JAX TRIPLE [3] SERIES

Essential Effect Processing Re-thought.

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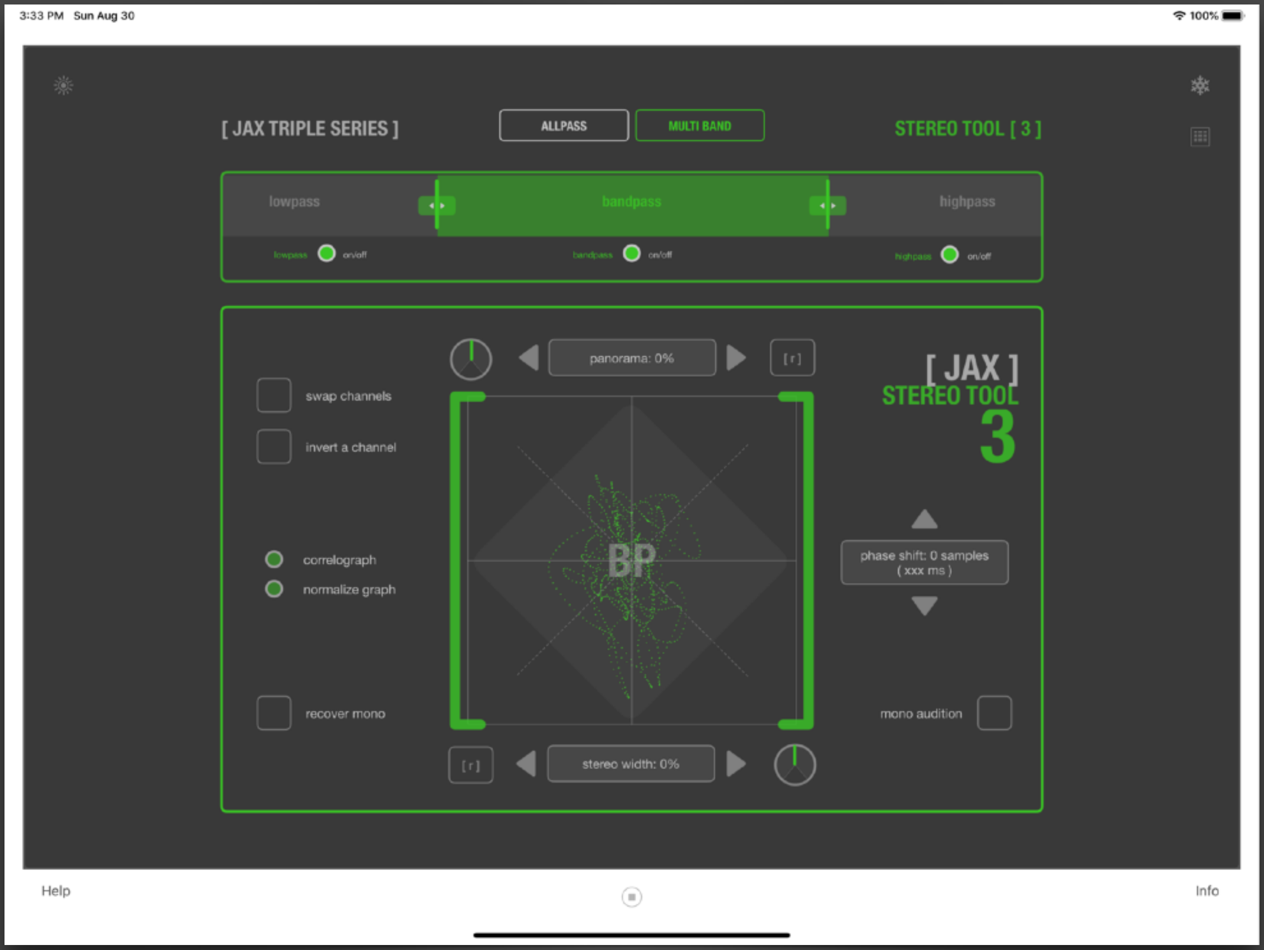
SPECIAL DEAL

IF YOU ARE AN ACTIVE SUPPORTING USER AND PURCHASED THE CLASSIC JAX STEREO TOOL 1/2 IN THE PAST, YOU WILL GET A FREE COPY OF THE NEW TRIPLE SERIES UNIT JAX 3Stereo! THE CLASSIC UNIT MUST HAVE BEEN PURCHASED PRIOR THE RELEASE OF THE TRIPLE SERIES JAX 3Stereo.

Please send us a copy or screenshot of the original Apple invoice via email at support@digitster.com and we will send you the free promotion code for lifetime access to the new unit. (This offer is not valid for our released free versions or versions that were free for a promotional period or were obtained with a free promotion code.)

JAX TRIPLE [3] SERIES

Essential effect processing re-thought.



The JAX TRIPLE Series is our collection of more essential audio units, but now based on an adjustable 3-band analog filter frequency split.

All the audio units in this series can be used like usually for common allpass (entire frequency spectrum) processing. But the important feature is the unique bandpass split function, which effectively will triple the effect processing engines and deliver easy to use parallel multi-band audio processing on demand.

Multi-band audio processing comes mainly from the professional audio mastering sections, where audio is commonly separated into several frequency bands to do selective modifications to these bands. It depends very much on the kind of audio material, how the frequency bands are adjusted to achieve the desired sonic results.

We have adopted this professional method and applied an analog modelled zero latency frequency split bandpass filter to all the effects in the series, so these effect processors can also be used in realtime without any additional latency for recording and live effect processing.

Each of the 3 stereo band effects will get its own set of automate-able and storable parameters. If the base effect is a reverb for instance, now 3 stereo reverberation instances will perform on 3 different frequency bands, each with its own unique set of parameters and finally, all is mixed together to the final stereo output again.

Note: All JAX TRIPLE SERIES effects are true stereo effect processors, so in fact not merely 3, but actually 6! effect instances will be performed in parallel connection.

So, the user can for instance apply a completely different reverberation to the high frequencies than to the mid and low frequencies. The result mostly will be significantly different, than passing the entire audio thru one single reverberation engine.

The reverb effect (JAX 3Verb) for instance, is merely an example for illustration. The new approach can be applied to everything, from frequency selective stereo wideners, phasers, saturation effects up to pitch shifters and what you can imagine. Everything consequently in triple band mastering quality.

We will successively release several essential audio effects with our JAX TRIPLE SERIES and replace some existing ones with the new method. The first unit is the recreation of our famous JAX Stereo Tool, widely used on many iPad Studios worldwide. ^^

JAX TRIPLE SERIES I JAX 3Stereo (formerly: JAX Stereo Tool)

Our classic JAX Stereo Tool was relatively successful over the last period, so we decided to recreate it for our first anniversary (2020) with the new triple split engine.

The basic functionality of the core has not been changed but now the user can now apply stereo corrections selectively to the frequency spectrum. This is a big difference and a great new possibility.

Even stereo problems are ideal for frequency selective modifications. Low frequencies usually need much less stereo widening than high frequencies for instance. Often these will profit from narrowing the stereo field. Mid ranges also can profit from selective stereo modifications. High ranges accept clearly more stereo widening to make the air kind of vibrating.

- Our unique phase shift feature can be applied to monophonic (pseudo stereo or narrow stereo) audio, to create interesting widening (stereoizing) effects or fixing phase problems in wrongly miked stereo recordings.

- The phase of selected bands can be inverted, which allows really interesting special stereo effects.

- Swapping channels of the middle band often drastically raises the psycho-acoustical loudness of the entire sound. This method is sometimes used in certain exciters (but these won't unleash that secret to the user).

- Selected bands can be set to mono or the lost mono signal in these frequency ranges can be recovered by inverting the correlation ...

- and so on.

The JAX Stereo Tool has many special functions more, that will come very handy for triple band operation. The new tool is predestinated for corrective high end mastering tasks, that otherwise must be corrected with overly complicated manual bus routings and allot of additional specialized effect processors.

The split filter for the frequency bands is an analog model, that adds subtile warmth to the entire sound, but actually does not add saturation or colourizing of its own. It is as transparent as even possible. Analog filter models do smoothly fade into each other. The separation is not isolating at the boundaries.

All parameters known from the classic JAX Stereo Tool are now available for each single selected frequency band. The continuous parameters are all smooth and can be automated without generating glitches into the signal. All parameters are completely saved with a preset.

In “Allpass” mode, the unit will operate the same way as in the classic mode (of JAX Stereo Tool 2) and no frequency split is done at all. The multiband feature will be bypassed in that case, which naturally also uses much less CPU.

In “Allpass” mode, the audio stream is processed with the AP parameter set, the correlograph will plot the phase differences for the entire audio stream.

In the new “Multband” mode, the first control is the band splitter, a handy widget for the frequency selective functionality. There are 2 handles, that can be adjusted to define the width of the frequency bands by just sliding to the left or right (or up and down).

Tapping one of the selective ranges will switch the user interface to the currently selected band and update its specific parameter set. This includes the correlograph. These parameter sets are still kept, if you switch back to allpass processing. So you can switch between these fundamentally different modi anytime and compare the sound.

You may switch between the different bands and also mute and activate them for listening to isolated bands or any combination of it.

Parameter Automation

We overworked the parameter automation and parameter smoothing completely with the new multi-band feature. It even allows to automate the split points and moving the middle band around in realtime. MIDI automation is also directly supported for the most important parameters.

All the continuous parameters are smoothed for preventing glitches and crackles in the signal flow as much as possible. But is lies in nature of things, that some parameters still will produce clicks (i.e. all boolean variables), if switched. This is unpreventable and the user must ensure, the effect is switched with such parameters merely at certain points of silence in the sound. (I.e. The mono switches and the inverting feature are of static nature and will produce audio artefacts if switched in realtime.)

Automation of stereo parameters and filter bands can give super creative results and is specially applicable when the sound requirements are not static, but changing drastically over time. For mastering we highly recommend to automate parameters to the things, that the audio material is needing in progression of time.

Or some parts of a song, that need some kind of excitement could be automated in the stereo field parameters to make them more interesting.

It is mandatory to read the original manual for JAX Stereo Tool 2, because there are all parameters explained in great detail. There are also loads of tips and useful hints.

(Note: A stereo processing unit at first sight seems to be a quite simple thing. But in fact, our JAX Stereo Tool is rather complex, the features are very powerful and there are many things you should know about stereo processing in general. A little tweak at the right places can improve the sonic result to perfection and now, even with the available frequency selective bands, everything completes for delivering a highly valued creative stereo correction and manipulation tool.)

Please always use high quality headphones for mastering tasks. We highly recommend the Beyerdynamic 770 / 880 / 990 Series of low impedance (les ohms) for mobile mastering. These are affordable and have a unique frequency range of 5 Hz to 35.000 Hz !!!

The following is copied from the comprehensive original JAX Stereo Tool 2 manual.

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JAX Stereo Tool (Classic, Version 2)

JAX Stereo Tool is a visual Audio Unit plugin for manipulating the stereo image of digital audio. The app comes with enabled microphone input by default. The built-in plug-in (Audio Unit) is available system-wide when you install the app. You can use it with all supporting host applications.

Audio Units are Apple's recommended method for providing shared audio effects for applications and audio editors. You can load the plugin into any host application that supports at least stereophonic audio effect plug-ins (in Audio Units format) with its own user interface. With version 2 the units are MIDI enabled, meaning some of their parameters can be automated with standard MIDI controllers.

JAX Stereo Tool has its main use as a specialized mastering effect that allows you to modify the stereo image of your audio by narrowing or widening it in a controlled manner. That is, the stereo output can be continuously adjusted in the basic function of pure mono to very wide stereo. This happens in real time and you can observe the result with the integrated correlograph, controlling and optimizing the audio thanks to this helpful additional visual information.

The phase of the source material is always kept stable with the JAX Stereo Tool and you have perfect control over the mono-compatibility of your material.

Mono-compatibility still plays a big role today, as extremely wide stereo sounds great on high-end stereo devices, but often it is astonishingly unsatisfactory on small playback devices such as mobile phones or small Bluetooth speakers, etc. This can lead to complete loss of frequency information and pressure of the audio signal, which will then considerably limit the sound experience.

The plugin is also great for correcting live recordings in real time where phase shifts may be a problem due to wrong microphone placement and distant locations.

JAX Stereo Tool therefore offers the possibility to correct the phase co-relations between the two audio channels up to a certain amount (100 ms). The conversion from mono to stereo is also phase-stable using a simple delay based adjustable (static, not modulated) sample offset.

With the JAX Stereo Tool, you can perform essential real-time correction actions that you will rarely find in other applications or even mixers. So you can for instance swap the two audio channels at the touch of a button or invert one of the channels, for example, correcting false cabling while recording afterwards or simultaneously (in real time) and combine even multiple actions to the stereo field at ones.

It is also possible to restore fully mono-compatibility from non-mono-compatible stereo in order to completely heal the signals that would be lost by phase cancellation with monophonic playback.

The Parameters:

At the bottom of the correlograph is the main "stereo width" slider control for adjusting exactly that, the stereo width. In the middle position, the audio signal is played unprocessed. The more the slider is moved to the left, the more the stereo signal is limited, that is, gradual converted into a monophonic signal. The resulting pure monophonic signal is what you would hear if the stereophonic audio were played back on a single or narrow speaker device without any modification. If you already notice a significant loss of audio energy by narrowing, then your source material is obviously not perfectly mono-compatible and has certain phase problems.

However, you can "recover" it. The "mono test" switch can give an immediate impression of the resulting mono-compatibility at any time.

On the other hand, moving the "stereo width" slider to the right broadens the stereo image, which means that the stereo components in the signal are boosted in relation to the mono content. Although most people perceive very wide stereo as “improving”, it tends to reduce mono-compatibility more and more. The energy of the mono signal is proportionally reduced with increasing stereo power. The stereo impression itself generally arises by shifting the phases of an audio signal by +/-180 degrees.

The Mono Recovering

With the switch "recover mono" you can make or "restore" a "lossy" stereo signal. A good example of this is an audio channel that has been completely out of phase, so that in the ideal mathematical case, there would be nothing (or very few) left to been heard, if it is played monophonically. In stereo playback, the same signal is clearly perceptible, with mono playback only silence or only very quiet noise can be heard or the whole thing sounds somehow hollow and/or extremely washed out. This paradox can hardly be perceived in rooms with large stereo boxes or under stereo headphones, because the brain usually “adds” these waves together. Nevertheless, a mobile phone with only one speaker (or two very narrow speakers) could not play such hi-fi audio satisfactory and the brain no longer can reconstruct the (now definitively) lost stereo information.

The parameter for the restoration of the mono-compatibility is therefore at first somewhat difficult to understand. However, the principle is very simple. First, the signal that would be canceled out by the shifted phases is extracted. Then it is rotated in phase and added back to the mono portion (left side of the slider). This way the signal is recovered and made audible again. So you can now fade (switch "recover mono" turned on) between the restored mono signal and the original stereo signal. Then select a good setting for the newly assigned "stereo width" and the audio signal is as far as possible restored with a partial of the lost signal and so corrected in its mono-compatibility.

Sometimes it makes sense to completely invert one of the channels or move it in phase. With the switch "invert" you can also invert one of the channels at the touch of a button. With the switch "swap" you swap the two channels, right and left completely. This may be necessary if, for example, a device was incorrectly wired during recording or if a connected signal processor twisted the channel or such things.

The shift of the phase with the "phase shift" is always constant and is realized by a fixed delay of one of the stereo channels. The maximum possible shift is around 100 ms. You should be very careful with that, because a constant phase shift on ‚phase correct' audio material can lead to significant frequency loss. However, it does wonders if the channels have been shifted exactly in the right relation to each other during recording. Such phase problems will happen very quickly due to problematic microphone placement or due to unpredictable room reflections or similar complex interfering factors. Here you have to manually try something around to find an ideal setting for the right recovering balance.

Sometimes it also happens that one of the two audio channels is constantly too quiet in the stereo image. You can easily correct this with the "panorama" parameter (upper slider control), which is switched before the stereo processor.

The Correlograph

The correlograph will visualize the phase differences of an audio signal on a 2D space.

A good mix is always characterized by an optimal balance between stereo vision and mono-compatibility. If the correlograph shows a predominantly upright oval graphic, this is usually good. If the oval tilts to the right or left, this circumstance indicates that the signal is not centered (it appears to be panned), that is, that one of the channels is too loud or too quiet and thus needs to be corrected in panorama. For example, if the oval has only a vertical bar, it is monophonic and has no stereo information at all. A monophonic signal can not be broadened with the stereo widener. However, there are different approaches. Sometimes a slight and constant change of the phase helps to create a pseudo stereo impression, even from monophonic input like a microphone. If you do so, you must carefully watch the resulting frequency range and response and possible comb effects and the energy of the audio material. There is no general rule to this. It very much depends on the frequency content of the audio material, which usually changes quickly.

If the oval of the correlograph collects mainly on the horizontal, then the audio is generally not or badly mono-compatible and there are strong phase cancellations in action. Such a mono signal would be very problematic, quiet and spongy to listen with monophonic or narrow devices. A horizontal line therefore means complete phase cancellation. Here you can invert one of the channels to restore mono-compatibility or simply try "recover mono" mode. Just experiment with the features.

With the JAX Stereo Tool you have all possibilities to analyze, control and correct the stereo image of your audio. If you are satisfied with the result of the settings, you can bounce the result into a new file. This is usually done by an external audio application, hosting the audio unit.

The JAX Stereo Tool is an essential tool in the mastering setup.

The following will describe planned releases for the JAX TRIPLE SERIES.

The JAX TRIPLE SERIES I JAX 3Verb

The 3Verb is our advanced 3 band algorithmic reverb implementation. So each of the bands can get its individual, independent, fully featured reverberation effect.

Our new flagship reverb can do many things that are impossible with usual reverberators. There are 2 independent delay units and an advanced shifter module implemented for creative applications.

A Mastering Reverb

Conceptually it was developed specifically as a mastering reverb, because of the implemented multiband feature. Mastering requires special reverbs, which allow advanced frequency separation and specially adjusting mixing controls.

Such reverbs are commonly used for closing frequency “holes” in a mix. This is always “adding reverb in a selective way”. The emphasis here lies on the word “adding”, not just applying.

Because of the fact, that a common reverb usually will spread over the entire frequency spectrum with uncontrolled energy and decays, common reverbs (even convolution reverbs) are not well (rather generally never any good) suited for usage as mastering reverbs because of the problematic, heavy frequency and transient masking occurring with them.

If you for instance apply a “canyon” reverb to an entire mix, even if only to a certain percentage, all will sound like played in that canyon and therefore wash out the transients and rhythmic definitions, making it a piece of damage, finally.

Users must understand, that a 3-band reverb is fundamentally different in usage and its sound possibilities than a common reverb. Although, JAX 3Verb has a “classic” mode too, which is called “allpass”. In this mode only one (the full) frequency band is used for the reverberation. This can be used for common and creative reverberation with single tracks and voices.

Mastering reverbs, on the other side, will do some things differently. Correctly applying a mastering reverb to any audio mix should result in smooth transition of the frequency selective parts into the entire existing sound and a mix can profit from certain improvements.

So the mastering reverb is much more respecting the input and more subtile than just applying a whole reverberation to the entire spectrum, which most likely must fail, the more the mix content is complex.

Our JAX 3Verb fulfils all requirements of a high class mastering reverb and can be used additionally for building of all sorts of creative “stand-alone” effects too. It is not limited in its possibilities compared to other reverbs, it is rather actually widely extended and conceptually improved.

The Multi-Band Engine

With our multiband feature, the energies of the 3 bands can be controlled perfectly and adjusted with different decay times and so create any combination of reverb tail development over time and the desired frequency spectra. This is ideal for closing holes in the mix and for adding special effects.

It is fundamentally different than to modify the reverberation result with EQs and filters afterwards. Instead of feeding the entire sound into the reverb and then modify the result to fit into the mix, you will select the frequency band for the reverberation first. And you will have the chance to edit each frequency band with different sets of parameters. Often even a single, well defined frequency band will deliver the desired result. Everything should be somewhat subtile.

The 3Verb is thought for both, using on a mixing send bus, routing its return signal or alternatively as a mastering reverb, directly processing onto the mix.

In these cases, the separated dry/wet controls are of special importance. And this is the reason that we did not combine these controls, as seen in other products. If you want to close holes in a final mix and use the “mix over” approach, the dry parameter must be set to 100 percent, of course, otherwise the sound would wash out. The bands and the wet parameter must be adjusted carefully, to fill exclusively the frequency gaps and the possible frequency deficits in the sound.

But the unit can also be used more commonly, to create real interesting sounding and extraordinary halls for usage on any kind of audio material. The unit supports common allpass processing for simple cases, without the band split mechanism.

The algorithmic reverberation kernel as it, is not overly complex and has got the most known parameters for keeping the usage as simple as possible. There are no such special controls as EQ or compressors, like seen in other products, because these would conflict with the frequency selective 3-band scheme. But there are some other special parameters, like the pre/post delay units and the shifter module for instance, which separate our reverb from others.

The Parameters

- Input: adjusts the input volume into the reverberator for each band individually. The bands dry level is not affected by this.

- Output: allows to adjust the output volume of the reverberator for each band. This also does not affect the amount of the dry signal.

- Dry, Wet: These parameters are separated for better control of the reverberation adjustment as needed in different situations. A Wet value of zero effectively means no reverb at all, but internally it will be rendered (i.e. for automation continuity). The dry parameter has very special function for mastering tasks and is completely separated from the reverberation processing. This prevents complicated bus routings.

- Size: will adjust the reverb sizes (decays of the reverberation tail, corresponding to a virtual “room” size). This parameter scales differently with each chosen model.

- Model: the model will allow to select different depth reverberation models for each single band. There is a “early”, a “medium”, a “late” and an “ultra late” model available. Sometimes this is called “gravitation” or similar.

This parameter very much covers the responsiveness of the reverb tail. The combination of different models on different bands can effectively simulate all combinations of so-called early reflections /

long developing tail relations. This parameter seems to have a relation to the delay parameter, but is a separate adjustment.

Note: Users often love ultra long developing reverbs as a kind of “bench mark” for the quality of reverbs. ^^ Now, with JAX 3Verb you can create (ridiculously) irrealistic long reverbs that cannot exist in nature. Even in combination with the freeze or the delay feedback feature, this will allow creative application of such super reverbs, for what usage ever. ^^

- Damp: will effectively reduce the energy of the reverb tails over time. This corresponds closely to the prominent “material” (absorption) theorem of rooms and spaces. The parameter also increases or decreases the lower energy parts of the wet signal (emphasizing the lower frequency impact). An amount of zero produces extreme frequency boost in the upper ranges, which can make some trouble. Please be carefully with the adjustment and do not overdo removing natural impulse damping.

- Width: The stereo spread of the reverb tail can be adjusted continuously with this parameter. Zero adjustment will deliver a mono sound of the reverberation, which can be useful in certain situations. The parameter is adjusted in a way, that allows over-boosting the stereo phase of the wet signal. The dry signal is not affected by this.

- Freeze: Normally the reverb energy will decrease naturally over time. With this parameter the (unnatural, irrealistic) freezing of the reverberation can be forced, creating an everlasting reverb tone or the impression of infinity of a space. This parameter also can be automated to the point of needs, which is very effectual. The parameter is even more interesting, when applied only to selected bands.

- Pan: Reverb panning can be applied by this parameter, which in some situations can help to correct the stereo image and to create impression of movement. The pan parameter is applied exclusively to the wet part of the signal.

The Integrated Delay Units

JAX 3Verb has got 2 different delay units, that can be used simultaneously. These are specially integrated into the reverberation engine. The first is a pre-delay, that will be applied to the input before the reverberator. But the dry signal is not affected by this.

The second delay is applied after the reverberator and is also strictly separated from the dry signal. Both delays have the same parameter set but their sonic effect is clearly different. Delays can be switched completely off.

- Delay Tempo: The delay (simulating early reflections) is rarely used on mastering reverbs, because such reflections usually will introduce rhythmic problems if applied to a mix. Therefore we used a tempo delay here. The parameter adjusts the time offset for the delay, which is auto-calculated by a given tempo value. The possible tempo ranges between 40 and 480 bpm. If the song tempo is known, the delay will automatically fit to the rhythm by adjusting it to the song's tempo.

- Delay Division: This is the division of the tempo delay in fractions of a quarter beat. A division of 1/1 means one delay per beat.

- Delay Feedback: For simulating early reflections, usually no feedback is used. But we have implemented a feedback, for special purposes. Modern music styles will profit from this implementation, as the reverberation can exactly fit (move) to the rhythm and the delay itself is adjusted with a tempo value rather than a time parameter.

The delay will be applied only onto the reverberation result, not to the audio input.

The delay actually extends the 3Verb to a kind of hybrid effect for interesting tempo dependent reverberation. Many reverbs (especially convolution reverbs) have problems with fitting to a song tempo, because the early reflections are an unchangeable part of the reverb and usually cannot be adjusted to any specific tempo without introducing further problems (i.e. by stretching).

Algorithmic reverbs however are not limited by this.

Note: For a specialized multiband delay effects, please take a look at JAX TRIPPLE SERIES 3Lay. It has got the delay units without the specific integration into a reverb engine.

Real Stereo Processing

JAX 3Verb is a real stereo triple band reverb, which means, the input of each band is not mixed to a mono signal prior feeding the reverberation. Many commercially available effects obviously will mix stereo to mono prior feeding the reverb, without even asking you.

Each signal way of the triple band processing in JAX 3Verb is consequently stereo, also the delay for instance. The additional panning feature will even allow realtime panning of the reverb tails.

The triple reverb engine has got an auto-leveling limiter at the output, preventing unwanted loudness bursts and distortions. If the reverberation starts pumping or losing energy with some frequency content, please check and re-adjust the single bands in its volumes or decrease the global input/output levels. The parameters of the integrated limiter are also available as (hidden) audio unit parameters, not specifically exposed to the user interface.

The user interface of our 3Verb will display the selective frequency spectrum of the reverberation for visual information of what"s happen at each selected frequency band. However, it is not thought for any scientific purpose. Only the reverberated signals are analyzed, isolated and maximized in their magnitudes in a special way.

The spectrum is scaled to a “musical” scale, which means that it corresponds to the frequencies of the MIDI key mapping on a virtual keyboard, where the middle screen position (key 64) has a quite low frequency of merely 311 Hz. 440 Hz (concert tone) is mapped to key 70 for instance on the chromatic scale. The highest MIDI Tone (key 128) has 12.500 kHz, the lowest (key 1) has 8 Hz.

A musical frequency scale is the most logical and consequent frequency analysis scale in audio production, we think.

A Word to the World of “Shimmers”

The so-called shimmer reverb is very popular but also quite misused. And so are many available effects, aiming to deliver something like that.

Generally, a real shimmer reverb does NOT pitch shift the reverberation nor does it pitch shift anything, but will rather emphasize the higher harmonics and frequencies. This creates the natural “shimmer” effect.

So it has not really something to do with “pitch shifting”.

However, we have implemented experimental pitch and frequency shifters for creative purposes. But we want to point out, that such experimental stuff is not part of the reverberation core.

With the frequency selective reverberation of the JAX 3Verb, you can create natural sounding real “shimmering” reverbs by just adjusting the frequency bands in the desired manner. Any reverb, that features a simple high pass filter, will effectively be able to create “shimmer effects” that sound natural and pleasing and also convincing.

Because it lies in the nature of things, that emphasizing the higher frequencies effectively will boost the higher order harmonics automatically, but in a natural way. So there is no “pitch or frequency shifting” required to create shimmering reverbs. Pitch shifted reverbs quickly will sound like fake. The human brain is extremely sensitive, identifying fakes.

A rare situation where we can imagine the usefulness of a pitch shifter in connection with a reverb is, when temporary positional moving emulation is required, i.e the sound of a fast moving noisy object (a fast driving alarm horn for instance), that creates the short impression of a pitch shifted sound. But this is rather a frequency shift, than really a pitch shift.

The (experimental) Shifter Module

The experimental shifter module can modify the reverb tail to some extent for creative usage. Adding pitch- or frequency shifted duplicates of the reverberation can drastically create new frequency content and this is sometimes more massive than it could be created with any other method, like harmonic exciters or such.

Adding new frequency components has a certain importance for a mastering reverb, because usually a reverb will only alter frequencies, that are available in the audio material. By shifting frequencies in certain ranges, new fundamental frequencies can be created with all their sub harmonics. Therefore the shifters can be tuned in semitones and possibly automated too. Gaps of harmonic content will be closed successfully by using frequency shifting in controlled manner.

There are several shifting algorithms for selection. Some of them need extraordinary high CPU power and are intended to be used selectively and sparingly.

- tdp, a time domain pitch shifter. This is a delay based pitch shifter that tends to become grainy with extreme amounts, but has very good performance.

- fdp, frequency domain pitch shifter. This will use FFT (fast fourier transform) for shifting the pitch. It has clearly higher quality because it is much smoother but needs very much CPU power. It may not execute on all bands without overload with older devices.

- bode, a classic analog modelled bode frequency shifter. Frequency shifting is fundamentally different than pitch shifting and will often result in inharmonic, metallic sound. This kind of frequency shifter also needs some processing power.

Please remember, that there are a maximum of 3 instances of real stereo processing modules (6 instances) performing at ones in parallel connection. The shifters are not an integral part of the reverb an should not be used for everything.

The shift for each band can be adjusted to either

- off
- +/= one semitone (+=100 cents) or
- +/=12 semitones or
- free.

The “cent” mode is good for minimal detuning, the “semitone” mode is good for melodic adjustment and “free” is good for special effects. In “bode” mode, the semitone adjustment is equal to “free”, because it is not chromatically adjustable.

These parameters can be automated. The chromatic semitones will automatically snap to the values for easy melody creation.

The percentage of mixing the reverberation with the shifted part can be adjusted with the “amount” parameter. A value of 0 will bypass the effect, 1.0 will output 100% of the shifted reverberation. Good results will be achieved with moderate amounts. A 100 percent adjustment will sound most likely unnatural. The right mix does all the magic.

All the shifting generally will only be applied to the reverberation parts, not the dry parts of the signal. The shifter module is placed directly after the reverberator with mode 2 (post connection) or before the reverberator in mode 1 (pre connection). The result is quite differently sounding in these both connection modi.

The JAX TRIPLE SERIES I JAX 3Xite

3Xite is a multiband exciter for adding new harmonics to the sound.

Harmonic excitement can replace common equalizing effectively. Equalizing will only boost or damp what"s available in the frequency spectrum. A harmonic exciter can create new frequency content, previously not present in the audio material.

The JAX TRIPLE SERIES I JAX 3Boost

The 3Boost audio unit is a loudness boosting tool and thought for superseding our famous classic !Make Louder effect.

There are 3 independent limiters, which will operate on the selected bands. The classic mode is performed in allpass mode.

The classic effect tended very much to loudness pumping on a single band, if the levels are high. This can be minimized with fine adjusted bands. Each band can have its own set of parameters for the reaction time adjustment.

These limiters are very transparent. The tube models of the JAX !Make Louder are not implemented. We recommend JAX 3Fat for multiband saturation or the JAX 3Xiter for analog style sound colourizing.

The JAX TRIPLE SERIES I JAX 3Crush

The 3Crush effect is thought as a replacement of our JAX Decimator audio unit.

Frequency selective bit and rate reducers are much more effective and versatile and the sonic results can be fine-tuned for every taste.

We also added some optional filters for reducing the side effects of the raw decimation.

The JAX TRIPLE SERIES I JAX 3Fat

3Fat adds pleasing saturation / desaturation effects to the multiband processor. The effect will modify the waveform in the manner of a waveshaper and it is bi-directional.

Values above zero will fatten (saturate) the sound, values below zero will thining the sound (desaturate).

This effect creates new harmonics to the sound.

The JAX TRIPLE SERIES I JAX 3Lay

This is a 3-band delay effect, where each band can get its own set of parameters.

The delays are stereophonic and can be synced to the host tempo. Frequency selective delays can create complex rhythmic results.

The JAX TRIPLE SERIES I JAX 3Phase

This is a 8 stage phaser, which will apply a bank of allpass filters to each of the 3 bands.

The popular effect is taken from our JAF Filter Collection, with the addition of an LFO. The LFO can operate in the audio rate frequency range, but also very slow movements are possible, which is for instance good for subtile stereo enhancement.

The JAX TRIPLE SERIES I JAX 3Shift

The 3Shift effect will apply frequency band selective pitch and frequency shifting. There are 3 different modi integrated, which all have fundamentally different sonic effects.

One of them is a Bode Frequency shifter, the other 2 are different pitch shifters, one operating in the time domain, another operating in the frequency domain.

Please note that all pitch shifters add latency to the sound.