. A file chooser dialog box appears where you can enter the name of the preset you wish to save. You can create sub folders and place your preset files inside, and these will appear as categories in the user presets. Presets from *Verberate 1.x* can be imported into *Verberate 2* by clicking "Load user preset file..." and looking up the preset file. Please note that this won't be 100% accurate in all cases, so we recommend to keep *Verberate 1.x* installed for legacy projects.
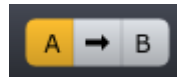
Undo and Redo

You can undo (or redo) changes to the parameter settings by clicking the circular arrow back or forward buttons:

Undo and redo buttons

A / B Comparisons

It is frequently useful to be able to quickly compare different parameter settings. You can do this using the A / B comparison buttons:

The A / B comparison buttons allows you to quickly compare different settings

You can keep two independent sets of parameter settings, the A and B settings, and switch between them using the corresponding buttons. The arrow button copies the settings from A to B or the other way around depending on which parameter set that is currently active.

The Processor Menu

The last button in the header section displays the processor menu:

You can click the processor menu button for the processor specific menu

The processor menu allows you to access the processor help topic directly along with other processor specific features.

3.6.1 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 walls or other objects. Due to the propagation time, these reflections arrive at the listener later than the sound from the direct path. After a certain build-up time, there are usually so many reflections that no distinct echoes are distinguishable, but rather a smoothly decaying sound.

The first few reflections, usually called early reflections, are important cues for our perception of an acoustical environment. For that reason, most digital reverberation units differentiate between early reflections and the dense late reverberation. *Verberate 2* simulates both the early reflections and the dense reverberation in a way that comes extremely close to what can be measured in a real acoustical environment. An important tool when analyzing the reverberation of real rooms is the impulse response, which can be measured by playing a very short impulsive sound (the impulse) and recording the resulting reverberation. The figures below are examples of impulse responses obtained from *Acon Digital Verberate 2* as well as from a real measurement of a concert hall.

Results of an impulse response measurement of the Tokoy Hall preset in Acon Digital Verberate 2 (the first 300 milliseconds).

A measured impulse response from a real hall (the first 300 milliseconds).

So-called convolution reverbs can use these impulse response measurements to recreate the acoustical space. However, they only capture a snapshot at a specific time. Real acoustical surroundings will have slight variations due to air currents caused by temperature differences or fans. Also, performers or people moving in the audience will cause slight variations. These small variations may seem subtle. However, the effect towards the end of the reverb tail will be significant, since the sound is reflected a large number of times before the reverb tail fades out. The new *Vivid Hall* algorithm models these random variations without artifacts like chorus effects or pitch changes, and is therefore capable of simulating reverberation of real halls with a higher degree of realism.

Dispersion Effect in Metal Plates

Mechanical plate reverbs are popular for their special sonic qualities. These vintage units produce reverb by placing a transducers on a metal plate. The powerful *Decay Editor* in *Verberate 2* makes it easy to simulate the longer high frequency decay that is special for plate reverbs. However, unlike how sound travels in air, high frequencies travel quicker than lower frequencies in a metal plate. *Verberate 2* lets you simulate this effect using the *Dispersion* parameter that you can set anywhere between 0 and 100% (for maximum time difference between high frequency and low frequency content).
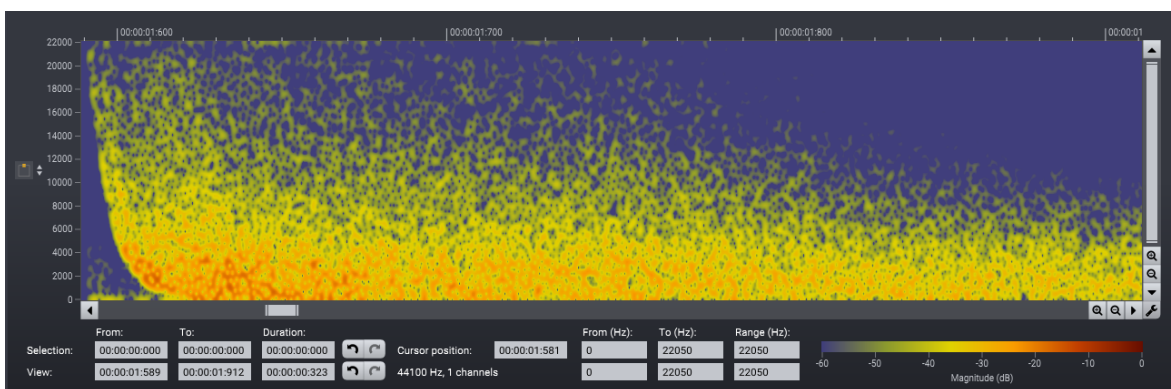
This spectrogram shows how high frequency content arrives earlier than low frequency content when dialing in Dispersion in Verberate 2.

Swirl Effect

Vintage digital reverb units commonly use frequency modulation techniques in the feedback loop to increase echo density without adding too much resonance. Such feedback modulation is no longer required in modern algorithms, but the feedback modulation can be sonically pleasing and desirable. The *Vivid Hall* algorithm makes it possible to simulate the effect of feedback modulation and we call this effect *Swirl*.
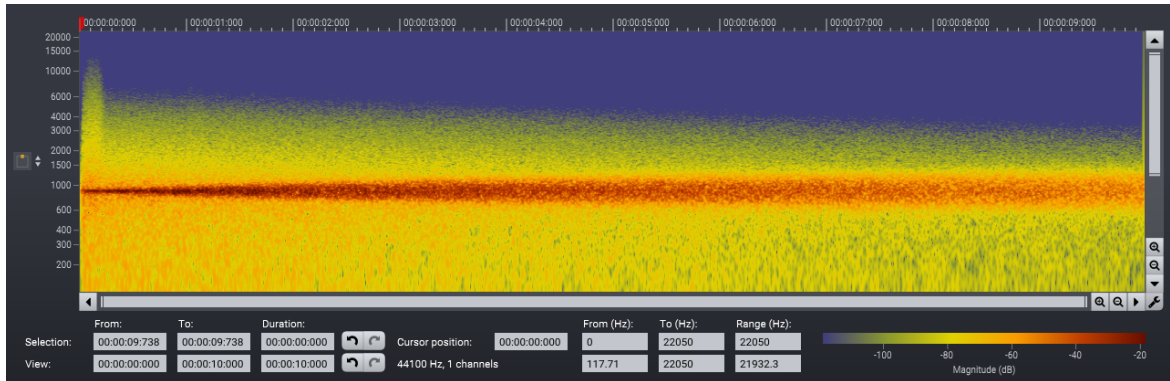
This spectrogram shows the effect of the Swirl on a sine wave input (880 Hz). Notice how the red line broadens towards the end of the reverb tail, giving the reverb a cloud-like vintage flair.

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