JAX SPECTRO SERIES

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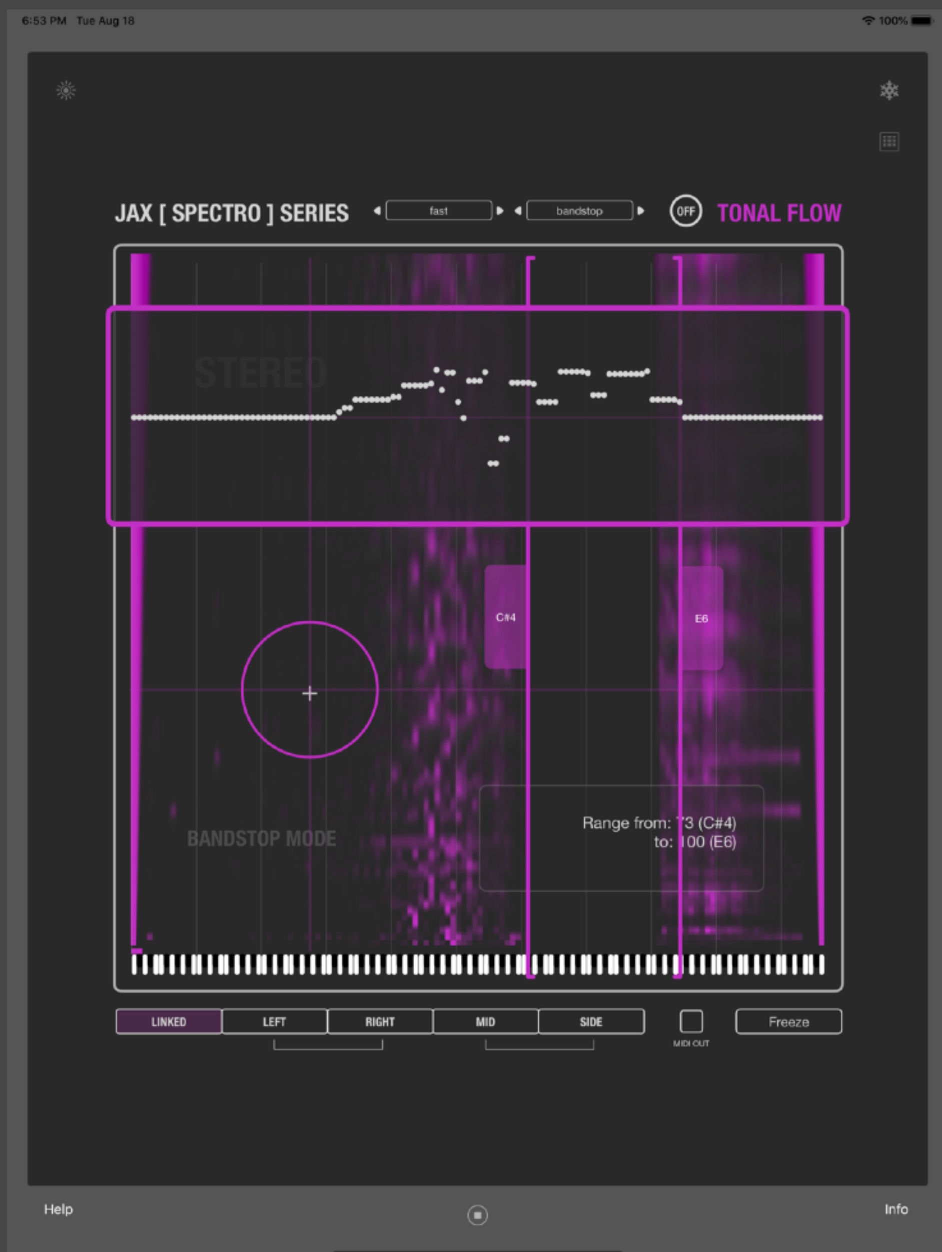
The JAX Spectro Series is a collection of audio units, which are based on a modified FFT (Fast Fourier Transform) algorithm. It provides several filters and tools for analyzing and altering audio in the (musically relevant) frequency domain for realtime usage. The tools are intended for usage as mastering grade audio processors. Pleas note, that FFT will conceptually introduce a latency.



The FFT Paradigm

FFT always introduces latencies of certain size if used in realtime processes. This lies in nature of things, as there always must be collected a certain amount of samples before the FFT can perform.

A critical range are the lower frequencies. The deeper the bass analysis / modification is required, the larger the latency usually must be. For analysis and mastering this may negligible but for real time recording and playback tasks it will become problematic because latencies usually cannot be compensated with this kind of immediate operation.

On the other side, digital filters based on FFT can deliver very accurate results, that are not achievable with any other effects, like analog modelled filters and tools (like our JAX Chromatic Series for instance) or like traditional time domain pitch shifters, which are mostly delay-buffer based, and many things more.

The main focus of our FFT tools is the detailed analysis of audio material, especially the tonal content and strict frequency separation while filtering. We have tried to provide tools, that are not overly scientific, but more musically senseful and intuitively to use.

Many available tools focus very much on the mathematical theorems rather than any musical usefulness. So our frequency scaling is mainly done in the most musical way, the chromatic semitone scale, where each analysis bin corresponds exactly to a MIDI key range.

Some important words to our UIs (user interfaces)

We are often asked why we are continuously using our own, admittedly somewhat special, user interface paradigms.

The answer is, that we ourselves find it most easy to use, most productive and most suitable for touch screen usage. And therefore we will not change it.

Unlike the majority of other products, obviously coming (and just copying blindly from the desktop computers), we will not adopt such terrible things like classic “popups”, endless nested pages and/or similar mouse specific behaviours, that initially were developed for desktop computers mouse pointer tracking and key strokes.

Our products are developed for usage as control interfaces on mobile devices by touching them with fingers. °°

Popups are for instance by nature modal and screen locking. This actually will break any workflow and is, by the way, the most awful behaviour on mobile platforms, we can even imagine. Even Apple uses this behaviour often (where it would be avoidable) and we find it just terrible. Also invoking endlessly new pages by tapping and sliding like crazy completes this entire mobile schizophrenia. This is by no means our style of working. ^^

We also will not implement any other such commonly used mouse pointer related behaviours, because on mobile devices, there is usually (God beware!) no mouse pointer and usually also no keyboard.

- Our UI controls are all inherited from well thought , easy to adjust and specific parametric sliding controls, which allow to touch and move the finger in any direction for adjusting or selecting values. These values drive always relative increases/decreases, not following the exact touch pointer location, for fast and smooth and scaled adjusting behaviour. The speed (scaling) of the adjustments are dependent of the currently selected screen size. And the screen size is freely adjustable. This offers a great flexibility.

- We will generally not use multi-touch, because of the fact, that it potentially will confuse the logic of an audio plugin. The danger of wrong adjustments by accident is much too high and the multi touch is reserved for zooming and moving with our units. Humans are by nature not multi-tasking b.t.w. However, there may be rare situations where it really makes sense to use multi touch. And we actually will be aware of such exceptions.

- Our UIs are basically self adjusting in their size, because of the simple fact, that all host applications provide different screen sizes and we want to support zooming in and out freely, for optimal workflow and parameter readability and access.

Audio Units are plugins, that must work everywhere. Advanced hosts will allow you to freely adjust the screen size of an Audio Unit. Our units are made to fit to anything and anywhere. We find fixed size screens, just copied from the widows platform plugin ^^ ... eeermm .... ~~sh~~.. kind of somehow “lazy”.

NOTE: **The scaling and moving of our UIs can be always prevented with the “freeze” button at the right top corner aka 'snowflake'.**

- The contrast of our user interfaces is freely adjustable, we even provided this feature with the very first releases, where the “dark mode” in iOS was not yet available nor even thought. Latest releases will allow to save the contrast parameter with the presets to provide persistence.

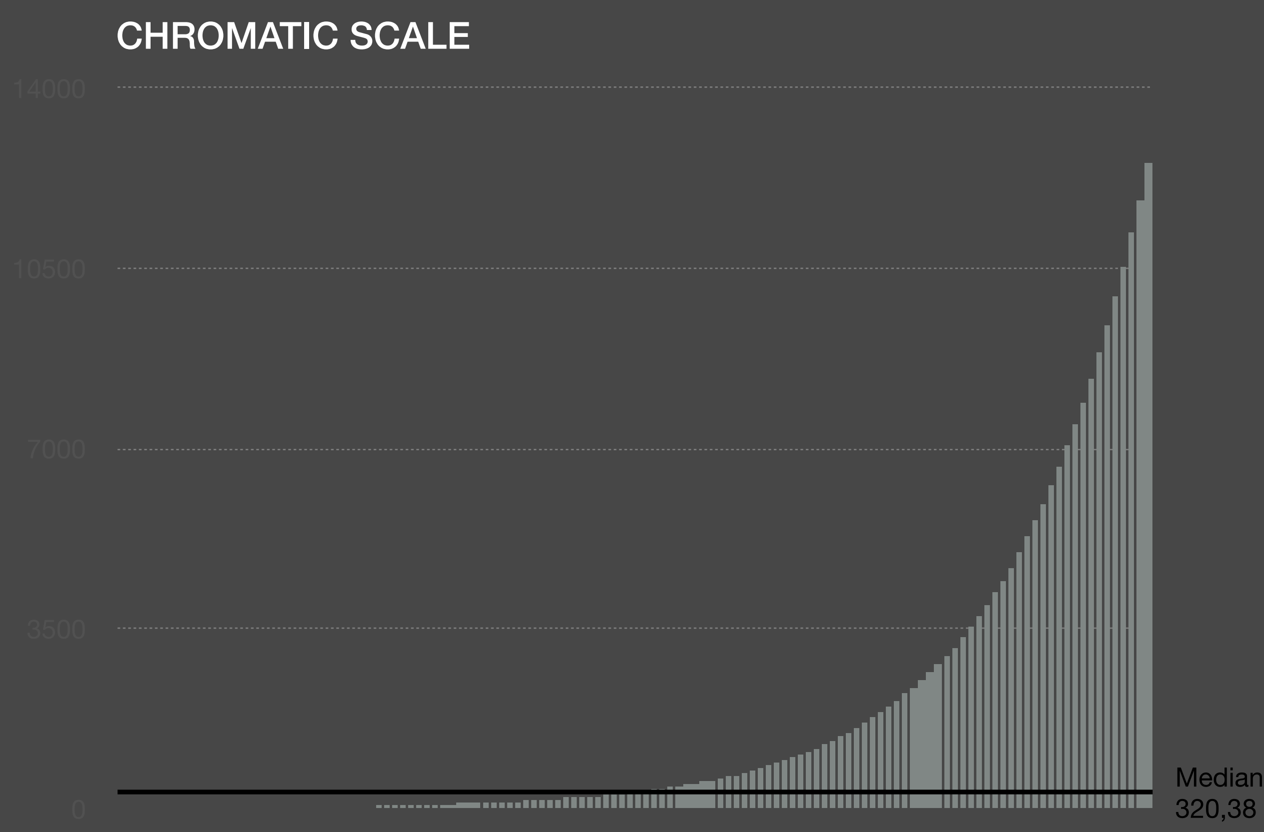
The control button overlays on top of the screen are persistent with all our user interfaces and can be invoked to lock the screen or adjust the contrast of the user interface on demand at any time.

The JAX SPECTRO | Tonal Flow
(Release Name: JAX ISOLATE)

This new audio unit, our first release in the Spectro Series, is a highly specialized triple action realtime FFT processor, that delivers

- a tonal analyzer with flow spectrogram,
- a tonal range bandpass/bandstop filter and
- a 128 band tonal adjuster (eq/filter)

in a useful combination. The frequency bands are consequently mapped to the musical range, in fact the MIDI key frequencies (chromatic mapping, equally tempered). And this will separate our tool quite much from any other FFT processors available.



3 different operation modi can be selected:

1.) The Analyzer

In analyzer mode, the effect will display the chromatic spectrogram for visual information. With this special kind of flow spectrogram, melodies and tones can be easily identified. The speed of the flow is adjustable.

Note: If you did not work with tonal spectrographs yet, you must adopt the fact, that a usual spectrogram is scaled from linear to anyhow logarithmically. We have been gone far, by focussing concretely on tonal content. So each dot in the spectrograph corresponds to a MIDI key, a chromatic mapped frequency, which can be seen in the frequency plot above (please see also frequency table at the end of this document).

As you can see in this plot, the chromatic scale is extremely different to a linear or usually taken other logarithmic scales (including Bark scales) and it merely covers a fraction of an FFT analysis result. The most of the FFT data is not even audible for the human ear, but usually presented with many common analyzers, data that rather can be considered as being irrelevant.

The median of the chromatic scale (MIDI key 64) even has only 320,38 Hz, which would not even be easily visible on a linear frequency scale, despite of not being able to extract any tonal/musical information from this at all.

So a tonal spectrogram will supress all irrelevant data and merely unleash the relevant musical data. A tonal spectrogram is not thought for scientific frequency analysis.

Note: Our JAX CHROMATIC SERIES are very much related to this. But the chromatic series will use analog modeled bandpass filter banks for chromatic analysis, which will result in better bass frequency detection.

MIDI Output (Audio to MIDI)

There is (in connection with the range control) a MIDI output option, which attempts to generate MIDI notes from the selected range analysis. This is working with polyphonic results(*).

(*) Please note, that a perfect extraction of melodies from audio material is just not possible with FFT, due to the nature of things. You have to select the effective tonal range for the operation carefully, otherwise you most likely will get quite rubbish via the MIDI output of the unit. The frequencies in the deeper bass range may be insufficient for a successful melody extraction, due to the reduced resolution of a realtime FFT.

The MIDI output function is available only within the analyzer mode, not the filter modi. However, you may adjust the effective range with the filter modi (bandpass filter) prior, for listening and adjusting to the tone range.

In analyzer mode the filters are bypassed and only the input will be analyzed. This is using far less performance than one of the 3 filter modi. With the filter modi, only the output will be analyzed and displayed in the flow accordingly.

The range control in the analyzer mode will affect the MIDI note detection algorithm and output. It will not apply filtering to the audio. So the analyzer mode does not introduce the latency to the audio output, which is passed thru completely untouched (resynthesis/inverse FFT bypassed).

2. The Bandpass/Bandstop Filters

2., 3.) The second and third mode uses the tonal range bandpass filter, whereby a range of tones can be selected and the rest is filtered out of the signal. This filter has an extremely steep slope, which means, that frequency content can be isolated (i.e. isolating the singer, a melody or a bass line completely inside a mix and so on).

The effect is invert-able, so that effectively a bandpass can be cut out, up to single tone frequencies, which can be isolated with this.

The inverted bandpass filter is called bandstop and an own mode, selectable with the mode switch (which is basically a slider control, it will be adjusted by pressing and moving left/right or up/down).

Note: For untrained listeners, the range mode may sound strange at first. Strict isolation of frequencies is sounding anyhow “artificial”. and quite different to any analog modelled filters and EQs.

Users must get a good feeling of how strict frequency isolating is sounding. This is a requirement for successfully adjusting values with the adjuster mode for getting the desired results. Usual EQ-ing and filtering is different in the required adjustments.

The range control can be automated but in manual mode, the filter is switched merely if the handle of the adjustor was released. The 2 range modi (bandpass/bandstop) may raise sound bursts with abrupt changes. The correct limiter adjustment is the key for smooth sound transitions, even with parameter automation. The limiter parameters are exposed as hidden parameters (not directly available on the user interface).

4. The Adjuster

With the “adjuster” bandpass control, the tonal frequencies can be drawn freely to provide 128 bands of equalization/filtering with the audio material in realtime. The tones are adjusted by just drawing any shape with the finger. On top of the screen, an overlay control will appear, where each tonal band has got its own slider.

You may move fast with the finger across the screen, for closing gaps and move slow, for adjusting fine values per band.

Frequencies outside the MIDI frequency spectrum will be passed thru and not cut, except in bandpass mode. MIDI usually covers a limited frequency range from approximately 8 to 12000 Hz across the entire key range of 128 possible MIDI keys.

Most of the frequency content that FFT is usually analysing lies outside the audible range and even more outside any useful tonal range. If you want to adjust ultra high and low frequency content, so JAX Tonal Flow is not the correct tool.

The maximum boost (sliders full open) is multiplying the frequency of each selected band by a maximum factor of 4.0 (400 percent). The middle position always corresponds to 100 percent. If one of the drawn band sliders is adjusted to zero (bottom), the tone is not just damped, like on usual EQs, but filtered out completely with an ultra steep slope (output is 0 percent of the selected tone range).

So users can create kind of “musical” “frequency comb” filters (do not mismatch with delay based comb filters). There may be several presets, that allow for instance to select adjustments for common predefined musical scales and other special effects and so on.

The main usage of the adjuster is the extreme sound modification possibility by creating completely different sounding frequency content from any audio material, which is just impossible with usual filters and EQs. The strong isolation and the extreme boosting can modify a sound drastically, keeping it musically intact. Everything can be automated too.

Self Adjusting Volume

Please note, that boosting (drawing the sliders above the middle line) will effectively raise the volumes of the selected filter bands by the factor of max. 4.0 (below the line will always reduce against zero). This makes a complex intelligently chained dynamics processor at the output necessary, because otherwise, especially the tonal boosts would result in nothing less but digital distortion on the entire precious sound.

Our Spectro Series audio units therefore include a self adjusting limiter device, which even tries to correct really extreme selective frequency values without the expected sound distortion. The tonal character of the modified audio material may change drastically but should not distort (as for instance on many other EQs and filters, even so-called “studio EQs”...) with our auto-gain-adjusting approach. The key for this is the integrated auto-adjusting limiter device.

This actually makes our Spectro Series processors so extremely powerful and quite different compared to the competitors mastering products. A generally distorting or “saturizing” EQ or filter will effectively and fundamentally damage the entire piece of music in an unrecoverable manner. Also sometimes this is called “colouring” EQs, which are rather problematic than any useful and very much a matter of (bad) taste.

Our JAX Spectro Tonal Flow filter can be used as an tonal, extremely modifying, maximal transparent, chromatically tuned 128 band EQ, that usually creates zero saturation or distortion when boosting frequencies. A medium speed limiter ensures that the volumes do not increase far above the zero db level for long time. However, extreme adjustments of values can cause distortion over a short time. A brickwall limiter is recommended extra.

An example of the auto adjustment: Boosting the bass frequencies extremely will result in an auto-balanced frequency spectrum, where the bass is clearly emphasized, as expected, without getting the frequencies distorted. The surrounding frequencies then will appear more or less clearly damped as a result of the internal auto adjustment. This is the only comfortable method for auto-corrective adjusting, without the need of manually re-levelling everything or applying saturation/distortion to the peaks.

Please do not mismatch this with a spectral compander (which is available as an powerful extra unit). A spectral compander expands or compresses the frequency band relations in time dependency, driven by signal followers. With our JAX Chromatic (analog) and JAX Spectro (digital) spectral companders, you are able to perform consequent musical spectral compression, which is fundamentally different than just equalizing or even exciting selected frequency ranges.

The Processing Modi

All 3 main operation modi in JAX Spectro Tonal Flow will provide sub modi for selecting the current stereo division. Consequent stereo division is another powerful feature that not many mastering processors will offer.

- “Linked Stereo” means linked channels processing. This is probably the most commonly used processing mode. Both stereo channels will be mixed to mono for the analysis part and each of the stereo channels will be processed separately (real stereo processing). But all level adjustments are applied equally to both stereo channels, regardless of their (possibly different) loudness.

- “Left” and “Right” will display and process both stereo channels independently. Differing adjustments will be made by switching between these two modi for each of the stereo channels. This mode can be used, if independent channel adjustment is a requirement. It can also be used to create extremely panning frequency divisions, which effectively alters the final stereo image of the sound and its psycho acoustic expression.

- “Mid” and “Side” will separate the stereo phase (correlation signal) from the mono part of the stereo signal by subtracting. Both of these split channels now can be selected and adjusted separately with the full set of filter band parameters. The signal will be mixed together again, after the processing to deliver the new stereo result. This mode is mainly for optimizing and editing the stereo phase and other stereo-dependent adjustments. Separate editing of the mid/side signal can drastically modify the width impression of a sound and correct special stereo and also (sometimes) dynamic problems with following dynamics processors, caused by strong differing stereo phases.

Please note, that you will get 2 completely independent sets of parameters (each 128 bands) with the Left/Right and the Mid/Side processing. You usually switch between these different parameter sets by pressing the the mode switch buttons on the bottom of the screen, for viewing and adjusting the selected parameter sets.

Note for parameter automation: Levels that are made for one of the selected stereo modi, will not be applicable to another fundamentally different mode, if switching between these modi. This means, if for instance a mid/side processing was automated and the user later switches to left/right stereo processing, the existing automation will not affect now the newly selected stereo mode, as this won't make any sense. All parameters for the different modi are strictly separated and stored with a preset.

About FFT Latency

The FFT filter approach usually introduces latencies of around 1024 to 8192 samples but it is the only available method that actually can be used to filter tonal ranges with nearly "perfect" isolation. If precision in the bass range is a requirement, the latency must be clearly larger, otherwise it can be chosen smaller.

The reason is, that FFT operates linearly inside the frequency domain and the human perception is scaled quite differently. Sometimes this is called the "Bark" scale, but we do not use this. Different is also the musical scale, which is scaled tonal logarithmic, everything around the concert pitch of exactly 440 Hz. With the musical scaling to exact "tones" we tried to combine human perception with useful musical theory, to deliver a profitable system for musicians, based on the otherwise purely linear fourier transform theorem and all its descendants.

A Mastering Tool

JAX Spectro Tonal Flow is thought mainly for mastering (post) processing and for creative sound manipulation in sequenced environments and audio editors. It can achieve things that are just not possible wit usual filtering and equalizing (i.e. strict isolation). But because of the higher latencies, it is not very well suited for usage with real time recording or real time effect processing.

Parameters and Presets

JAX Spectro Tonal Flow has (hidden) input and output volume controls along with all exposed audio unit parameters from the user interface. The audio unit b.t.w. comes with over 640 (128 x 5) grouped and exposed band adjustment parameters. Each single band parameter can be automated this way separately. The audio unit host application must support parameter groups with large sets of parameters for doing this comfortably.

JAX Spectro Tonal Flow presets can be saved by the user and also shared. The file format uses human readable *.plist files, which are comfortably editable on the MacOS or with any text editor (XML formatted, UTF-8). But we do not recommend to edit these files manually, because some required knowledge of the format is required.

The user saved presets are identical with the standard preset, that are saved by host applications.

The following pages include other planned products in the JAX SPECTRO SERIES

MIDI FREQUENCY TABLE

MIDI KEY	FREQUENCY (Hz)
128	12543,85
127	11839,82
126	11175,3
125	10548,08
124	9956,06
123	9397,27
122	8869,84
121	8372,02
120	7902,13
119	7458,62
118	7040
117	6644,88
116	6271,93
115	5919,91
114	5587,65

113	5274,04
112	4978,03
111	4698,64
110	4434,92
109	4186,01
108	3951,07
107	3729,31
106	3520
105	3322,44
104	3135,96
103	2959,96
102	2793,83
101	2637,02
100	2489,02
99	2349,32
98	2217,46
97	2093

96	1975,53
95	1864,66
94	1760
93	1661,22
92	1567,98
91	1479,98
90	1396,91
89	1318,51
88	1244,51
87	1174,66
86	1108,73
85	1046,5
84	987,77
83	932,33
82	880
81	830,61
80	783,99

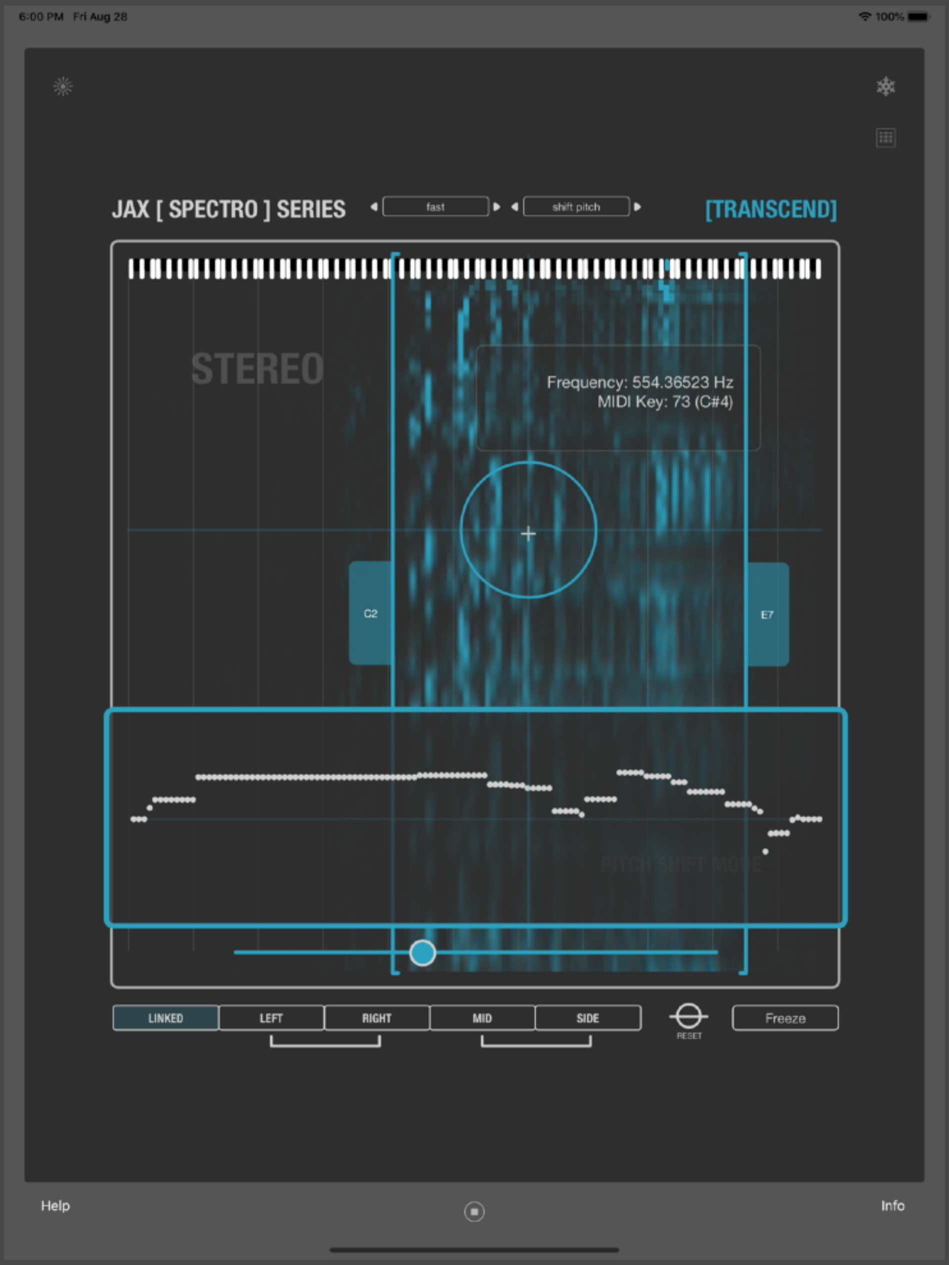
79	739,99
78	698,46
77	659,26
76	622,25
75	587,33
74	554,37
73	523,25
72	493,88
71	466,16
70	440
69	415,3
68	392
67	369,99
66	349,23
65	329,63
64	311,13
63	293,66

62	277,18
61	261,63
60	246,94
59	233,08
58	220
57	207,65
56	196
55	185
54	174,61
53	164,81
52	155,56
51	146,83
50	138,59
49	130,81
48	123,47
47	116,54
46	110

45	103,83
44	98
43	92,5
42	87,31
41	82,41
40	77,78
39	73,42
38	69,3
37	65,41
36	61,74
35	58,27
34	55
33	51,91
32	49
31	46,25
30	43,65
29	41,2

28	38,89
27	36,71
26	34,65
25	32,7
24	30,87
23	29,14
22	27,5
21	25,96
20	24,5
19	23,12
18	21,83
17	20,6
16	19,45
15	18,35
14	17,32
13	16,35
12	15,43

11	14,57
10	13,75
9	12,0098
8	12,0025
7	11,0056
6	10,0091
5	10,003
4	9,0072
3	9,0018
2	8,0066
1	8,0018



JAX SPECTRO I Multi Shift (release name: JAX [Tanscend])

This effect is probably very different than anything you have used yet. And so is the created sound by it.

We are (again) using the chromatic mapping approach, this time for individual pitch and harmonics shifting with single tonal bands. While a usual pitch shifter is shifting all frequencies by a certain factor (equally) across the entire spectrum, the well known side effects will occur. That is: shifting the formants of the material with the pitch, creating the awful! (awesome?) chipmunk! / helium sound.

It sounds even that way with obviously simple and clear monophonic voices, because complex tonal voices (like the human voice or expressive solo instruments for instance) have allot of formants, transients and sub harmonics, that all will be stretched unnaturally by this.

Formants are harmonics and inharmonics, that are not moving equally with the fundamental frequencies. Transients are even entirely different and will not move with the pitch at all.

The advantage of all such simple "deforming" pitch shifters is, that these can be applied to polyphonic material without a big problem, where special monophonic pitch shifters may be optimized for formant preserving and thus will not work with polyphonic material. Internal the latter processors will work completely different and require a clearly defined fundamental frequency (a solo voice).

However, JAX SPECTRO Multi Shift is not created for eliminating the formant stretch nor is it created for formant-corrected pitch shifting, but for selective creative range/tone shifting of an unusual kind.

If a range in the sound can be isolated well, so the chance for a selective pitch/formant/harmonic shifting is always there, creating endless new possibilities for modification and also for enriching a mix.

JAX Multi Shift can actually mangle any sound to its unheard extremes. It can unveil parts of a sound, that was not obvious and it can change the tonal relations to something completely different. It even can create controlled disorder of tonal frequencies due to its strict chromatic separation.

There are our well known 128 bands of tonal mapped frequencies (corresponding to the MIDI notes), which can be tweaked individually. The bands can be modified in 3 fundamentally different ways (also filtered}.

The single band sliders will be adjusted similar to the filter in the JAX SPECTRO [Isolate] (the drawer control), but the main purpose here is the adjustment of individual values for spectral manipulations.

So there are the 3 different modi:

1.) The chromatic pitch shift

In this mode, the pitch of each band is shifted individually. This happens by scaling the frequencies and their magnitudes with individual factors for each band. This is an isolated procedure. If all bands have the same pitch factor, for instance, the classic "pitch shifter" sound is created this way, although with some certain processing overhead but with the same amount of "heliumizing" and "chipmunkyfication".

2.) The chromatic harmonic shift

Rather than scaling the frequencies together with the magnitudes for each band, only the magnitudes will be shifted (moved to another position) and levelled with this mode, the frequencies remain at their original positions inside the spectrum and are not altered. Everything will pass the resynthesis (inverse FFT) then finally.

The resulting sound is completely different than pitch shifting but can get heavily out of tune too. This is caused by an extreme emphasizing or suppressing of individual formants, overtones and harmonics, which can create and isolate such strong new tonal content, that the impression of "pitching" is created as an audible side effect.

This all works so good with our 128 band scheme, because the frequencies are strictly separated by their tonal groups. It won't work with a usual phase vocoder very well, because FFT is a linear frequency process and will deliver quite strange, musically useless, atonal results then.

3. The chromatic filter

The filter works as a strong isolator of tonally separated content. Only the magnitudes of the single tones are scaled for each band individually from zero to a maximum factor of 4. This creates a chromatic multi-bandpass/bandstop filter/booster with 128 individual bands, that also can be used as an advanced EQ, a consequently chromatic (musical) EQ with swiss knife sharp slopes.

~~4. Combination mode 1~~

~~The fourth mode combines the mode 1 (pitch shift) with mode 3 (filter).~~

~~Pitch shifting can create strong unwanted harmonics, that can be suppressed and quiet parts can be boosted with the additional filter band adjustments.~~

~~5. Combination mode 2~~

~~This mode will combine mode 2 (formants) with mode 3 (filter).~~

~~Formants, sub harmonics and tones can be effectively fine-adjusted with the additional filter.~~

~~In the combination modi, 2 different drawer controls will be invoked and adjusted. You switch between these two with the buttons on top of the user interface.~~

By drawing individual control curves for the bands, the sound material can be adjusted freely and extremely modified in its entire tonal content. Each of the single parameters also can be automated. So this effect is more a creative sound manipulation tool and not specifically thought for corrective mastering tasks, like some of the other SPECTRO SERIES effects. But even with pre-mastering it can do certain effective things of improvement.

You can use the drawer and the range controls and then apply individual shifting to each single band inside the selected tonal spectrum. You have tonal control over the modified spectrum, more than a usual spectral filter ever could offer. Because a “tone” is something that can be understood by a musician with ease. A linear or even logarithmic filter is rather something “constructed”.

A bass line could be isolated and shifted by an octave or a melody by a fifth semitone in “chromatic pitch shift” mode. Some formants could be modified for a well separated range of single tones in “chromatic harmonic shift” mode. A singer can be cut out of a mix completely, with the rest of the song left and so on.

Most probably, you will just experiment with the tools and see (hear), what kind of spectrality can be unleashed with this. But also even very small adjustments of selected tones and ranges are often producing astonishing additives and can be applied with the dry/wet control to the entire mix.

The JAX SPECTRO | Multi Shift is build for internal mixing, so that the sound can be faded with the unprocessed part, without using an extra bus for send effects. This also has the advantage, that the original sound is internally latency compensated, this means, the input is delayed to the exact amount of samples of the FFT latency. External routing would probably produce a problematic (audible) sample offset with some resulting stereo phase problems. Not all host applications will implement latency compensation mechanisms for send effects or even at all.

Pitch and harmonics shifting, when added, will generate new frequency content from the available audio material. This can be used to fill up frequency gaps and also excite a mix in certain parts.

There is no general rule how to use JAX SPECTRO Multi Shift. We hope, you will like this effect so much how we do, and we are sure it will enrich your arsenal of the worlds most unusual but also very useful sound manipulation tools.

What is the main difference between JAX CHROMATIC [Isolate] and JAX CHROMATIC [Transcend]?

JAX Transcend does not have the special mastering controls for (left/right and mid/side processing). The JAX [Isolate] Filter is more accurate, has higher resolution and the entire effect is optimized for special mastering tasks.

The JAX SPECTRO | Analyzer

The analyzer delivers several classic octave division analyzers and also the grand 128 MIDI tone mapping (chromatic), known from our JAX Chromatic Analyzer. Despite of its completely different analysis method, both tools deliver similar results, whereby the FFT based approach is much more performant and much more narrow (separated). But the FFT approach has a small latency.

The frequency graph is consequently mapped to the musical frequency scale, this means, rather than several approximative logarithmic scalings, we use the exact tonal range of the MIDI frequency mapping also for the displayed graphs. This way, the actual keys can be identified visually with ease. Other analyzers often focus too much on the overtones, which is musically quite irrelevant and also confusing.

The JAX Spectro Analyzer is able to generate (polyphonic) MIDI output of the analysis results, based on effective range selection.

The JAX SPECTRO | Shift

This is a classical pitch shifter. FFT based pitch shifters have quite smooth results but can introduce phase problems (so-called smearing of transients and some problems with the stereo phase). The formants are always shifted proportionally with the pitch, the advantage of this is, that polyphonic audio can be shifted very well with it, even to extreme amounts and anti-aliasing is also no problem.

Many realtime time-stretch algorithms use this kind of pitch shifting because of the much smoother result.

FFT based shifters are ideal for sounds with low transients and for wide stereo sound (like convolution effects and reverbs). Delayline-based and granular based approaches tend to get gritty with higher amounts.

The JAX SPECTRO | Tonal Range

This effect allows to select a range of a user selected tones and performs either an ultra steep bandpass or a bandstop filter. It is pared with our graphical flow analysis tool and can expose a traditional frequency graph too.

This unit is included with the JAX SPECTRO | Tonal Flow.

The filtering is phase stable, as long there are no special phase shifts are applied intentionally.

A spectral filter is ideal for filtering with steep slopes, for instance for exactly separating, extracting or cutting out certain frequency bands from the audio signal.

The JAX SPECTRO | Tonal Adjust

Like our JAX Chromatic Series, which are based on an analog filter bank, this effect provides selected tone range filtering of the audio signal with several octave divisions per band. The default mode is the consequent musical (chromatic) mapping, where all frequency bands are centred to the MIDI keys in the range of the 128 defined semitones.

A FFT filter is much more performant and has a much steeper slope than its analog counterparts. Therefore the separated ranges sounds much more artificial.

The JAX Spectro Filter can be used for creative filtering in realtime or for surgical sharp frequency modifications, which are not possible with analog filter bands. It can of course also be used like a usual EQ or bandpass/bandstop filter.

The filtering of the bands is phase stable, transparent and auto-levelling without artefacts like distortions or unwanted saturation effects.

The JAX SPECTRO | Compander

Like our analog modelled filter bank in the JAX Chromatic Series, this effect is able to compress or expand the loudness relations of the processed filter bands.

The compander auto-adjusts the filters based on a bank of 128 envelope followers and is also phase stable. The speed of the compression/expansion is dependent of the speed adjustments of the follower for each single band. The Multiplier can get negative, which effectively expands. Values above zero will compress.

Analog to the special tools of the JAX Chromatic Series, there may be several special tools based on the JAX Spectro Series, which are available separately.