

# **IRCAM Trax**

FLUX:: Immersive

2/6/23

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**1**

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# T.R.A.X.

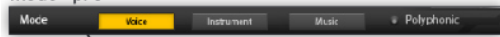
Real-time Voice and Sonic Modeling Processor



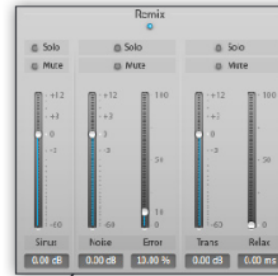
# Part I



Mode - p. 6



Remix - p. 13



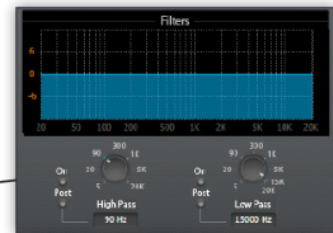
Target - p. 10



Source - p. 7



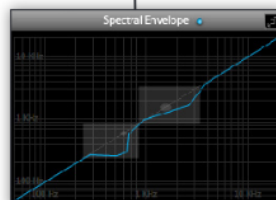
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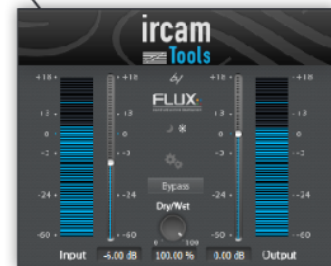
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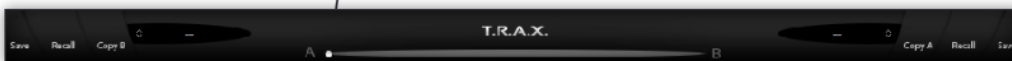
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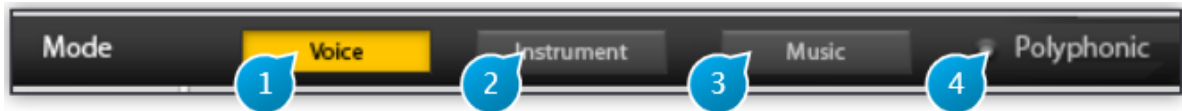


Main Input / Output section - p. 18



Preset section - p. 23

## 2 Mode



The Mode selection buttons puts the plug-in into the one of three modes, each best suited for one of three different types of audio material: Voice, Instrument and Music, which are explained in detailed below.

Mode serves as a starting point upon which to base more precise tweaking to achieve the desired result. It selects the type of algorithm used internally by the plug-in, initializes source settings with all-round usage values, amongst other things.

Selecting the right mode for your material should be the first step you perform before doing any further editing.

Please note that the algorithms used to process the signal are very different for the three modes, so setting the right mode here is crucial to proper operation, and failing to do so would heavily compromise the quality and accuracy of the audible result.

In some cases however, for example if you're crafting special effects for sound design, are interested in deliberately emphasizing artifacts or maybe creating distorted robotic voices, well of course, nothing should stop you from trying even the most weird and technically absurd combinations, your imagination and taste is the limit!

### 2.1 (1) Voice

Voice Mode is specifically targeted at voice processing, be it spoken word or sung material. By default, the Polyphonic option is disabled in this mode, as a monophonic voice (one speaker or singer) is the most common case. If you're dealing with a choir or harmonized voice, you should enable the polyphonic setting - but not when processing double-tracked vocals.

## 2.2 (2) Instrument

Instrument Mode should be used when dealing with any kind of material other than vocals, whether this originates from an acoustic, analog or digital source recording, as long as it originates from an instrument that is playing individual pitched notes. Polyphonic is disabled by default in this mode, and should only be enabled with a true polyphonic recording, such as a guitar or piano playing chords or a melody accompanied by a bass line. A mix of two different instruments, each playing an independent monophonic line, does not fall into this case, as it involves two different timbres, and the Music mode should therefore be used instead.

## 2.3 (3) Music

Music Mode should only be used for treating a full mix, that is material consisting of several instruments playing. In this mode, the Polyphonic option is enabled and locked, and processing is restricted to a global transposition, so the plug-in operates as a high-quality pitch-shift unit.

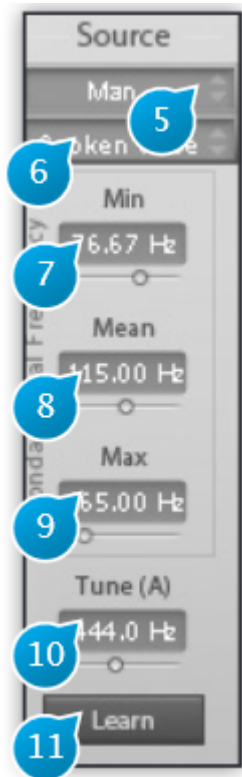
## 2.4 (4) Polyphonic

When enabled, the Polyphonic option disables some functions such as the pitch-tracking module. Pitch-tracking and extraction of overlapping notes in a reliable and artifact-free manner is currently extremely difficult to achieve and generally requires advanced user interaction.

Here monophonic means one-note at a time, and not monaural (single channel). The plug-in itself can naturally process any channel configuration, be it mono, stereo, as well as surround recordings. The mono sum of all input channels is used for the detection.

Sometimes it is not so straightforward to decide whether the source is monophonic or not, as in the case of a complex electronic sound texture, or if a monophonic part is drenched in echo and reverberation, leading to notes that overlap in time a lot. In this case you should also try the Music setting and let your ears decide which is best.

### 3 Source



The source panel settings are critical in obtaining a good and natural sounding result, should the latter be your objective, so you should always ensure to get these right before attempting any actual transformation of the audio content. The Source description is used to derive some fundamental processing parameters that are essential for high quality sound analysis and processing.

The source signal will be split into three distinct components: \* Pitched content \* Noise \* Transients or onsets

And moreover 2 sets of descriptors are extracted:

- The fundamental frequency and its variation over time
- The formant and resonator characteristics

Based on these analysis the subsequent transformations will be performed.

This analysis being a very sophisticated process is based on decades of research made at IRCAM.

The analysis make uses of a little prior information about the source material, namely the range within which the fundamental frequency of the audio is known to vary. This range can be defined in three ways, using high-level to low-level parameter manipulation: \* using named presets \* using learn mode \* manually fine-tuning low-level settings

### **3.1 (5) Source Preset**

This setting, only relevant to voice mode, should match the recorded singer's or speaker's age and gender.

Available presets: \* Man \* Woman \* Young Man \* Young Woman \* Boy \* Girl

### **3.2 (6) Source Registers**

The source register setting is only relevant to voice and instrument modes, and should match the source material register as closely as possible.

#### **3.2.1 Available presets in Voice mode:**

The following list of registers is available in Voice mode \* Spoken Voice \* Singing Bass \* Singing Baritone \* Singing Tenor \* Singing Counter Tenor \* Singing Contralto \* Singing Mezzo Soprano \* Singing Soprano

The selection of spoken and singing registers changes not only the analysis parameters but also the transformation algorithms that are linked to high level target controls. Transforming Man into Woman with Spoken voice will affect formant structure and pitch, while transforming Man Singing Tenor into Woman Singing tenor will only affect the formant structure.

#### **3.2.2 Instrument mode:**

The following list of registers is available in Instrument mode. If you do not know the register of the source, you can resort to manual tuning of the fundamental frequency, and refer to the chart below \* Undefined. Default, general setting \* Bass \* Baritone \* Tenor \* Counter Tenor \* Contralto \* Mezzo Soprano \* Soprano

### 3.3 (7) F0 Min

This low-level parameter defines the minimum of the allowed variation range for the source fundamental frequency. When doing manual adjustments, this frequency should roughly be set to that of the lowest note expected in your source material. This parameter determines the window size for monophonic material. Setting it too low may introduce reverberation artifacts, may prevent the detection of fast onsets, and moreover, may introduce pitch detection errors. Setting it too high will create artifacts in low pitch notes and may as well lead to fundamental frequency estimation error.

### 3.4 (8) F0 Mean

This low-level parameter sets the expected average fundamental frequency. You can tune this by ear or determine it precisely using the frequency corresponding to the note most often played, for example, or the root key of the song if you know it. This value is used mainly for the ambitus control in recto tono mode.

### 3.5 (9) F0 Max

Maximum of the allowed variation range for the source fundamental frequency.

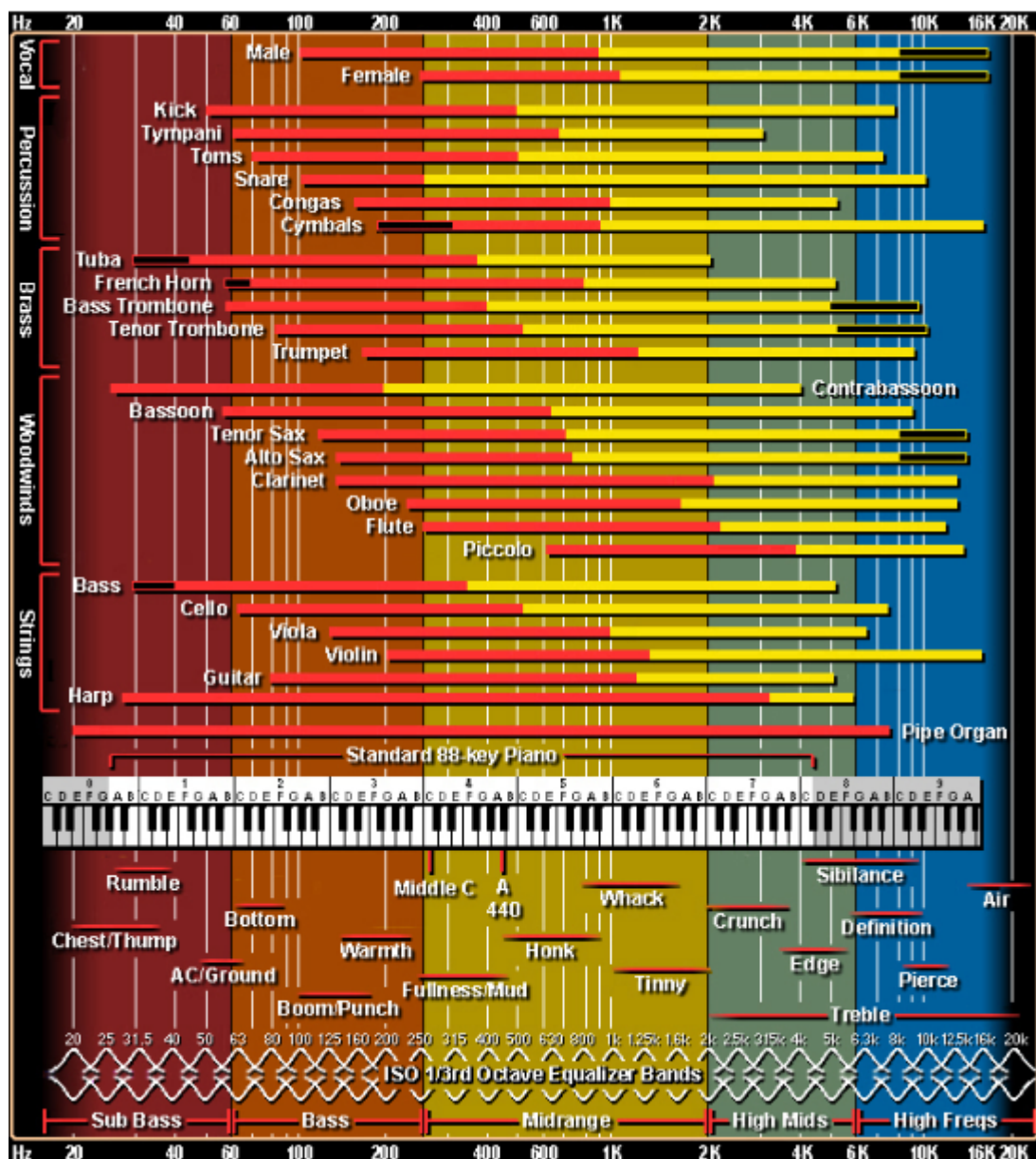
### 3.6 (10) Tune (A)

Sets the actual reference tuning used in the source material, which is defined as the frequency of A4 in Hertz. The default value is 440Hz, which matches the standard concert tuning almost always used nowadays.

This frequency will be used as the reference for transposition; you should therefore adjust it accordingly if your instrument is tuned in a non-standard way, or working with a recording of unknown origin.

### 3.7 (11) Learn

The Learn button, when engaged, puts the plug-in into fundamental frequency parameter learn mode, wherein the incoming audio is analyzed to refine F0 min, mean and max values. The learn mode is linked to the presets and can be used only to refine preset parameters. It will not work if the selected preset does not match the audio.



(c) [http://www.independentrecording.net/irn/resources/freqchart/main\\_display.htm](http://www.independentrecording.net/irn/resources/freqchart/main_display.htm)

## 4 Target

For spoken voice it is good practice to playback between 5 to 15 seconds of any new audio material with Learn engaged to let the plug-in adapt to the specifics of the audio material. For singing voice or instrument sounds it is essential that the audio segment that is used covers the complete pitch range that will be processed. Otherwise the internal parameter settings will not achieve optimum quality for the transformation.

!> Do not forget to switch off learn mode when doing the actual processing, or you may encounter strange behavior!

Now that you have set the source parameters, here comes the fun part: transforming the source material, according to target parameters. T.R.A.X. allows for a large palette of transformations such as:

- \* Pristine quality up and down pitch-shifting, preserving formant and transient integrity
- \* Re-tuning the material to a fixed-pitch, a given musical scale, etc.
- \* Altering perceived gender, age, etc.
- \* Special effects for sound design and such





In all of these contexts, unwanted artifacts are kept to a minimum and the natural quality and timbre of voices and instruments are preserved, thanks to the innovative analysis algorithms used.

Please note, however, that in the case of extreme transformations, the perceived quality of the voice might degrade. We strongly believe this not due to a flaw or a limitation of the algorithms, but more to the fact that the human brain is extremely well trained and sensitive to the perception of voice, with all its intricacies. In other a words, please do not reasonably expect a computer to turn an amateur singer into a seasoned Opera pro or your 5-year old son into the 8 o'clock veteran news presenter. This would involve something much more complex than pitch and formant modification, such as prosody, and some understanding of the meaning of the underlying words or musical emotion conveyed.

## 4.1 (12) Target Preset

Only available in Voice Mode, the Target Presets cover a selection of gender and age combinations. Available presets

- Man

- Woman
- Young Man
- Young Woman
- Boy
- Girl

## 4.2 (13) Target Register

Available in Voice and Instrument Mode, the Target Register presets allow to change the register of the material. Ensure source and target registers match if you want to preserve register.

Register is locked to spoken voice when the source is defined as a spoken voice also, as in this case there is no “melody” for the target voice to follow.

### 4.2.1 Registers available in Voice mode:

- Spoken Voice
- Singing Bass
- Singing Baritone
- Singing Tenor
- Singing Counter Tenor
- Singing Contralto
- Singing Mezzo Soprano
- Singing Soprano

### 4.2.2 Registers available in instrument mode:

- Undefined. Only choice available if “Undefined” is selected as Source register.
- Bass
- Baritone
- Tenor
- Counter Tenor
- Contralto
- Mezzo Soprano
- Soprano

### **4.3 (14) Transpose**

Applies a global transposition, also known as pitch-shift to the incoming audio, expressed in semi-tones. This affects the pitched content only, preserving formant and transient content for the most natural result. An octave shift corresponds to a Transpose factor of 12 semitones.

### **4.4 (15) Formant**

Applies a global frequency-shift of the formant content, independently of pitched and transient content.

### **4.5 (16) Link**

When engaged, the Formant shift factor is locked onto the transpose factor value, and only the Transpose slider can be moved.

### **4.6 (17) Inv.**

When the inverse button is engaged, the Formant shift amount follows the Transpose factor, in inverse fashion, i.e. formants are shifted by an amount opposite of the pitch.

### **4.7 (18) Voice Forger Target F(0)**

The VoiceForger sub-section is used for more sophisticated pitch and formant transformations.

Here the incoming source characteristics extracted by the detection stage are used as input to a series of modifiers. The first possible modifier,  $F(0)$ , changes the fundamental frequency of incoming notes, “squeezing” them back into a series of reference target intervals, defined by the modes which we explain below, with the Flat $\leftrightarrow$ Expr. controlling the “tightness” of this “squeeze”.

This can be used to correct a less than perfect take, impart a robotic character to a voice, make an instrument sound like a synthesizer, etc.

- **Recto Tono**  
Modifies the pitch with regard to the reference mean F0 as defined in the Source parameters, by an amount specified by the Flat<>Expr (FE) control. With FE set to -100% (Flat), no pitch variation remains in the processed sound, which gives a completely monotonous voice, reminiscent of early voice synthesizers.
- **Recto Tono (Tuned)**  
Same as above, but taking into account the source tuning.
- **Chromatic**  
Brings back the pitch of the notes towards a chromatic, equal-temperament scale.
- **Mode 1 tr 1 (tone)**  
Attracts pitch towards the closest note on a succession of full-tone (2 half-tones) intervals.
- **Mode 1 tr 2 (tone)**  
Same as above, transposed 1 semitone above root key.
- **Mode 2 tr 1 (tone, semi tone)**  
Works on a succession of intervals made up of groups of a full-tone followed by a semi-tone.
- **Mode 2 tr 2 (tone, semi tone)**  
Same as above, first interval is a semi-tone.
- **Mode 2 tr 3 (tone, semi tone)**  
Same as above, first interval is a tone.

## 4.8 (19) Integration

This controls smoothing of the time-variations of the fundamental frequency. Increasing it will therefore give more progressive note-to-note transitions, for a more “legato” playing type of feel.

## 4.9 (20) Flat <-> Expr. (Voice and Instrument Mode only)

The Flat <-> Expression knob defines the amount of pitch correction taking place. When set to “Flat” (-100%), this controls how close towards the target reference notes (single note for Recto Tono, chromatic scale, etc.) the fundamental frequency is pulled. Conversely, when set to “Expr.”, it exaggerates the pitch differences with respect from the target reference notes. As you expect, a 0% expression setting leaves the pitch unaffected by this section.

#### **4.10 (21) Young <-> Old (Voice Mode only)**

This high-level control gradually alters the perceived age of a voice, either sung or spoken.

#### **4.11 (22) Male <-> Female (Voice Mode only)**

This high-level control gradually alters the perceived gender of a voice, either sung or spoken.

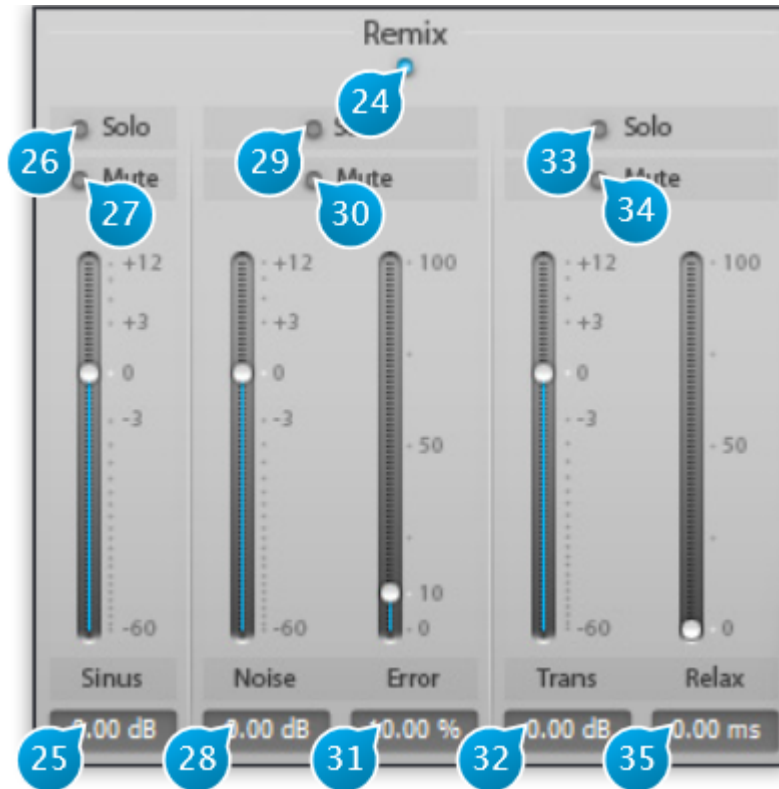
#### **4.12 (23) Breathy (Voice and Instrument Mode only)**

Increase the “breathiness” of a voice, or an instrument, making it whisper.

Increasing this close to 100% will give you that chain-smoker voice timbre, ala Don Corleone in the Godfather, without the risks and costs involved. You can also use this with a wind instrument to accentuate the breathing sound which gives so much character to a performance.

This panel acts as a mini-mixing console where the various components of the audio are sent to and summed-back to the output. It allows for further tweaking of the final result and isolation of each part of the sound, and one can also use this to achieve special effects.

## 5 Remix



### 5.1 (24) Remix On / Off

Toggles the Remix console in and out of the signal path.

When disengaged, all signals are simply summed back with equal gain, and the processing does remain active.

## **5.2 (25) Sinus**

This channel contains the sinusoidal components of the source sound, with or without modification depending on target, modulation and spectral envelope settings.

Sinusoidal components represent the regular part of the sound, e.g, the pitched part of musical instruments or the long resonances of cymbal and bells

## **5.3 (26) Sinus Solo**

Isolates the Sinus Channel.

## **5.4 (27) Sinus Mute**

Mutes the Sinus Channel.

## **5.5 (28) Noise**

Contains the noise part of the material, i.e. anything that doesn't have a defined pitch and does not possess a transient quality.

## **5.6 (29) Noise Solo**

Isolates the Noise Channel.

## **5.7 (30) Noise Mute**

Mutes the Noise Channel.

## **5.8 (31) Noise error**

Sets the amount of allowed error, in terms of statistical content, at the analyzer stage. This controls how the pitched and noise content are allowed to overlap. Most of the time you can leave this at the default value of 10%.

## **5.9 (32) Transient**

Contains the transient part of the material, which has a rapidly changing energy profile. In other words, this is the attack or percussive phase of a sound, if any. Please note Transients are only computed when the Transient button is enable in the Options panel (See 47).

### **5.10 (33) Transient Solo**

Isolates the Transient Channel.

### **5.11 (34) Transient Mute**

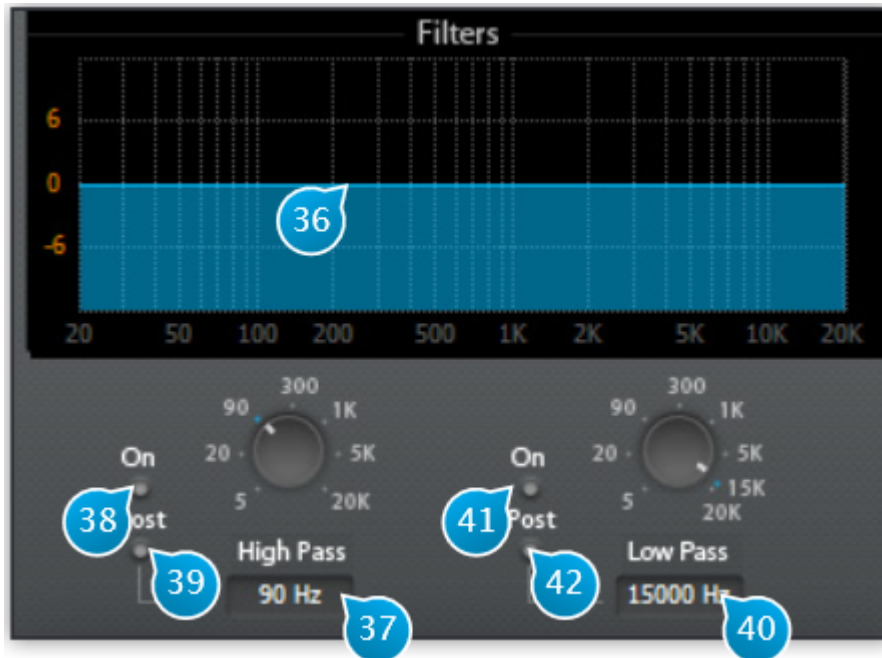
Mutes the Transient Channel.

### **5.12 (35) Transient relax**

Increasing this above the 0ms default setting allows to relax the transient detection scheme by the given amount, to let more of the tail of a transient pass through. The effect is similar to that of the release control found on a compressor A two-band filter section that can be applied on the incoming or outgoing material, depending on the Post setting.



## 6 Filters



### 6.1 (36) Filter Curve

The filter curve displays the overall frequency response of the filter section.

### 6.2 (37) High Pass

Inserts a 6dB/octave high pass filter in the signal chain.

### 6.3 (38) High Pass On

Toggles the High Pass Filter in and out.

## **6.4 (39) Post High Pass**

Determines whether the filter acts on the input or output signal. When off, which is the default setting, the corresponding frequencies are filtered out prior to analysis and transformation, which can be used to prevent possible tracking errors induced by any excessive rumble or mains interference picked up during recording. If you only want to equalize the signal, enable Post to circumvent this behaviour.

## **6.5 (40) Low Pass**

Inserts a 6dB/octave low pass filter in the signal chain.

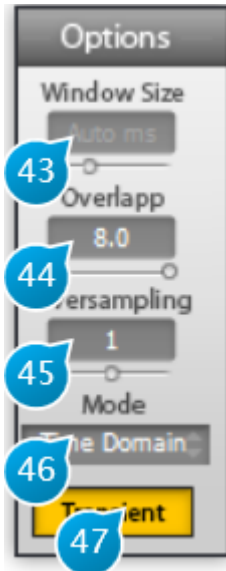
## **6.6 (41) Low Pass On**

Toggles the Low Pass Filter in and out.

## **6.7 (42) Post Low Pass**

See 39.

## 7 Options



The options panel controls a number of parameters affecting the analysis-re-synthesis engine. The default settings should prove adequate as a starting basis.

### 7.1 (43) Window Size (Music Mode only)

Computed automatically in Voice and Instrument mode, the window size is set manually in Music mode. The window size determines the time frame base used to extract time-localized data in time from the incoming material. As general guidance, window size should be small for fast tempos and larger for slow music.

### 7.2 (44) Overlap

This determines how much extraction windows overlap during a given time-frame, in other terms, the update rate of the analysis for a given window size. Increasing this setting usually gives better results but also leads to more CPU resources being used, as more data has to be

analyzed. The default setting of 4 is a good trade-off, but you can increase provided you have a reasonably fast machine and hear a noticeable difference.

### **7.3 (45) Oversampling**

Oversampling effects the sampling of the spectral representation of the source material. The quality difference achieved with oversampling is mostly noticeable with complex material with a broad frequency spectrum such as a full mix or a instrument such as the piano, but as the performance hit is quite high, you should only enable it if necessary, and if your hardware is sufficiently fast to handle the extra processing demands.

### **7.4 (46) Mode**

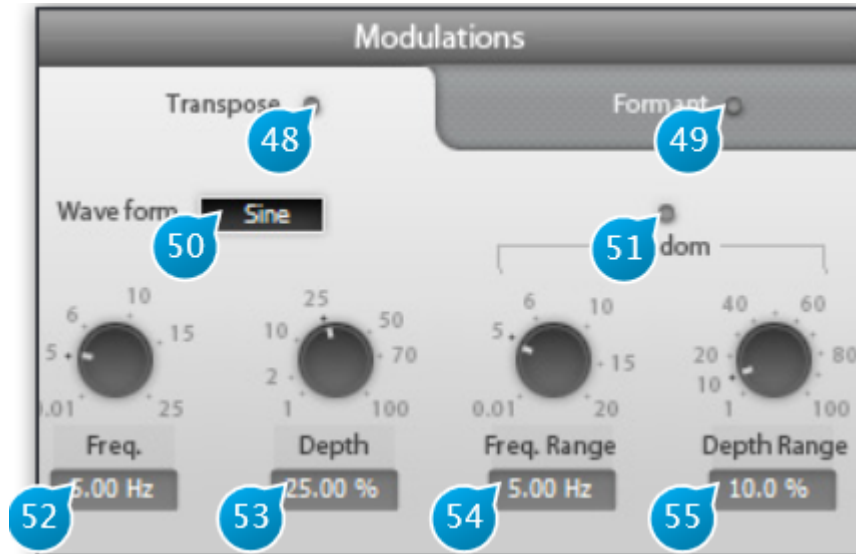
Sets the internal analysis/re-synthesis engine type to use.

- Auto: default setting, selects best setting for lowest CPU utilization and best quality depending on the value of transformation parameters
- Frequency domain: lowest CPU utilization, performs best when pitching up
- Time-domain: highest quality and CPU utilization

### **7.5 (47) Transient**

Toggles transient processing on (default setting) and off. You can disable transient processing if you don't need to treat transient separately, which will allow you to gain a little CPU. In this case however you will naturally not be able to control the transient level independently of other content.

## 8 Modulation



The Modulation panel allows for some additional modification of the material's pitch and formants.

### 8.1 (48) Transpose

Toggles pitch modulation on and off. This is akin to a LFO on an analog synthesizer, except with this you can apply it to any kind of material. You can easily achieve wild results here and go overboard if you're not careful, so it is advised to toggle this switch on an off quite often when doing adjustments to check you're not overdoing the effect.

### 8.2 (49) Formant On/Off

Toggles formant modulation on an off.

### 8.3 (50) Waveform

Three classic modulation waveforms are available:

- Sine
- Triangle
- Sawtooth

Deciding which is best is really a matter of taste and depends entirely on the incoming material.

### 8.4 (51) Random

As the name implies, this adds some random amount to the modulation, which is great to make pitch variations seem less obvious and predictable.

### 8.5 (52) Freq.

Modulation Frequency in Hertz (Hz). The faster this setting, the faster the pitch and/or wobble. Can be used to make a voice quaver at a few Hz, do whammy effects popular with electric guitar players, simulate an old warped record, but be warned this can induce seasickness if used without restraint.

### 8.6 (53) Depth

Depth of the modulation, in percent. Determines the amount of rise and fall of the pitch and/or formants. Go for a value around or below the default 25% unless you're looking for extreme results.

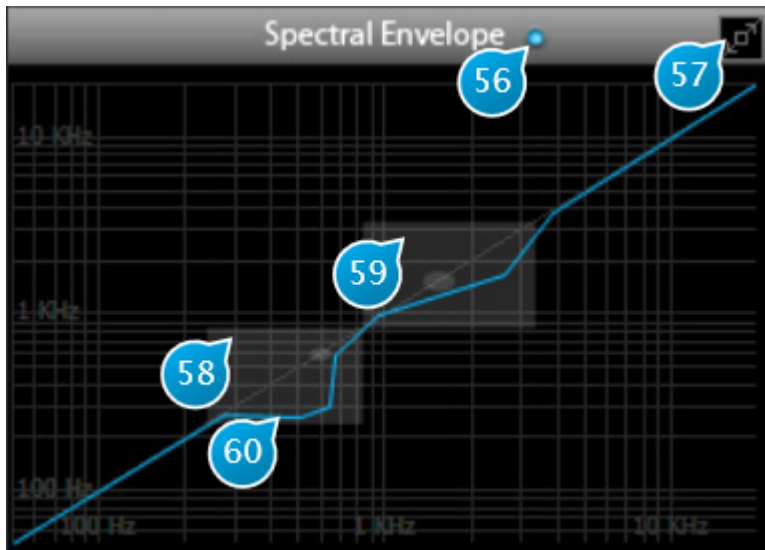
### 8.7 (54) Freq. Range

When the Random button is on, this sets the extent of the random variations around the modulator base frequency.

## 8.8 (55) Depth Range

Determines the amount of randomization applied. Setting this to 100% gives completely random modulation, going towards 0% makes the modulator resemble more to the original waveform.

## 9 Spectral Envelope



The Spectral Envelope allows for complex remapping of the spectrum envelope, according to a freely definable curve. At startup, the spectral envelope is a diagonal line, giving a 1:1 remapping of the spectrum, i.e. no modification.

### 9.1 (56) Spectral Envelope

Toggles Spectral Envelope processing on and off.

### 9.2 (57) Full Size Display

Toggles the size of the Spectral Envelope editor, for detailed editing.



### 9.3 (58) First formant region

The first transparent gray square indicates the region of the first formant (200...800Hz) for an average human voice, which corresponds to the first resonance of the vocal tract, and the gray disks to the average peak resonance location (500Hz). Alongside with the second formant, this is the area that has the most influence when processing voice.

### 9.4 (59) Second formant region

Second formant, ranging from 600 to 2.8kHz, peaking at 1.5kHz.

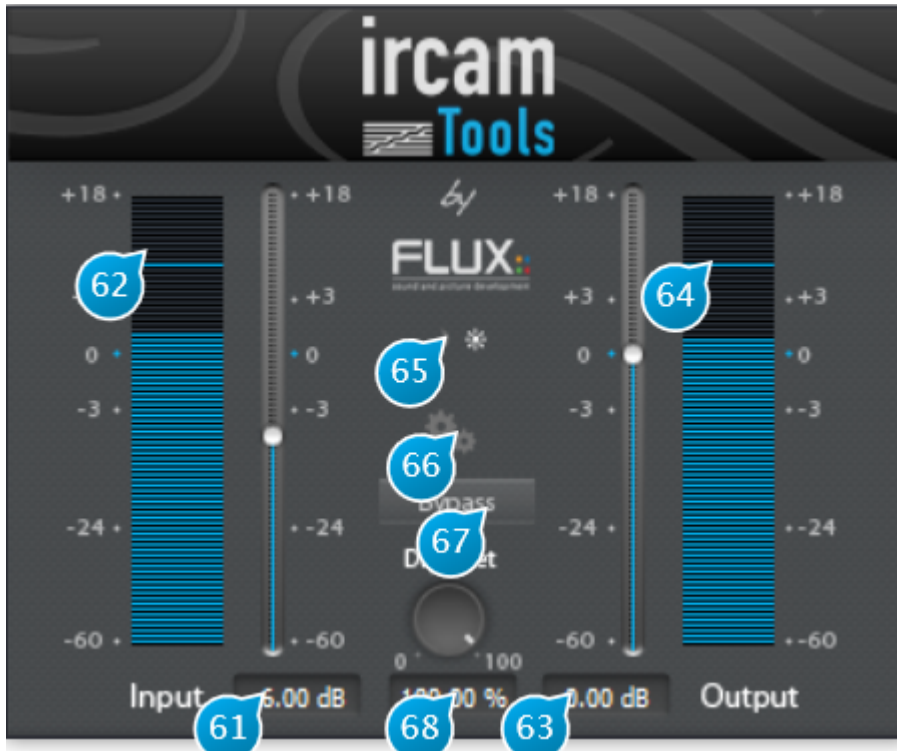
### 9.5 (60) Spectral Envelope Curve

The curve defines how the input spectrum frequencies, on the horizontal axis, are remapped (transposed) on the vertical axis.

Defining and modifying the curve is done by manipulating line segments using the following methods

- Double-click and drag on the diagonal blue line to add and place a control point
- Click and drag an existing point to move it
- Alt-click an existing point to snap it back to the diagonal line
- Ctrl-click (MacOS: Apple/Command + click) an existing point to constrain its movement and prevent it to go past or beyond its immediate neighbors, which is useful when doing fine adjustments with a lot of points close to each other.

## 10 Main Input - Output Section



### 10.1 (61) Input Level

Adjusts the level of the signal fed to the plug-in, in dB increments.

### 10.2 (62) Input level-meter

Shows the current peak level of the input signal after applying input gain, in RMS, with reference at -18 dB FS.

## **10.3 (63) Output Level**

Used to trim the output signal and possibly avoid any overloading of the signal in the rest of the signal-chain.

## **10.4 (64) Output level-meter**

Shows the current peak level of the input signal after applying output gain, in RMS, with reference at -18 dB FS.

## **10.5 (65) Day - Night**

Toggles between two interface schemes, which, as the name implies, are best suited to high or low light environments respectively. In a dimly-lit studio environment, switching to the nighttime scheme with its darker color palette and lower contrast helps to minimize eye-fatigue when doing long sessions.

## **10.6 (67) Bypass**

Bypasses the plug-in processing by routing the input direct to the output. The actual processing is still performed in the background allowing for a true and smooth transition between the processed and the actual incoming signal.

## **10.7 (68) Dry/Wet**

When used as insert effect, one can dial the right amount of “wet”, processed signal with respect to the “dry”, direct input signal.

The default 100-percent wet setting is mostly intended for the typical and preferred use in a send-effect configuration. You can get some interesting chorus and harmonizer type effects if you blend the dry and wet signals.

## **Part II**

# **Cross Synthesis & Source Filter Tools**



## Introduction:

These plugins can only operate in a fixed 2 input - 2 output channel configuration only at the time of this writing. The plugin treats the stereo input as a dual mono input, and the two channels stereo output are identical. You will most likely want to use two separate DAW channels to be processed. The most convenient way to achieve this is to create a dedicated buss, with the plugin as an insert, and two separate direct sends from each source channels to this buss, panned hard right and left. You should probably also bring the corresponding channel faders all the way down and set the send as “pre-fader” to hear the result of the processed signal only.

# 11 Cross Synthesis



## 11.1 (69) Oversampling

Adjusts the spectral domain oversampling factor, set to zero (none) by default. Increasing this setting can increase the processing the processing quality, depending on the audio material, although at the expense of higher CPU usage, which roughly doubles for every increment.

## 11.2 (70) Overlapp

This determines how much extraction windows overlap during a given time-frame, or in other terms, how often the analysis is updated for a given window size. Increasing this setting usually gives better results but also leads to more CPU resources being used, as more data has to be analyzed. The default setting of 4 is a good trade-off, but you can increase provided you have a reasonably fast machine and hear a noticeable difference.

## **11.3 (71) Window Size**

Window size determines the time-frame base used to extract time-localized data in time from the incoming material. It is the most important parameter for all spectral domain signal transformations. It affects the capability of the algorithm to detect and identify the individual components of the signal (sinusoids, noise and transients/onsets) and to treat them independently. As general guidance, window size should be small for fast tempos and larger for slow music and at the same time larger for spectrally dense or low pitched sounds and smaller for spectrally sparse or high pitched sounds. In some cases, e.g. fast orchestral passages (that are spectrally dense and at the same time fast) the spectral and temporal characteristics ask for different windows sizes. In these cases finding the optimal window size requires some experimentation with different settings.

## **11.4 (72) Transient**

For the T.R.A.X. Cross Synthesis plug-in, toggles transient processing on and off.

## **11.5 Amplitude:**

This sub-panel determines how the output amplitude is affected by the left and right channels.

### **11.5.1 (73) Amplitude Left**

Sets the percentage of the left channel used to derive the output amplitude.

### **11.5.2 (74) Amplitude Right**

Sets the percentage of the right channel used to derive the output amplitude.

### **11.5.3 (75) Amplitude Link**

Links left and right channel amplitude controls in inverse fashion, which gives a kind of panning control.

#### **11.5.4 (76) Product**

Determines the amount of the product of the amplitudes of both channels that will be mixed into the output amplitude.

#### **11.5.5 (77) Power**

Exponent of the power law that is applied to the amplitude product. Default value of 0.5 corresponds to a square root, which produces the geometric mean of the amplitudes of both channels and is the setting that best preserves the signal power of the product amplitude spectrum when compared to the original input amplitude spectra. Smaller values of the power control will equalize low and high amplitude components while larger values will amplify the differences in the amplitude spectrum (amplifying spectral components in locations that are strong in both input spectra and suppressing spectral components that are weak in either or both input spectra).

### **11.6 Frequency :**

This sub-panel determines the frequency content of the output with respect to the inputs.

#### **11.6.1 (78) Frequency Left**

Sets the percentage of the left channel used to derive the output frequency spectrum.

#### **11.6.2 (79) Frequency Link**

Controls the gain applied to the dynamic processing input. This setting may affect the dynamics signal detection.

#### **11.6.3 (80) Frequency Right**

Sets the percentage of the right channel used to derive the output frequency spectrum.



## 12 Source Filter



The Source Filter is an advanced type of vocoder to morph the spectrum and dynamics of one signal into another.

The plugin relies on a source-filter model of sound to gradually combine a source track and a filter track into the output, by gradually mixing their dynamics, i.e. time envelope or signal level contour, and spectrum, i.e. the structure of amplitudes versus frequency.

The source-filter model fits the physical description of human voice production quite well, with the source being the vocal cords and the filter the vocal tract. Some instruments fit this rough description too. For example, for the clarinet, the source would be the reed and the filter the instrument cavity. Simple analog synthesizers also use the source-filter paradigm, in form of one or more frequency-controlled oscillators fed to various kinds of low-order filters.

NB: Ch.1 corresponds to the left channel of your DAW track, Ch2 to the right one.

## 12.1 (81) Oversampling

Adjusts the oversampling factor, set to zero (none) by default. Increasing this setting can increase the processing the processing quality, depending on the audio material, although at the expense of higher CPU usage, which roughly doubles for every increment.

## 12.2 (82) Overlap

This determines how much extraction windows overlap during a given time-frame, or in other terms, how often the analysis is updated for a given window size. Increasing this setting usually gives better results but also leads to more CPU resources being used, as more data has to be analyzed. The default setting of 4 is a good trade-off, but you should try to go above this value if you have a reasonably fast machine and hear a noticeable difference.

## 12.3 (83) Window Size

Window size determines the time-frame base used to extract time-localized data from the incoming material. As general guidance, window size should be small for fast tempos and larger for slow music.

## 12.4 (84) Temporal envelopes

The slider controls the blend of dynamics between the two channels. In other words, this controls how much of channel 2's versus channel 1's dynamics will be imprinted onto the output signal.

## 12.5 (85) True Env.

Sets track analysis to True Envelope mode, an advanced proprietary IRCAM algorithm.

## 12.6 (86) Max F(0)

When True Envelope mode is selected, this sets the maximum allowed fundamental frequency to track, depending on the input material. Avoid setting this unnecessarily high as this might increase the possibility of tracking errors.

## **12.7 (87) LPC**

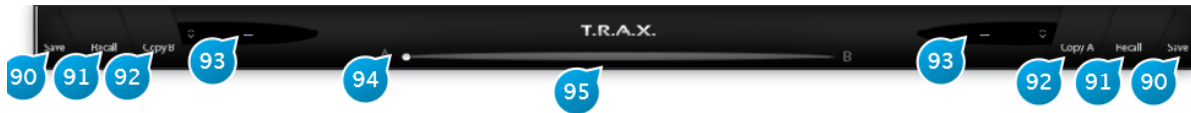
Sets track analysis to LPC (Linear Predictive Coding, for filter coefficients prediction) mode, which is a classic source-filter estimation method.

## **12.8 (88) Order**

Determines the order or number of coefficients used in the LPC prediction algorithm. The higher this setting, the better the LPC can adapt to the hills and valleys in the spectral envelope, but this also increases the chance of over-adapting to the short-term variations in the signal, which leads to artifacts.

## **12.9 (89) Mix**

## 13 Preset Section



### 13.1 (90) Save Preset

Saves a snapshot of the current settings for future use.

Short description and assorted comments can be provided, which comes in especially handy when sharing presets with other users, when the preset is part of a large preset bank, or to identify the author and source.

Entering a descriptive keyword is a good practice to be able to quickly sort your presets, according to character, the type of space they simulate (e.g. hall, room, etc.), and the intended usage (e.g. voice, percussion, guitar, etc.)

A preset can be locked to prevent any further editing.

To re-save your preset under a new name, open the preset manager by clicking the corresponding (A/B) preset slot, then select New, enter a name for your preset, and finally press Save.

### 13.2 (91) Recall Preset

Recall the settings from the currently selected preset, overwriting any current settings of the plug-in.

### **13.3 (92) Copy A-B**

Copy current settings to the other parameter slot (A to B or vice versa).

To try out a variation of the current settings without erasing the reference, press this button, switch to the other preset slot, and adjust your parameters of choice, then switch or morph between them.

When copying a preset to a slot, the morphing slider will automatically fly to the corresponding slot.

### **13.4 (93) Preset Name**

Displays the current preset name, if any. Clicking the associated button (up and down arrows) brings up the preset manager.

### **13.5 (94) A - B Morphing**

Gradually morphes parameters from A to B slots or vice versa.

The parameter set associated with the current morphing slider position can be saved as a preset. In addition, when the morphing slider is in an intermediary position, any edit made to a parameter switches the slider back to slot A or B, whichever is closest to the current position.

### **13.6 (95) Automation**

Enabling the Automation control switch makes the morphing slider exposed and available for automation read.

When engaged, keep in mind only the morphing slider value is used for automation, and other parameter values are ignored. This behavior is intended and necessary to prevent any parameter conflicts that would otherwise occur.

As a consequence of this, you need to make sure the Automation switch is engaged when mapping the morphing slider mapped to a control surface hardware knob or slider. On the opposite, when not engaged, the plug-in will listen for any parameter automation, except the morphing slider.

# 14 Presets Management

## 14.1 From the Plug-in interface

### 14.1.1 A-B Sections

A plug-in features two preset sections : A & B. Clicking on the slot of a specific section reaches the shared preset bank.

From the preset management window you can select the preset you want to recall in the specific preset section.

### 14.1.2 Save

Save replaces the selected preset by a new one under the same name featuring the current settings. If you want to keep an existing preset without your new modifications, just select an empty place into the preset list, enter a new name for this modified preset featuring the current settings and press Save.

### 14.1.3 Recall

Once a preset is selected from the preset list it must be explicitly loaded into the section A or the section B by using the recall button. A preset is effective only after it has been recalled. Double-clicking on the preset name from the list, reloads the preset into the selected slot.

### 14.1.4 AB Slider

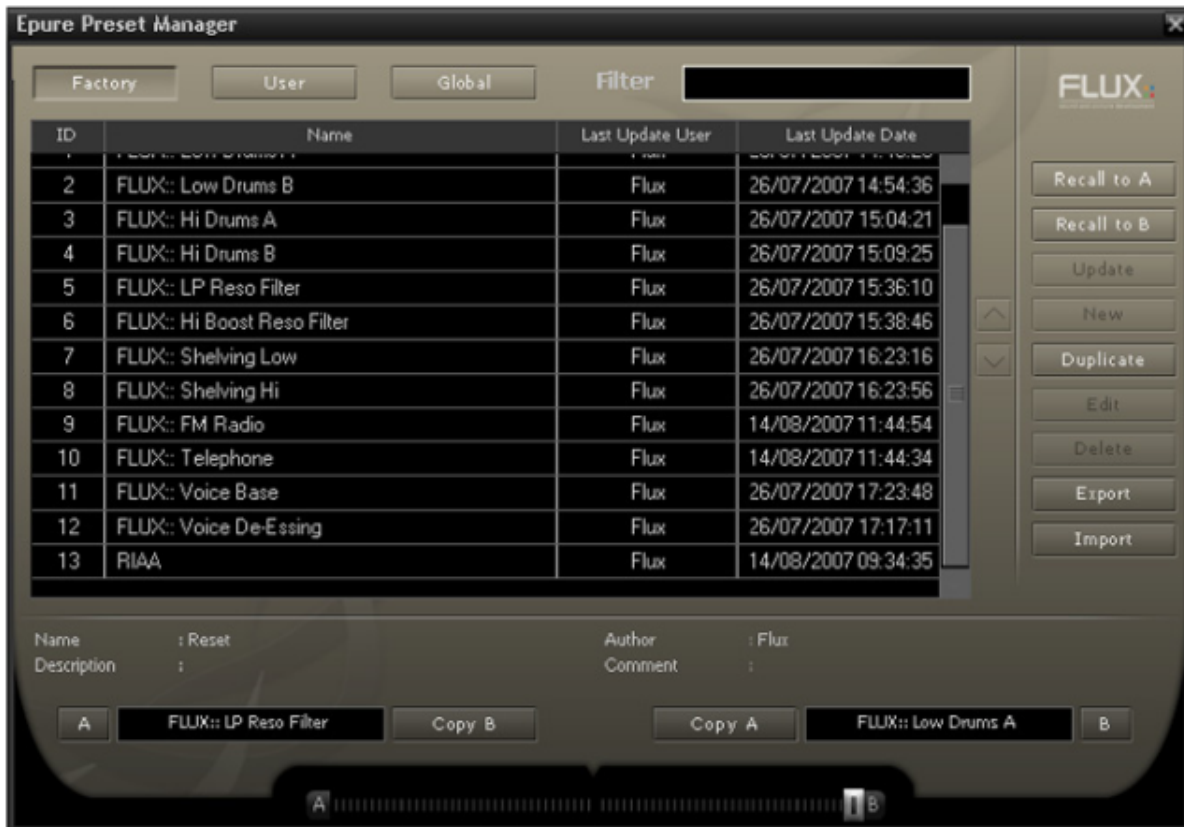
This horizontal slider has no unity nor specific value display. It allows to morph current settings between two loaded presets. A double-click on one side of the slider area toggles between full A and full B settings. The results of an in between setting can be save as a new preset.

## 14.2 From the Preset Management Window

The Presets Management Window features three preset banks:

- The Factory bank gathers presets that can't be edited by users.
- The User bank is dedicated to the users presets.
- The Global bank features presets for A, B and morphing sections. A single global preset includes A and B section content and the morphing slider position.

A Preset can directly be recalled into the preset section selected by the morphing slider position, by double-clicking on its name on the list. The preset lists can be filtered. This filter is applied to any preset information such as name, description, author, comments or key words.



### 14.2.1 Recall A

recalls the selected preset into the corresponding section.

### **14.2.2 Recall B**

recalls the selected preset into the corresponding section.

### **14.2.3 Copy A and Copy B**

buttons allow to easily create a variation around a preset.

### **14.2.4 Update**

allows to save the current settings for the selected preset.

### **14.2.5 New**

creates a new preset in the list.

### **14.2.6 Duplicate**

creates a new preset in the list from the selected one.

### **14.2.7 Edit**

gives access to the specific windows which allows to change preset name, description, key words.....

### **14.2.8 Delete**

suppresses the selected preset.

### **14.2.9 Export**

creates a file reflecting the content of the preset bank.

### **14.2.10 Import**

adds existing presets into the preset bank.



### 14.2.11 Ordering arrows

orders the presets into the list.



Please Provide Preset Informations

Name  
My Old Compressor

Description  
4 ratio

Comment  
Bass

Keyword  
Electric bass, Kick

☒ Locked

Ok Cancel

The preset protection if engaged, allows only its original modification author to uncheck and edit. So you can protect your presets in a multi-user configuration. Protected presets can only be modified using the session of their creator. If used in another user session they can only be imported or deleted.

# 15 Credits

SuperVP is a registered trademark of Ircam.

Design and C++ implementation of signal analysis and signal transformation algorithms (SuperVP): Axel Röbel.

Design and C++ implementation of voice transformation algorithms (VoiceForger): Snorre Farner, Xavier Rodet and Axel Röbel.

Thanks for additional contributions, instructive discussions and advice to Frederic Cornu, Chunghsin Yeh, Alain Lithaud, Niels Bogaards, Norbert Schnell, Fernando Villavicencio Marquez, Greg Beller, and Joshua Fineberg, Gael Martinet.

Ircam R&D Director : Hugues Vinet  
Collection Manager: Frederick Rousseau  
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## IRCAMTOOLS

Collection Manager for IRCAM: Frederick Rousseau  
Collection Manager for Flux:: Gael Martinet

## IRCAMTOOLS TRAX, TRAX CS, TRAX SF

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## Flux::

Head software engineering: Gael Martinet  
Developeppers: Gael Martinet, Samuel Tracol, Siegfried Hand  
Designer: Nicolas Philippot  
Contributions: Lorcan Mcdonagh

# 16 Specifications

## 16.1 Availability

IRCAM Trax is available in:

AU / VST / VST3 / AAX Native/ *AAX AudioSuite*

*\* AAX Native & AAX AudioSuite in Pro Tools 11 and later*

## 16.2 Processing

EVO Touch provides :

- Up to 2 channels Input/Output in VST/VST3/AU/AAX.
- 64-bits internal floating point processing.
- Sampling rate up to 384 kHz.

## 16.3 Hardware Requirements

A graphic card fully supporting OpenGL 2.0 is required.

- macOS : OpenGL 2.0 required – Mac Pro 1.1 & Mac Pro 2.1 are not supported.
- Windows : If your computer has an ATi or NVidia graphics card, please assure the latest graphic drivers from the ATi or NVidia website are installed.

## 16.4 Software License Requirements

In order to use the software an iLok.com user account is required (the iLok USB Smart Key is not required).

## 16.5 Compatibility

All major native formats are supported

### 16.5.1 Windows – 10, in 64 bits only.

- VST (2.4)
- VST3 (3.1)
- AAX Native\*
- AAX AudioSuite\*

### 16.5.2 macOS (Intel and ARM)

All versions from Sierra (10.12) to latest. (Compatible with previous versions but not supported)

- VST (2.4)
- VST3 (3.1)
- AU
- AAX Native\*
- AAX AudioSuite\*

\* *AAX Native & AAX AudioSuite in Pro Tools 11 and later*