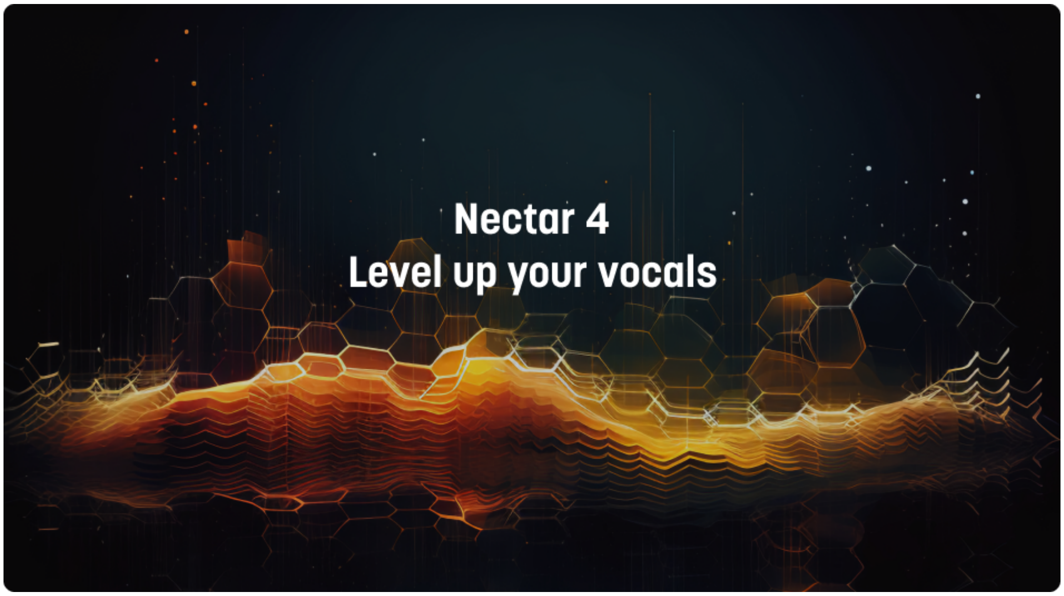
Nectar 4 Help

Introduction



Achieve professional-quality vocals effortlessly with iZotope Nectar 4. This comprehensive vocal mixing tool provides a complete set of features for mixing, producing, and designing vocals.

HELP IS HERE

Vocal Assistant gets you a finished vocal sound instantly. Now with the new Vocal Assistant page you can broadly adjust aspects of your vocal or dive deep into the module chain for endless tweaking.

1. Vocal Assistant Page: A simple and powerful page for making broad decisions about your vocals.
2. Unmask: Puts your vocal at the forefront of your mix by carving away other competing mix elements directly from Vocal Assistant's menu screen.

IN SWEET HARMONY…

Nectar 4 now comes with the most simple and powerful Voices module, giving you a new way to build stacks of complex harmonies in your vocal productions fast.

INSTANTLY CONSISTENT

The brand new Auto-level module is rebuilt from the ground up to get your musical vocals or spoken word content instantly consistent in your production. Tame the vocal noise that gets in the way of your finished musical vocal sound with the new Tame Noise feature or ride the level of an instrumental with Sidechain Mode enabled, getting your vocal to sit perfectly in the mix.

WHO'S BACKING YOU TODAY?

With iZotope's singing voice conversion technology you can now create artificial background singers that sit under your main vocal with the Backer Module. Finish up demos with different flavors of vocalists, creatively destroy ad-lib tracks, or create something new altogether. Import your own acapella to further customize your sound.

REFERENCE ANY VOCAL

The popular Audiolens application is fully compatible with Nectar 4, giving you the ability to tone match the sound of any vocal. Audiolens magically separates the vocal from your reference track to provide Nectar the best possible information to create a starting point for you in the Assistant.

BUILD YOUR OWN CHAIN WITH NEW COMPONENT PLUGINS

Nectar 4 Advanced comes with a powerful suite of 13 individual plug-ins. Integrate the power of Nectar's system into your DAW template to build your own custom chains for every vocal in your production. *VST3, AUv2, AAX only

A MATCH MADE IN HEAVEN: Celemony Melodyne 5 essential

As an added benefit to Nectar 4's built-in Pitch correction, iZotope has partnered with Celemony to include Melodyne 5 essential: GRAMMY-award winning pitch and time correction. Melodyne adds surgical, intuitive, and truly transparent pitch adjustment capabilities that are an industry standard in professional studios around the world.

IMPROVED, REVAMPED MODULES AND REFRESHED USER INTERFACE

1. **Improve Vocal Assistant Page**: Instantly create custom presets using machine learning technology to optimize your vocal's fit in the mix.
2. **New Backer Module**: With iZotope's Backer module, you can now create artificial

background singers that sit behind your main vocal.

3. **New Voices Module**: Instant, easy vocal layers: the Voices module gives you the ability to create complicated layers for your vocal production without needing to learn voice-leading and harmonic motion.
4. **New Auto Level Module**: Automatically adjusts vocal levels before processing, eliminating the need for corrective dynamics or manual fader adjustments.
5. **New Component Plug-ins**: Comprehensive set of 13 vocal-focused production and mixing tools, including: EQ, Auto-Level, Breath Control, Backer, Pitch, Compressor, Voices, De-Esser, Saturation, Reverb, Gate, Delay, and Dimension.
6. **Improved Unmask**: Easily prioritize your vocal by reducing competing mix elements directly from the Vocal Assistant's menu screen. Dynamic EQ with Follow EQ Mode: Real-time tracking and removal of troublesome resonances, saving time and effort.
7. **Inter-plugin Communication**: Nectar 4 Mothership communicates with other iZotope plug-ins in your session to address tonal balance and masking.

Extras:

1. **Nectar 4 Breath Control plug-in**: Reduces unwanted breaths without compromising audio quality.
2. **Melodyne 5 Essential**: Award-winning time and pitch correction software.
3. **iZotope Relay**: Central hub for intelligent connection between your entire mix and iZotope plug-in collection.
4. **Hundreds of Professionally-designed presets**: Preconfigured settings for music and dialogue applications.

Vocal Assistant

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Vocal Assistant Workflow

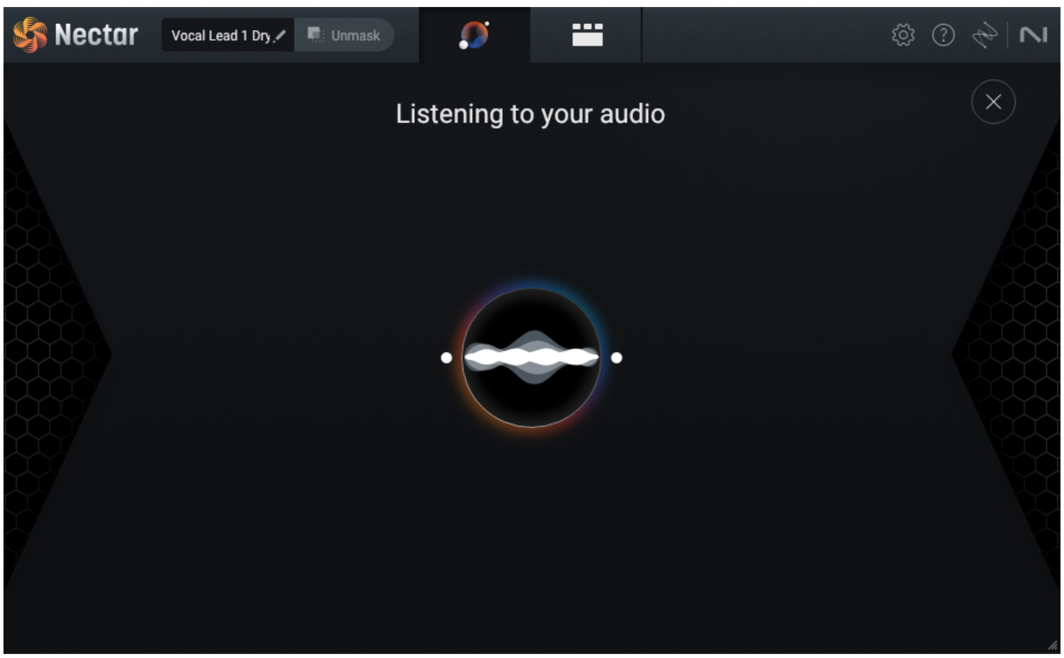
Vocal Assistant is available in the Nectar 4 mothership plug-in. Select the Assistant button in the header of the plug-in to open the Assistant.

NOTE

1. **Vocal Assistant needs audio input**: Make sure you are playing audio in your DAW.
2. **Vocal Assistant needs time to analyze your vocal**. If you are trying to run Vocal Assistant on a clip shorter than 20 seconds, you should enable loop playback before starting Vocal Assistant.

Assistant Analysis

An analysis pass will run first.



Vocal Assistant runs through the following steps during analysis:

1. **Analyzing input level to determine optimal RMS**: Enables Auto-Level module (Advanced only) and adjusts the input gain to ensure an optimal input level.
2. **Analyzing vocal signal to determine optimal settings**: Analyzes your vocal to identify characteristics that will inform the processing applied in the subsequent steps.
3. **Applying settings based on vocal content**: Applies an initial “character” **EQ** curve and Dynamics settings.
4. **Detecting vocal register**: Analyzes pitch information in your vocal track and uses that information to set the **Vocal Register** for optimal Voices and Backer settings.
5. **Learning subtractive EQ parameters to improve clarity**: Learns the frequency content of your vocal and sets subtractive **EQ** bands to frequency values that will

help to improve clarity.

6. **Detecting vocal sibilance to set de-esser**: Analyzes the frequency content of your vocal for harsh, sibilant frequency content. If sibilance is detected, Vocal Assistant will set the optimal cutoff frequencies and threshold value for **De-esser**.
7. **Applying dynamics for a controlled output level**: Configures **Compressor** settings to control the output level of your vocal.
8. **Setting reverb mix level depending on selected assistant mode**: Applies a subtle **Reverb** to your vocal to add a sense of space.

Vocal Assistant view interface

The following image outlines the key sections of the Vocal Assistant view:

1. Intent Controls
2. Target Library
3. Meters and Display
4. Key and Range

Intent Controls

Intent controls are parameters that allow for broad control over early decisions in your vocal production & mixing workflow. They work together with Vocal Assistant's intelligent analysis, Targets and the Signal Chain to give you a simple workflow for tonal, dynamics, fx, width, and voice decisions in your mix. Intent Controls are exclusive to Vocal Assistant view in the mothership plug-in and are not available in the Nectar component plug-ins.

The following control sections are available:

1. Shape
2. Intensity
3. FX
4. Voices
5. Width

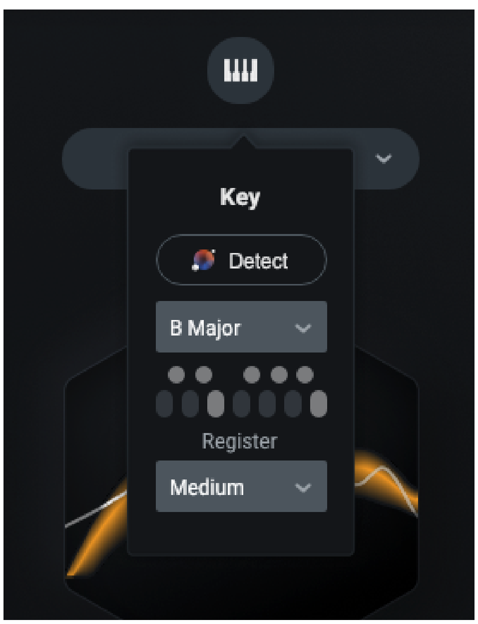
Intent Controls are linked to deeper controls in the Detailed View of the mothership plug-in. Here is what each intent control is mapped to in the Detailed View:

Intent Controls section: Control name	Mapped to **Module Name**: Control name
Shape: Amount	**EQ 2**: Frequency Gain
Shape: Power Button	**EQ 2**: Module Bypass
Intensity: Amount	**Compressor 1**: Threshold
Intensity: Amount	**Compressor 1**: Ratio
Intensity: Amount	**Compressor 1**: Attack
Intensity: Amount	**Compressor 1**: Release
Intensity: Power Button	**Compressor 1**: Module Bypass
Width: Width Amount	**Global**: Width
FX: Y	**Reverb**: Wet Mix
FX: X	**Reverb**: Width
FX: X	**Reverb**: Predelay
FX: X	**Reverb**: Decay
FX: Power Button	**Reverb**: Module Bypass
FX: X	**Delay**: Rate
FX: X	**Delay**: Depth
FX: X	**Delay**: Feedback L
FX: X	**Delay**: Feedback R
FX: X	**Delay**: Amount
FX: Y	**Delay**: Wet Mix
FX: Power Button	**Delay**: Module Bypass
FX: X	**Dimension**: Depth
FX: X	**Dimension**: Feedback
FX: X	**Dimension**: Width

Intent Controls section: Control name	Mapped to **Module Name**: Control name
FX: Y	**Dimension**: Wet Mix
FX: Power Button	**Dimension**: Module Bypass
Voices: Amount	**Voices**: Wet Mix
Voices: Left and Right Arrows	**Voices**: Styles
Voices: Power Button	**Voices**: Module Bypass
Backer: Amount	**Backer**: Wet Mix
Backer: Left and Right Arrows	**Backer**: Targets
Backer: Power Button	**Backer**: Module Bypass

Key and Range

You can modify the Key and Vocal Range settings by clicking the following icon.

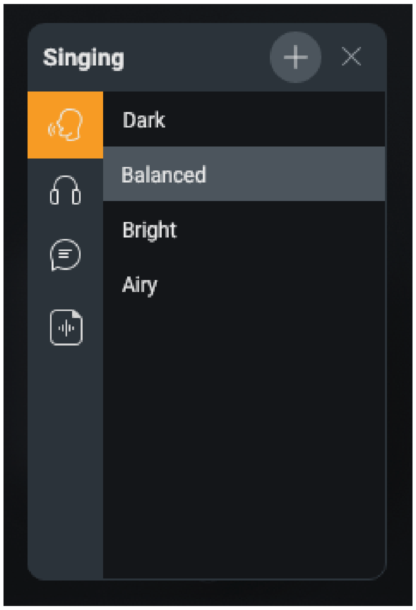


NOTE

Only Octaves and Unison settings will be available in Voices until a Key is set.

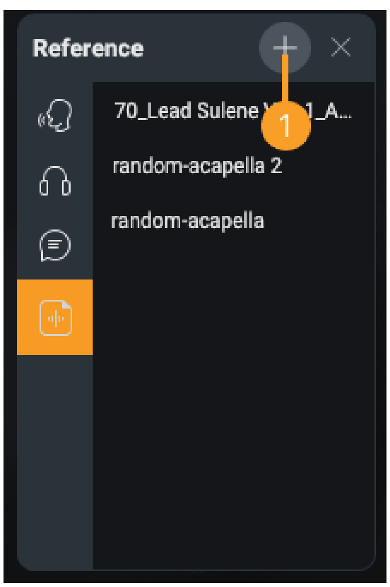
Target Library

The Target Library contains vocal targets and user generated reference targets. Setting a target provides helpful starting points for the Intent controls in the Vocal Assistant View and the modules in signal chain in the Detailed View.



By clicking the “+” button you can upload your own reference target to the Target Library to guide the Character EQ of Nectar 4.

Target Curves saved by the Audiolens application will appear here.

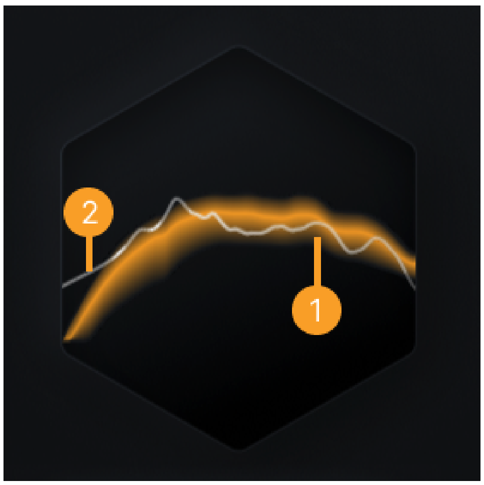


Meter

The Tonal Balance Meter in the center of the Vocal Assistant shows the input signal against the target curve. Adjusting the Shape control will push your input signal closer to the target curve. Custom target curves will appear in this meter when added to the Target Library.

The following image outlines the metering information:

1. Input Signal
2. Tonal Balance Target Curve



Global Header Controls

To re-run Vocal Assistant processing before applying settings, click the **Relearn** button.

To disable Nectar's processing select the **Bypass** button.

To exit the Vocal Assistant screen, click the Detailed View button. Here you can make detailed adjustments to modules in Nectar 4.

Auto-Level

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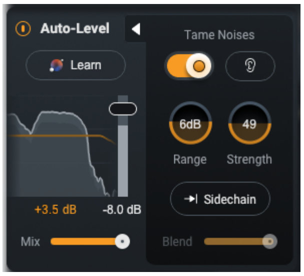
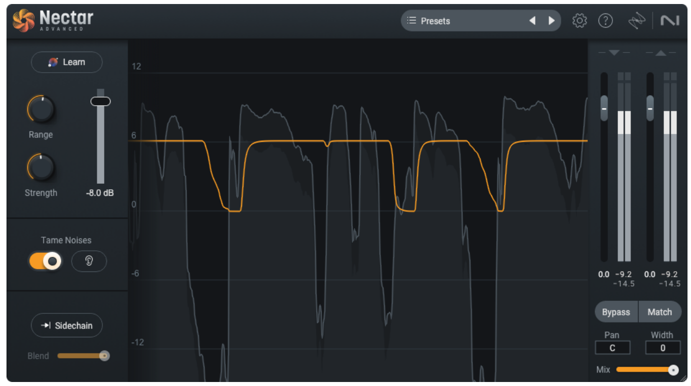
Overview

ALM (Auto Level Module) is an intelligent and transparent alternative to compressors. Placed at the start of your signal chain, it acts like a recording engineer, effortlessly managing vocal levels. ALM's Tame Noise feature distinguishes between sung and unsung vocal content, ensuring only tonal elements are leveled. This gets rid of the need for manual volume automation, saving you time and effort. ALM provides consistent volume levels without introducing unwanted artifacts like traditional compressors, giving you a clean and natural sound.

Auto-Level is available as a mini-module in the mothership plug-in and as a component plug-in with Nectar 4 Advanced. The layout differs slightly between the mini-module and the component plug-in, two images are included throughout this chapter where the layouts differ.

mini-module	component plug-in
	

Controls

Learn

Clicking the Learn button will enable a short analysis pass so that Auto-Level module can set an optimum RMS target.

mini-module	Component plug-in
	

Target

The Target level can be set by adjusting the vertical handle in the Auto-Level module. The input signal will constantly adjust its level to hit the desired decibel level.

mini-module	Component plug-in
	

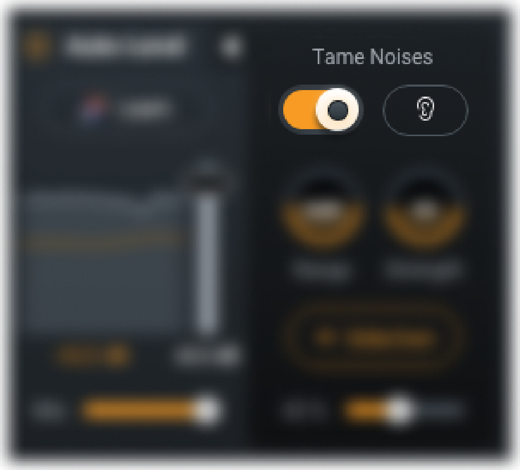
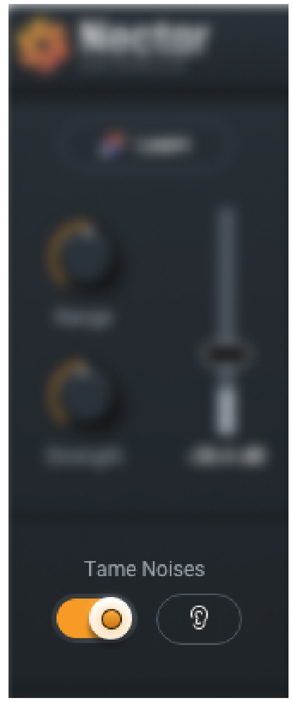
Mix

Adjusts the amount of leveling applied to the input signal. A setting of a 100% will apply leveling the entire input signal. A setting of 0% will result in no leveling applied to the input signal.

mini-module	Component plug-in
	

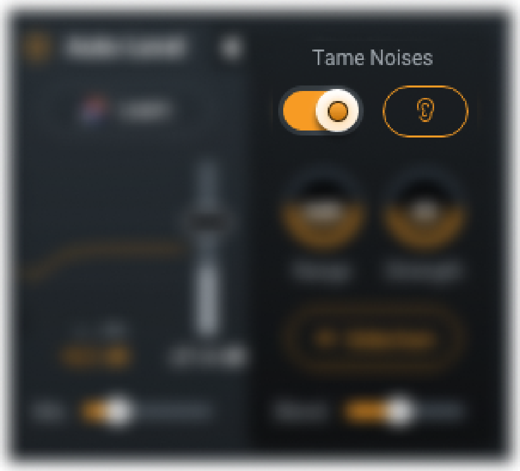
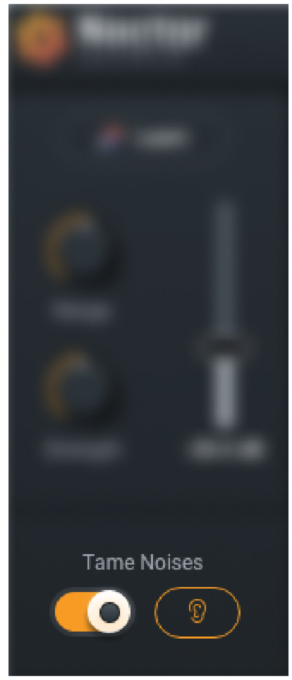
Tame Noise

Tame Noise is an intelligent setting in the Auto-Level Module that can decipher between the sung material and noise introduced by the vocalist’s performance. By enabling Tame Noise, any non-sung or “noisy” content will be ignored by the leveling applied to the input signal, making it less pronounced in the final leveled vocal. Examples of this are sounds from the throat or mouth that a singer may introduce into their signal as they ramping in or out of a vocal passage. Most professional engineers must manually manage this type of material with clip gain or automation. This mechanism replaces the need for that manual process.

mini-module	Component plug-in
	

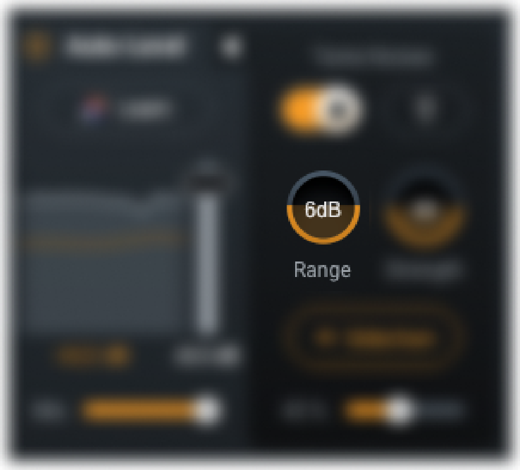
Listen

Enable this to hear the atonal information Tame Noise is detecting in the input signal.

mini-module	Component plug-in
	

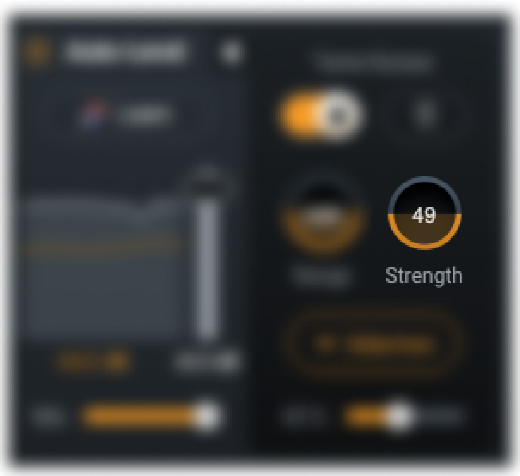
Range

Adjusts the total allowed leveling of the input signal. A setting of 6 decibels will limit the amount of leveling to a maximum of +6 or -6 dB in leveling. A setting of 12 will result in extreme leveling applied to the input signal. A setting of 1 will result in subtle leveling applied to the input signal.

mini-module	Component plug-in
	

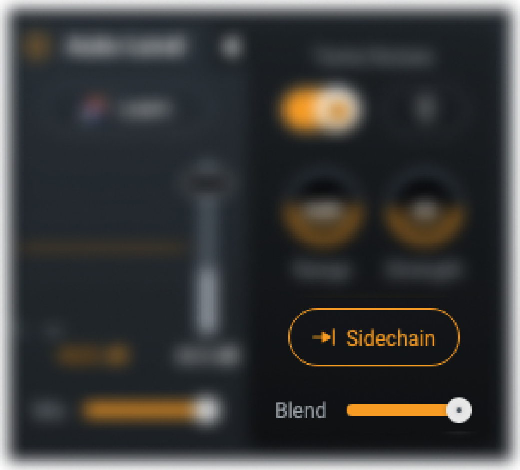
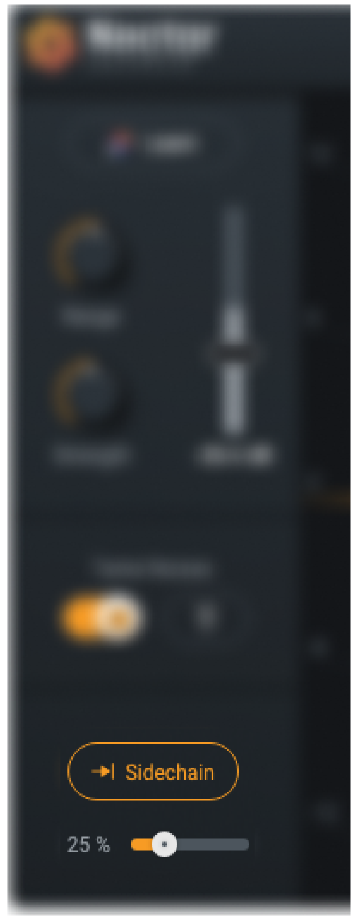
Strength

Adjusts the sensitivity of the leveling applied to the input signal. Higher settings will results in faster and more dramatic leveling applied to the input signal. Lower settings will result in slower and subtle leveling applied to the input signal. A setting of zero will result in no leveling applied to the input signal.

mini-module	Component plug-in
	

Side-chain Mode

The side-chain mode in the Auto-Level Module helps you fit your vocal into a dynamic arrangement in the even that a single consistent vocal level is not the ideal outcome of your mix. By leveraging the side-chain input, you can choose to have your vocal follow the level of an external source.

mini-module	Component plug-in
	

Enable

Enabling Side-chain Mode will cause the Auto-Level Module to apply leveling to the input signal using the side-chain input of Nectar 4 as a guide.

Blend

Mixes the RMS Levels of the side-chain input and the constant ALM target level. A setting of 100% will apply the RMS level of the side-chain to the entire input signal. A setting of 0% will blend the leveling applied by the target level only. **Note** you must have side-chain routing setup in your DAW to use this feature accurately.

Bypass

Select this to bypass Auto-Level processing.

Meters

The following meters are included in the Auto-Level module:

1. **Waveform Displays**
2. **Gain Reduction Trace**

mini-module	Component plug-in
Meters in mothership	Meters in component

Waveform Displays

The scrolling waveform meters display the amplitude of the input (non-leveled) and output (leveled) signals over time. The meters scroll from right to left, with the most recent information on the right.

The leveled output signal waveform is displayed in light grey, in front of the input signal waveform. The uncompressed input signal waveform is displayed in dark grey, behind the output signal waveform. When the signal is being leveled, the gain reduction applied to the output signal can be observed in the difference between the two waveforms.

Gain Reduction Trace

The yellow trace line indicates the gain reduction applied by the leveling over time. The trace can be used to monitor the strength, range, and atonal information if listen is enabled over time.

Voices

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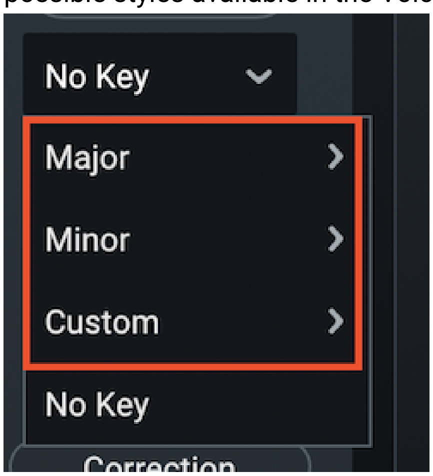
Overview

The Voices module can be used to build vocal harmonies by creating pitch-shifted copies of a vocal track. The voices module has 3 ways control harmonies in your music. You can use the Voices matrix to preview different Styles of harmonies, mix your voices using the Visual Mixer view, or put the Voices module into MIDI mode and perform your harmonies in realtime.

ⓘ SELECT A KEY TO EXPLORE ALL STYLES

You must select a key for your instance of Nectar 4 in order to explore all the possible styles available in the Voices module.



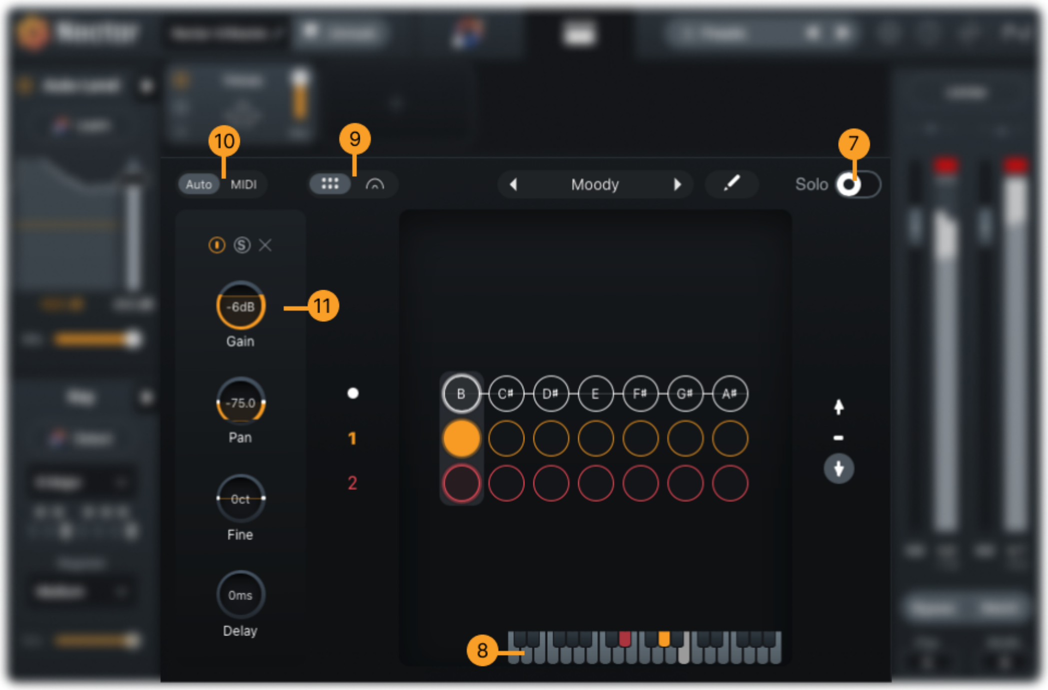
When "No Key" is selected, only Octave and Unison styles will be available.

Controls

The Voices module includes controls for adjusting processing applied to all voices, individual voices, a rich editing matrix for voices & chords, as well as an X/Y pad interface for mixing voices.

These are the notable areas in the Voices module:

1. **Style Picker** & **Custom Styles**
2. **Direction**
3. **Filter**
4. **Pitch Correction**
5. **Time Variation**
6. **Pitch Variation**



7. Solo
8. **Piano Inspector View**
9. **Visual Mixer View**
10. **MIDI Mode Toggle**
11. **Per Voice Controls**

Style Picker

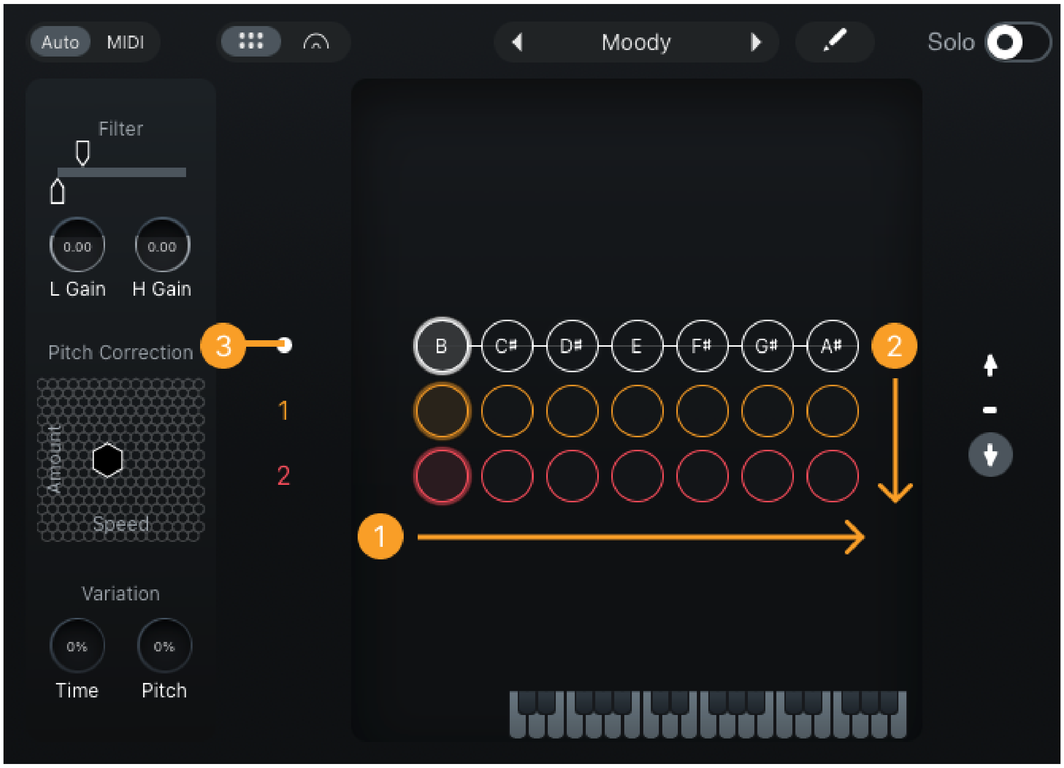
The Style Picker gives you the ability to quickly audition different styles of harmonies the Voices module uses to harmonize your vocal. The 11 different styles have set rules for harmonic motion.

If a style doesn't quite fit your exact use case you can edit the Style by selecting the "(pencil)" button and enter the Custom Style editing mode to modify each style. Below outlines the expected rules for the default settings in each automatic harmony.

View

The Auto-Harmony view includes automatic harmony settings for creating complex vocal harmonies in your music without needing to know music theory or harmonic motion. This features a rich editor that gives you the ability to define the chords and scales the Voices module uses to create harmonies. The layout is arranged in a matrix.

1. Voices (horizontal rows)
2. Chords (vertical columns)
3. Main Vocal (unlabeled row)



Individual voices are arranged in horizontal rows and the chords those voices sing together with are laid out in vertical columns. The unlabeled white Voicing is your input vocal. Voices that are arranged above your input vocal will sing in a register above your input signal and voices that are arranged below your input vocal will sing below your input signal.

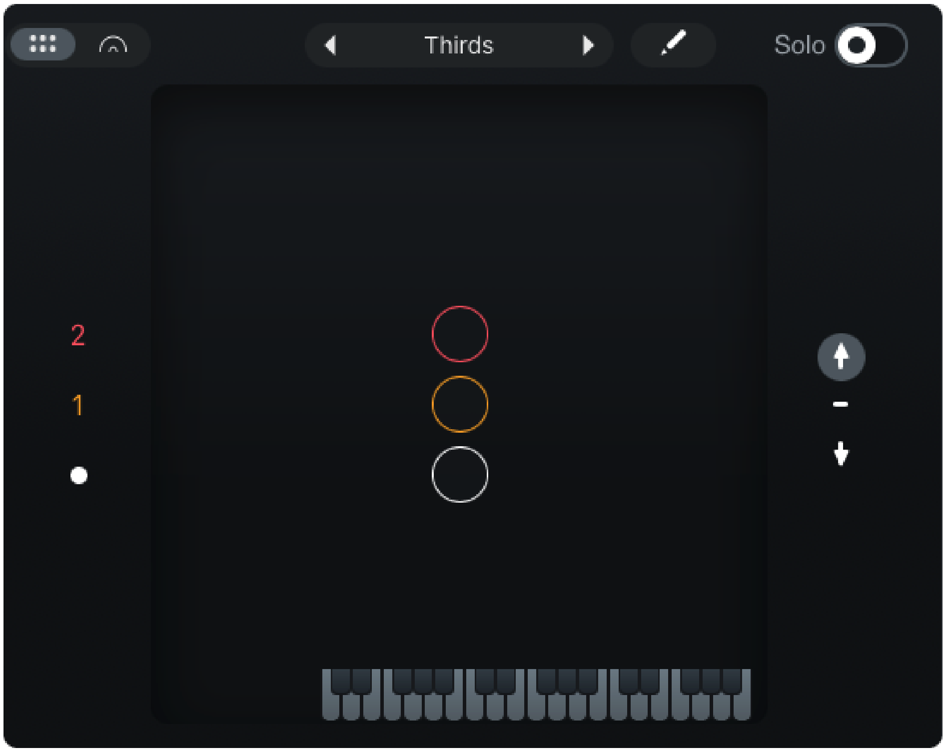
Interval Styles

There are two types of auto-harmonies available in the Style picker:

1. **Static Interval Styles**: Styles that create a static harmony interval. Static Interval Styles: Unison, Octaves, Thirds, Parallel.
2. **Automatic Interval Styles**: Styles that create a dynamic harmony based on the detected scale degree in your main vocal. Auto Interval Styles: Moody, Uplifting, Lush, Close, Closer, Happy, Sad.

Static Interval Styles

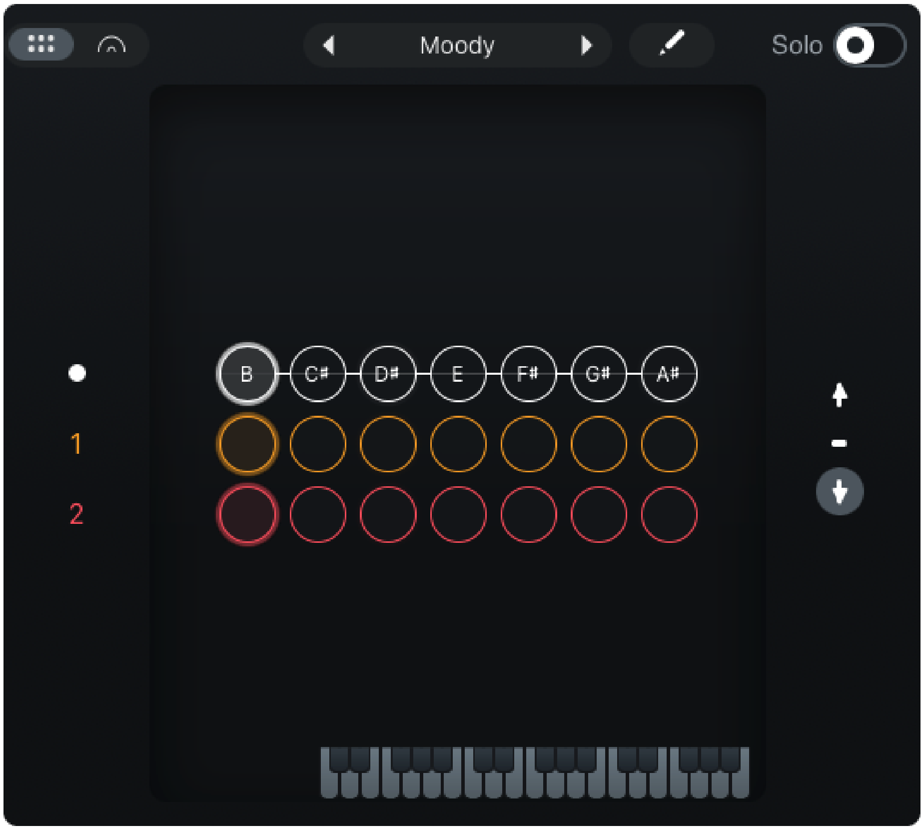
These styles will show only one column since these set a single static voicing above or below your input signal.



Style Name	Description
Unison	Creates doubles of the lead note.
Octaves	Creates voices that are octaves apart from the lead note.
Thirds	Creates harmonies that are (diatonically) either thirds above or sixths below the lead note.
Parallel	Creates harmonies using parallel triads.

Automatic Styles

Automatic Styles display several columns to allow for every note in the selected scale to have a unique note assignment for Western music styles.


Moody
Solo
B
C#
D#
E
F#
G#
A#
1
2

Style Name	Description
Moody	Creates harmonies primarily using the VII and i triads of the (relative) minor scale. When harmonizing the (flat) 6th scale degree of the (relative) minor scale as the lead note, the iv triad is used.
Uplifting	Creates harmonies primarily using the I and ii triads of the (relative) major scale. When harmonizing the 7th scale degree of the (relative) major scale as the lead note, the V triad is used.
Lush	Creates harmonies using the i7 and iio7 seventh chords of the (relative) minor scale. In this case, the use of the iio7 chord introduces the (non-diatonic) raised 7th scale degree/“leading tone” into the harmonies. This style is based on the (major) “sixth-diminished” scale popularized by the late jazz pedagogue Barry Harris.
Close	Creates harmonies using notes from the pentatonic scale resulting in closed voicings, particularly when the lead assumes the top voice.
Closer	Creates harmonies using notes from the hexatonic scale resulting in (even tighter) closed voicings, particularly when the lead assumes the top voice.
Happy	Creates harmonies using the I, IV, and V triads of the (relative) major scale. When harmonizing either the 1st or 5th scale degrees as lead notes, the I triad is used.
Sad	Creates harmonies primarily using the i, iv, and V triads of the (relative) minor scale. In this case, the use of the V chord introduces the (non-diatonic) raised 7th scale degree/“leading tone” into the harmonies. When harmonizing the (flat) 7th scale degree of the (relative) minor scale as the lead note, the v triad is used.

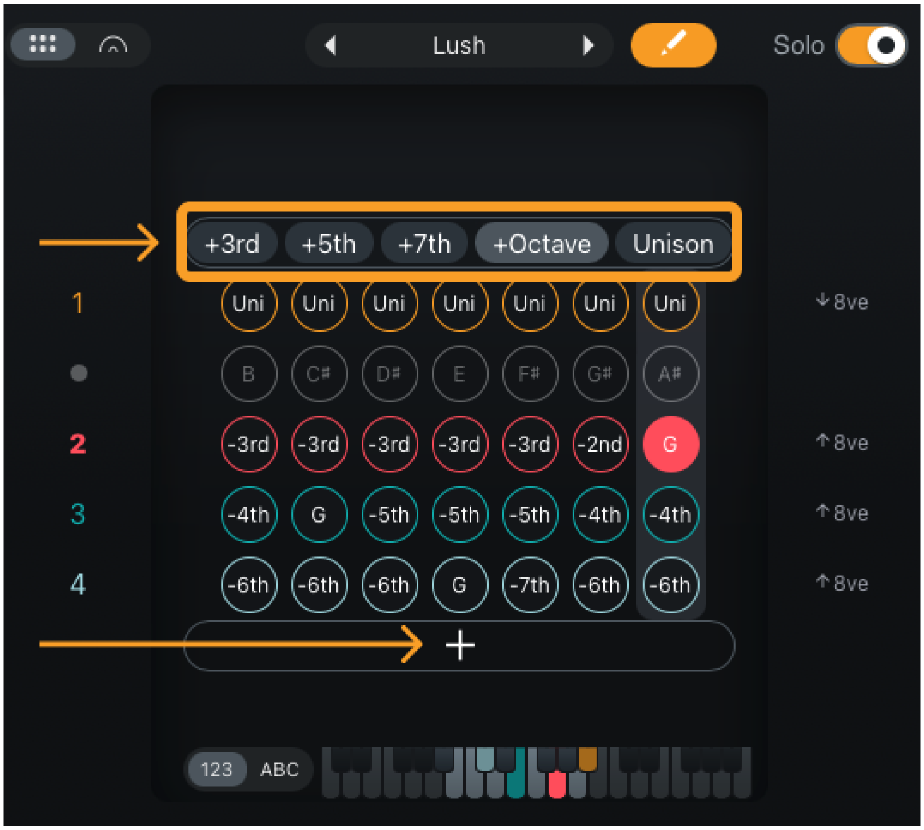
Custom Styles

You can save your own Style using the Custom Style editing mode. Click on the pencil button to the right of the style picker menu to edit your Custom Style.

The Custom Style editing mode has many features available for modifying how the Voices module harmonizes your input signal.

Quick Harmonies

Two large “+” buttons are available below and above the Custom Style editing mode. On hover you will be presented with recommended harmonies to quickly add to your Custom Style. Selecting one from below will add that voicing below the main voice, selecting one from above will add that voicing above the main voice. Selecting a Quick Harmony will create a new voice with every scale degree set to the Quick Harmony you selected.



Modifying Individual Notes

Select a note in Custom Style editor mode to modify it. A menu will appear that allows you to select any tonal or atonal notes available in the selected key.

Direction

The Direction buttons are a quick way to adjust the register of each Voice relative to the main vocal. Direction is available in the Style View and Custom Style Mode. Click the following buttons to make adjustments to Direction:

1. **General Direction Controls**: Adjusts the voices "(arrow up)" above, "(arrow down)" below or around "(-)" the main vocal.
2. **Per Voice Octave Shift**: Adjusts the octave shift of a specific voice in the Custom Style editor.

Filter

The small filter display on the left side of the Voices module panel includes low and high shelf filter controls for shaping the output of the Voices module.

> **NOTE**
>
> The post filters are applied to the wet (processed) output signal of the Voices module. To adjust the shelf filter values:
>
> 1. Adjust the L and H Gain Controls to modify gain.
> 2. Drag Filter handles left and right to modify frequency cutoffs.

Pitch Correction

Adjusts the speed and amount of pitch correction applied to the harmonies generated by the Voices module.

Variation: Time

Scales the amount of time offset applied to the generated voices. Increasing this control can help to reduce phase cancellation in the generated voices, as well as increase the chorus-type effect of multiple performers singing the same part.

Variation: Pitch

Scales the amount of pitch offset applied to the generated voices. Increasing this control can help to humanize the character of the harmonies by varying the pitch of each voice slightly. It can also be used to reduce phase cancellation in the generated voices.

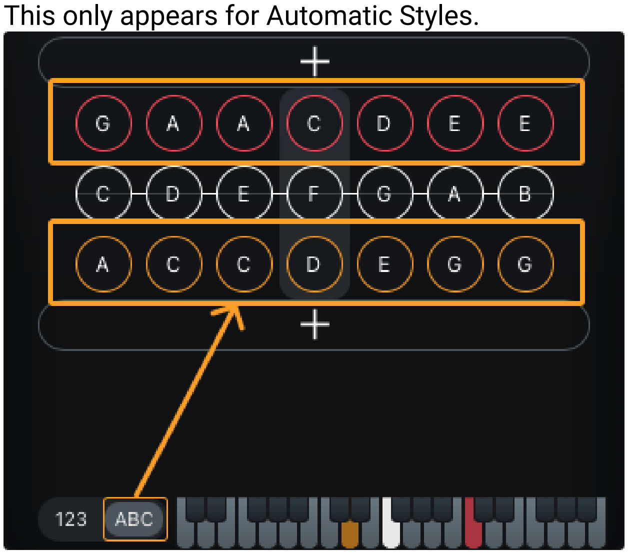
Piano Inspector View

The Piano at the bottom of the Voices module gives you a color-coded visual layout so you can see what notes, chords, and key you're working on at any given moment. In order to see notes laid out on the keyboard select one of the notes in the Voices view. This will show you all notes in the chord in the key you have set for Nectar 4. Selecting different notes in a chord will highlight that note's associated color in the Piano Inspector.

In the above figure, the seventh scale degree is selected in the key of B Major. The notes in the selected column are laid out in the Piano View with each note corresponding to its relative color.

Note View & Interval View Toggle

This toggle will present the Note information in the main editor in two different ways:

1. **To view Note names select "ABC"**: This setting will present the note information as the literal Notes in each scale degree key sung by each Voice. A comma will appear next to a note to indicate this note is an octave below the relative register. This only appears for Automatic Styles.


2. **To view Notes as intervals select "123"**: This setting will present note information as intervals relative to the main vocal.

Solo (Voices)

When enabled, the main voice will be muted and all voices created by the Voices module will be played back in isolation.

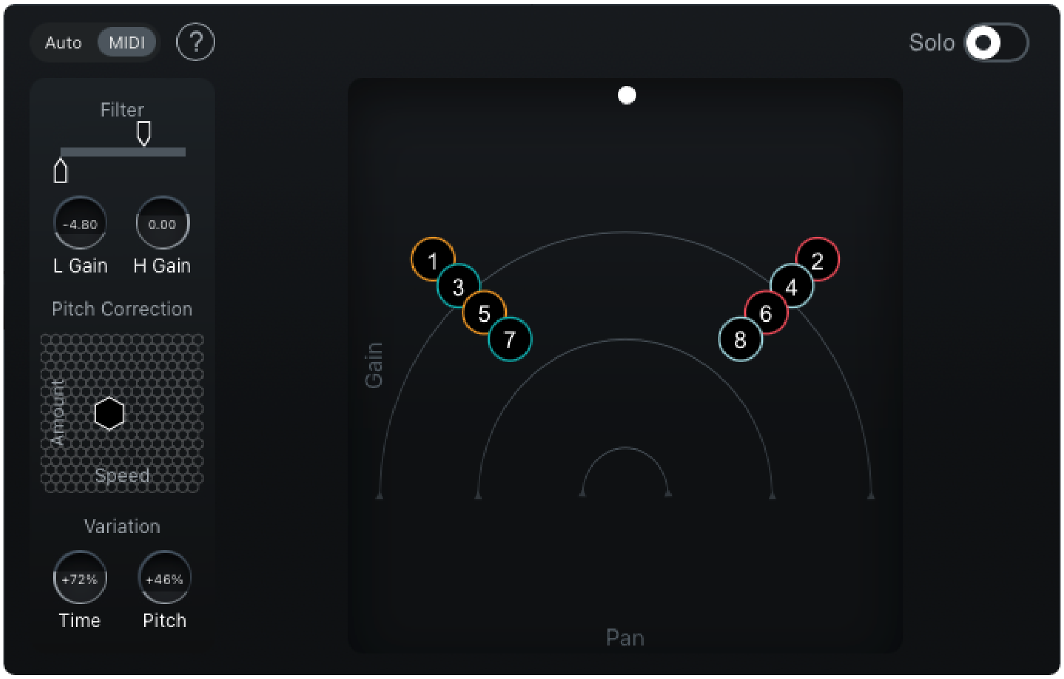
Visual Mixer View

The Voices module includes a node-based X/Y mixer interface for easily adjusting the level and pan up to eight voices.

The Mixer view in the Voices module panel allows for the adjustment and management of Voice nodes. The x-axis (horizontal) of the XY pad represents the stereo pan position of a voice. The y-axis (vertical) represents the gain (level) of a voice.

MIDI Mode

Enables the ability to control the pitches of Voices using a MIDI controller or MIDI track.



When MIDI mode is enabled, the nodes in the Voice XY Pad Controller will be highlighted when an incoming MIDI note is playing the voice. Voices are assigned to incoming MIDI notes in the order they are received. The voice node numbering determines the order in which voices are played, the lowest available voice node number will be played first.

EXAMPLE: MIDI NOTE VOICE ASSIGNMENT

For example, if four voice nodes are enabled in MIDI mode:

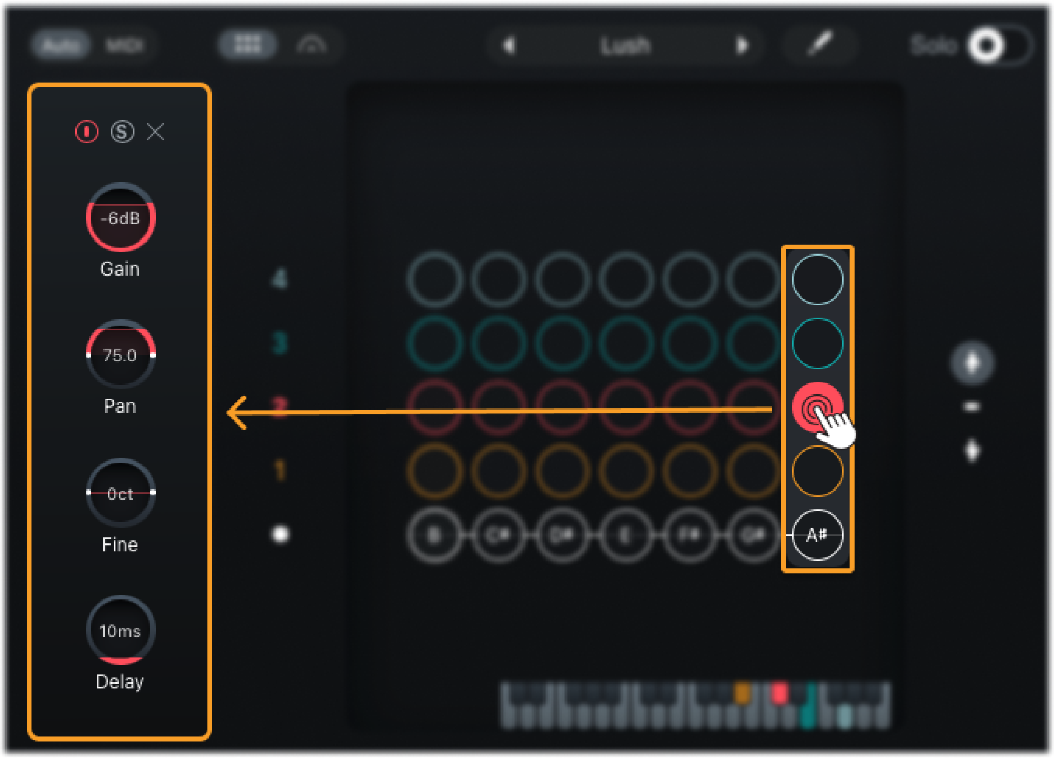
1. When a single MIDI note is played, Voice 1 will be triggered.
2. When a second MIDI note is played before the first note is released, Voice 2 will also be triggered.
3. If the first note is released and a third note is played after it, Voice 1 will be triggered again.

NEED HELP SETTING UP MIDI MODE?

Click on the ? button (to the right of the MIDI button in the Voices module) to view a setup screen with steps for configuring MIDI mode in your DAW without leaving the plug-in.

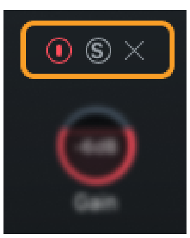
Per Voice Controls

The Note Inspector appears when a Voice is selected in the Main Voices View or Visual Mixer. Controls in this view are per-voice controls.



Header Controls

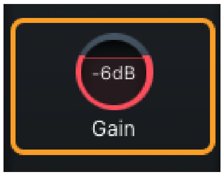
The top of the node inspector panel includes parameters for enabling, disabling and soloing the currently selected voice node.



1. **Enable**: Enables or disables processing of the currently selected voice.
2. **Solo**: Outputs the currently selected voice in isolation.
3. **Remove**: Deletes the currently selected voice.

Gain

Displays the amount of gain added to the currently selected voice. The Gain control allows for adjustments ranging from -20dB (decibels) to 0dB (decibels).



VOICE GAIN ADJUSTMENTS

The level of a selected voice node can be adjusted using the following methods:

1. Click and drag the node up (increase gain) or down (decrease gain) in the Visual Mixer view.
2. Click on the gain value readout in the voice controls panel and manually enter a value in the inline edit field.

Pan

Displays the stereo pan position of the currently selected voice. The Pan control allows for adjustments ranging from -100 (hard left) to +100 (hard right).

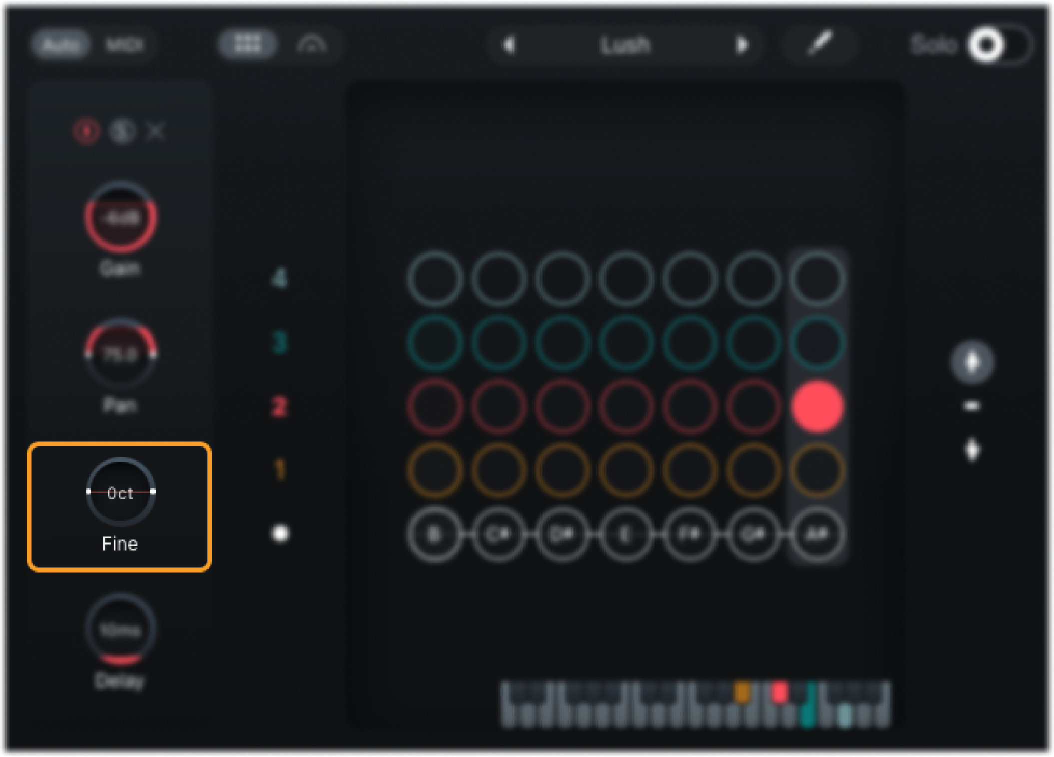
VOICE PAN ADJUSTMENTS

The Pan value of the selected voice node can be adjusted using the following methods:

1. Click and drag the node left or right in the Visual Mixer view.
2. Click on the Pan value readout in the voice controls panel and manually enter a value in the
3. inline edit field.

Fine

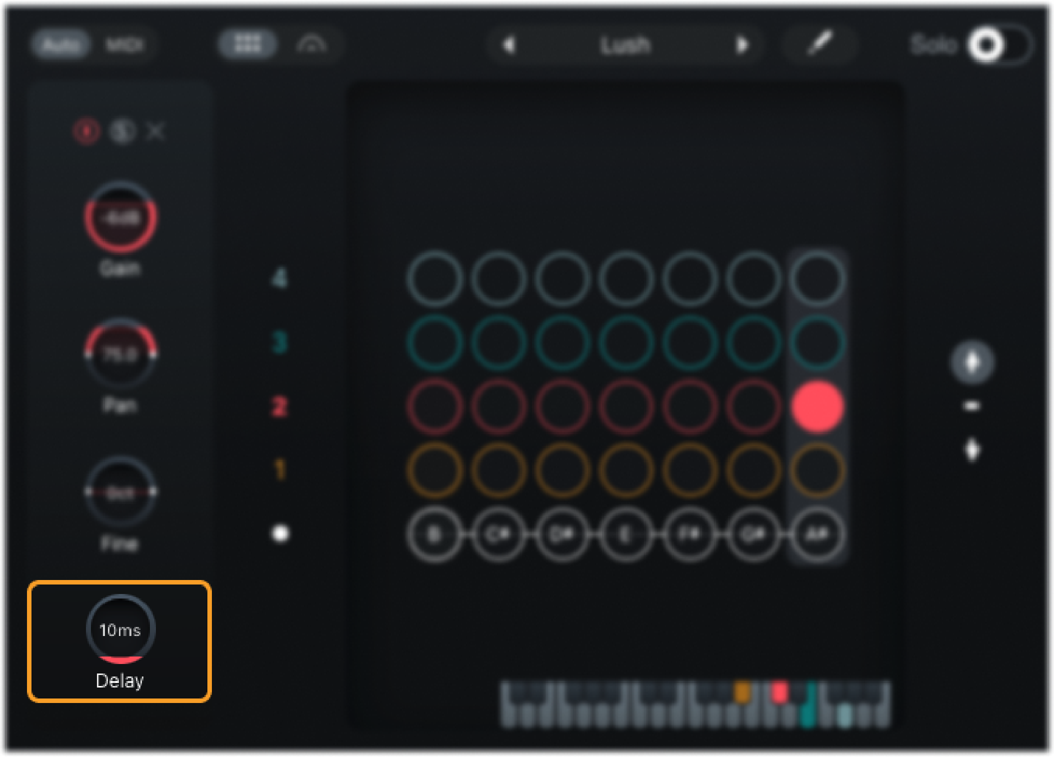
Determines the amount of fine pitch adjustment applied to the selected voice.



The Fine control allows for adjustments ranging from -100 ct (cents) to +100 (cents). 100 cents is equivalent to 1 semitone.

Delay

Determines the amount of time delay applied to the selected voice.



The Delay control allows for adjustments ranging from 0 ms (milliseconds) to 100 ms (milliseconds).

Global Module Controls

The module chain features common controls for each module, including Bypass, Solo, Remove, Reorder, and Wet/Dry Mix.

Backer

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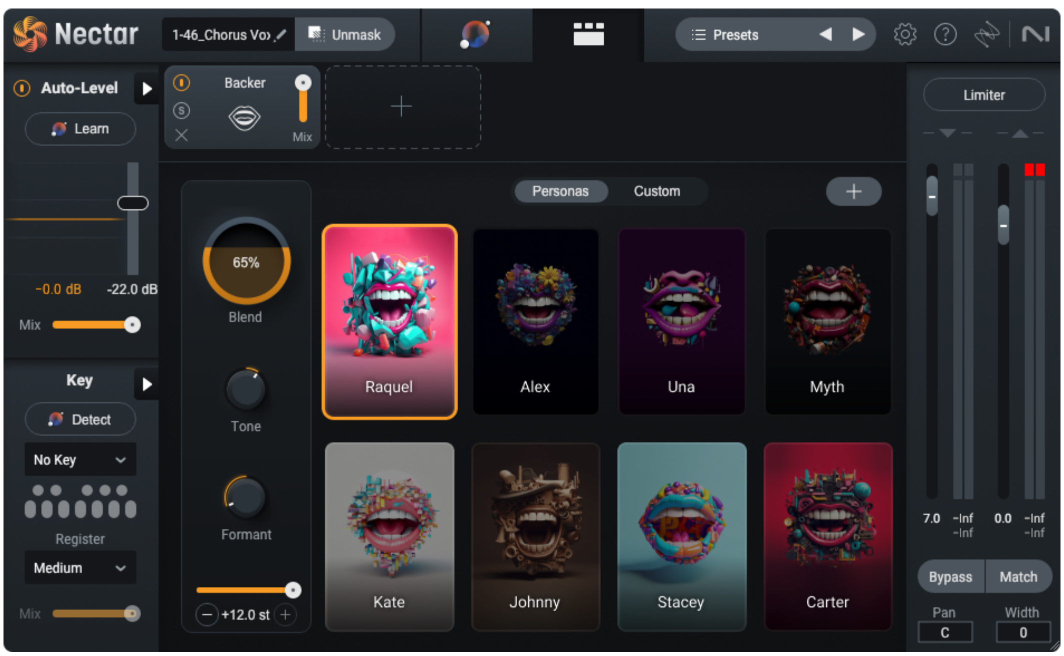
Overview

Backer uses singing voice conversion technology to transform the input vocal into a target (reference) vocal. Singing voice conversion is a technique employing artificial intelligence to capture and shift formants and sibilances.

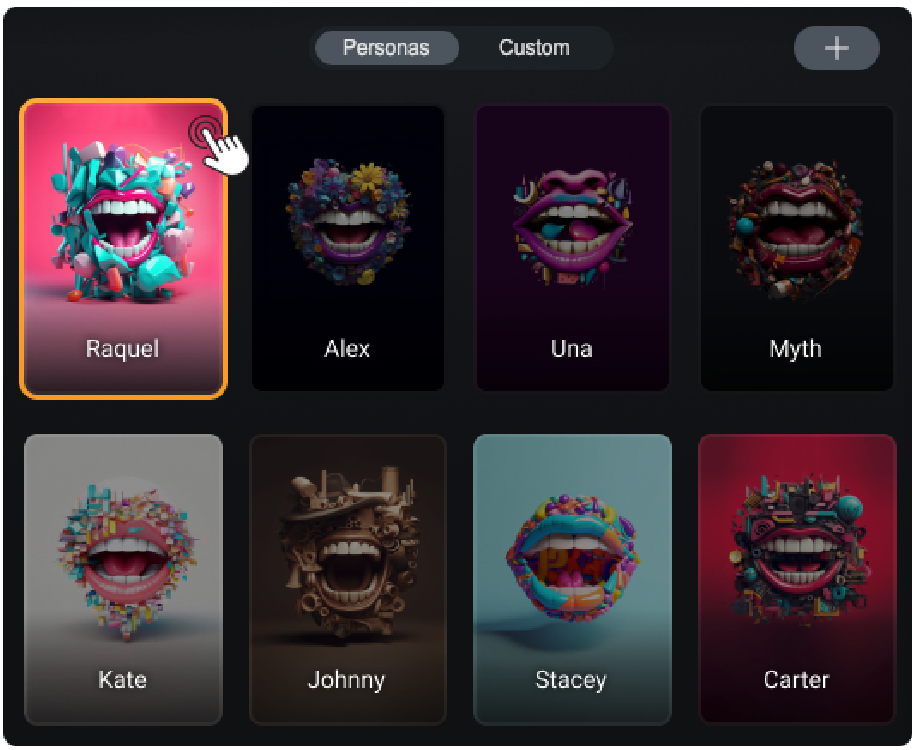


★ **VOCAL REGISTER**

Determines how the Backer module analyzes and detects pitch data in the input signal. The Mid option will work well for a wide range of vocal material. If the pitch correction processing is producing artifacts or other undesirable behaviors, try using the Low or High settings to achieve better results.

Target Personas

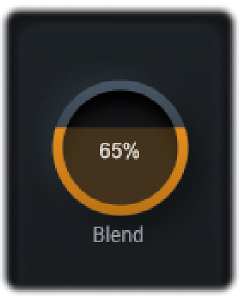
There are 8 target personas available in the Backer Module. Each one was trained on a different person's voice to capture the timbre. The different persona characteristics are described below. Each voice is available as an automation parameter in your DAW, so you can change voices throughout your arrangement.



Control Panel

The control panel hosts several tools for you to modify the sound of the Backer module. These controls are curated for voice processing.

Blend



This mixes the output of the module with the Target persona and the original signal. You can use this to add or remove the sound of the selected timbre in the module.

Tone

This is a Tilt EQ that brightens the high end or darkens the low end. -100 darkens the processed signal +100 brightens the processed signal.

Formant

Shifts transformed vocal formants. A setting of -12 will result in deeper vocal tones, while a setting of +12 will result in brighter vocal tones.

Pitch

Transposes the input signal up or down in pitch by an octave.

Octave Snap

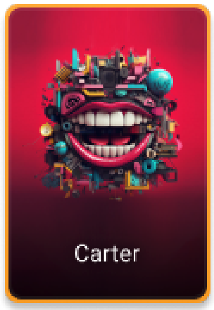
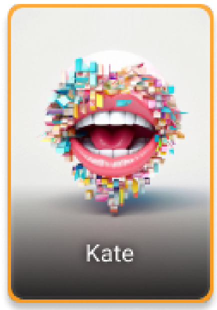
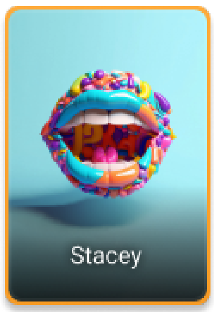
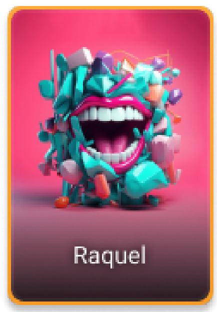
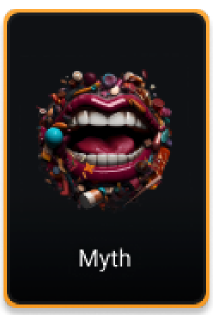
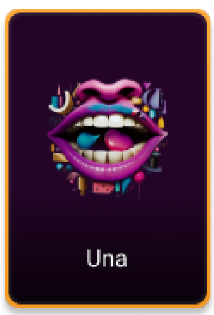
The + or - minus in the pitch controls are available to conveniently snap the pitch setting to an octave above or below quickly.

Target Registers

Note that depending on the chosen vocal register, there may already be a pitch shift (either up or down an octave) behind the scenes to transform to the target persona. You can use the pitch shift controls to modify source pitch if you do not want a pitch shift.

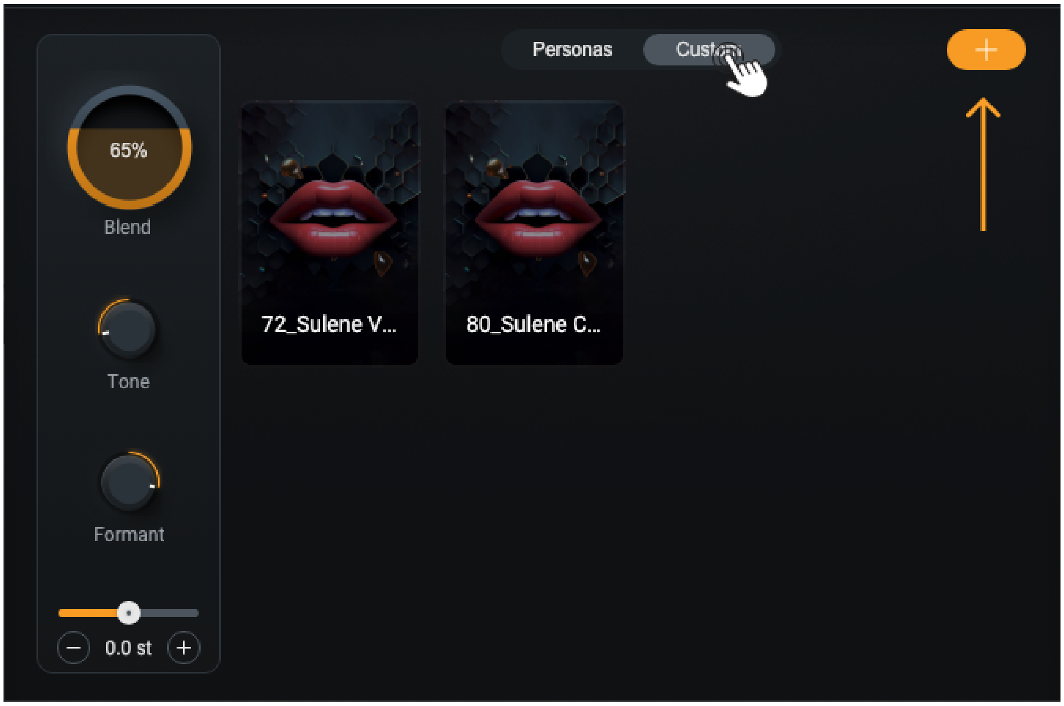
Example: If your source register (the input signal) is set to “High” and the target singer’s register is Low, a pitch shift of -12 semitones will be applied to your input signal.

Personas by Register

Low Register	Medium Register	High Register
		
		
		

Custom

The custom section allows you to upload an audio file to use as a Target for the Backer module. Click the + button to load a file from your desktop.

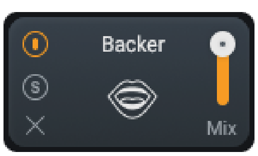


TIP

Backer is optimized for vocal files. Use an isolated vocal stem to get the best results.

Global Module Controls

The module chain features common controls for each module, including Bypass, Solo, Remove, Reorder, and Wet/Dry Mix.



GLOBAL CONTROLS

Learn more about these controls in the **Global Controls** chapter.

Pitch

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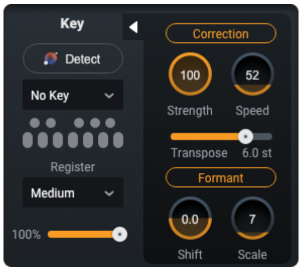
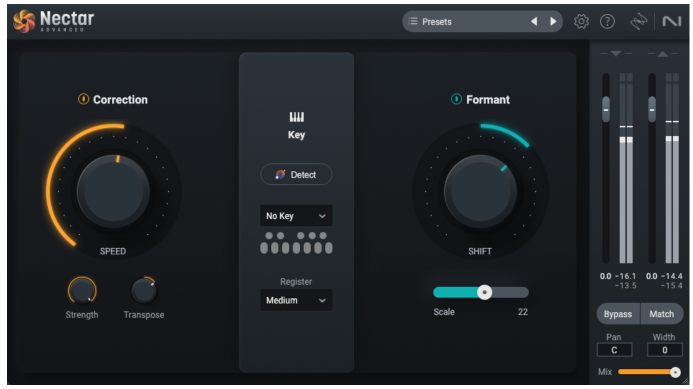
Overview

Pitch correction can be used to fix out-of-tune vocal performances by automatically adjusting incoming notes to conform to the pitches of a specified musical scale. The Pitch module features controls for adjusting and tailoring pitch correction processing.

NECTAR 4 & MELODYNE 5 ESSENTIAL

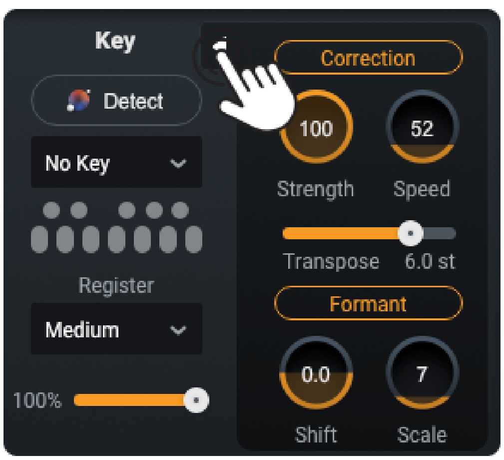
Melodyne 5 essential is included with Nectar 4. It can be used to edit the pitch and timing of individual notes within a vocal track.

Pitch is available as a mini-module in the mothership plug-in and as a component plug-in with Nectar 4 Advanced. The layout differs slightly between the mini-module and the component plug-in, two images are included throughout this chapter where the layouts differ.

Mini-module	Component plug-in
	

Accessing the Pitch Mini-Module

You can access the Pitch mini-module by clicking on the arrow button to the right of the Key label on the left side of the Nectar 4 mothership plug-in. This option is available below Key Detection in the Nectar Standard View.



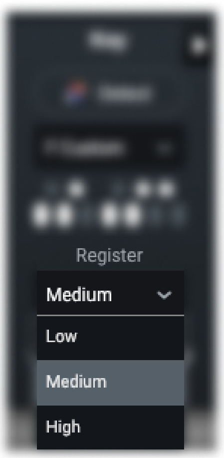
> **SHARED CONTROLS**
>
> The Vocal Register and Scale settings in the mini-module panel are shared by the **Pitch**, **Voices**, **EQ**, and **Backer** modules.
> These controls determine how Pitch & Voices detect pitch and apply correction to the incoming vocal. Additionally Vocal Register determines the vocal range of the input signal so that the Follow EQ feature can work effectively in the EQ and the Backer module can change the timbre of the input signal correctly.

Pitch Controls

> **PITCH MIX SLIDER**
>
> The Mix slider in the Pitch may introduce a doubling type of effect when set to a value that mixes wet and dry signals.

Vocal Register

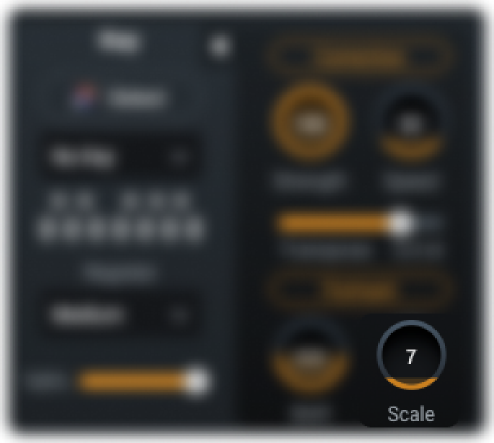
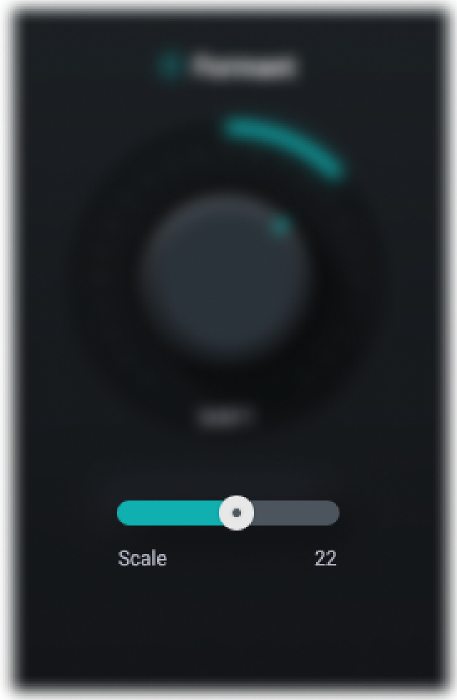
Determines how the **Pitch** and **Voices** modules analyze and detect pitch data in the input signal. The Mid option will work well for a wide range of vocal material. If the pitch correction processing is producing artifacts or other undesirable behaviors, try using the Low or High settings to achieve better results.

mini-module	component plug-in
	

Scale

Sets the type of scale used for pitch correction in the **Pitch** and **Voices** modules. This setting determines the notes that the incoming vocal will be corrected to.

There are three scale types to choose from: **Major**, **Minor**, and **Custom**. The **default** scale type selection is “**No Key**”.

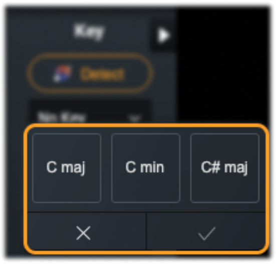
mini-module	component plug-in
	

Auto Detect

When Key mode is selected, the Auto Detect feature can be enabled to assist with selecting the root note and scale type of the incoming vocal.

Click the Auto Detect button and playback audio to allow Nectar to begin listening to the incoming signal. As audio plays back, the three most likely scales will be displayed in the Auto Detect window.

Select the scale you would like to use and click the checkmark button to set the root note and scale type. Click the x button to dismiss the Auto Detect window without changing the current scale settings.

mini-module	component plug-in
	

★ CALIBRATING NECTAR'S REFERENCE PITCH

By default, pitch detection and correction assumes the input material was tuned to a standard A = 440Hz reference pitch. When working with material that was tuned to a different reference pitch, adjusting the Calibration Pitch value in the Options menu will improve the results of pitch detection and correction.

Scale Type

Sets the scale used to correct the incoming signal. You can choose from the following scale types:

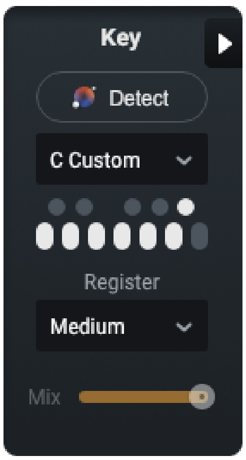
1. **MAJOR**: Notes included in the selected Major key will be used as a target pitches for correction.
    1. **Example**: When C# Major is selected, the following notes will be used as target pitches for correction: C#, D#, E#, F#, G#, A#, B#
2. **MINOR**: Notes included in the selected Minor key will be used as target pitches for correction.
    1. **Example**: When E Minor is selected, the following notes will be used as target pitches for correction: E, F#, G, A, B, C, D

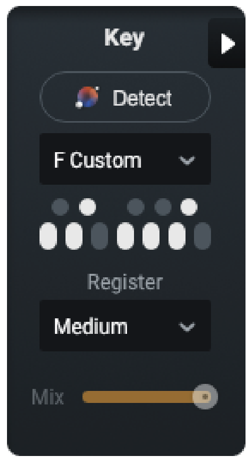
Custom

When the **Custom** scale type is selected, a keyboard display will appear.

Clicking on individual keys will enable or disable them from the custom correction scale. When a key is disabled in this display, it will be excluded from the available target pitches when processing.

Custom scales will automatically transpose if another Custom note is selected. ***Example***: If a ‘C’ custom scale is set to Major notes with a Bb (minor 7th) and then ‘F’ Custom is selected, all the notes will transpose to the same intervals with F as the root note.

<table>
<tr><th>Custom C Major with a b7</th><th>Custom F Major with a b7</th></tr>
<tr><td>


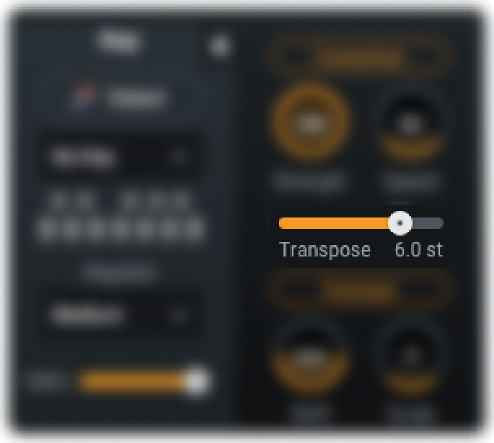
</td><td>


</td></tr>
</table>

Transpose

Determines the static amount of pitch shift applied to the corrected vocal. This control allows for a transposition range of -12 semitones to +12 semitones and is set to 0 (no transposition) by default.

> **NOTE**
>
> Static transposition is only applied by the Pitch module. This processing is always first in the signal flow of Nectar, so all modules will be affected downstream in the signal flow of this processing.

mini-module	component plug-in
	

Correction Controls

The following controls determine the character and speed of automatic pitch correction processing. These controls only apply to processing in the Pitch module.

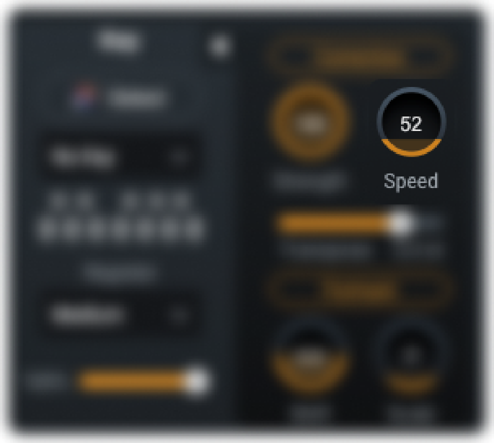
Enable Correction

Enables or disables Pitch Correction processing.

Speed

Determines how quickly incoming pitches will be corrected to notes in the selected scale.

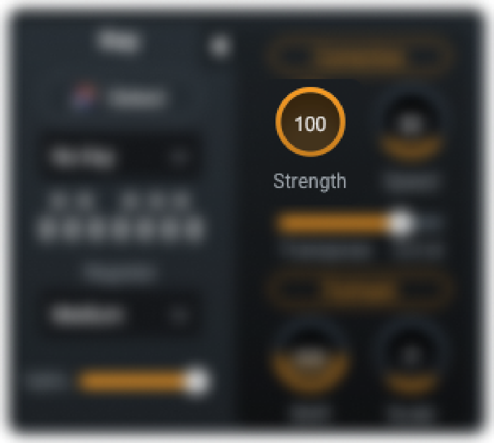
Speed allows for a range of 0 milliseconds to 200 milliseconds. Lower speed settings equate to faster correction response, which can result in more robotic sounding pitch correction. Higher speed settings equate to slower correction response, which can result in more natural sounding pitch correction.

mini-module	component plug-in
	

Strength

Determines how strictly incoming signals will be corrected to notes in the selected scale.

Higher values will eliminate or reduce natural vibrato in the processed vocal. Lower values will retain vibrato in the processed vocal, allowing for more natural sounding results.

mini-module	component plug-in
	

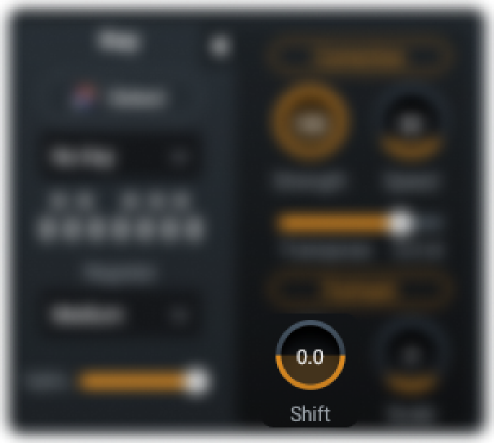
Formant

Enables or disables advanced controls for fine tuning formant shift and formant scaling.

By default, Nectar preserves vocal formants in the incoming signal just as they were recorded. Formants contribute to the timbre of the human voice, an important aspect of a natural sounding vocal. In some cases, it may be necessary to adjust the formant shift and scaling parameters to achieve the best results.

Formant Shift

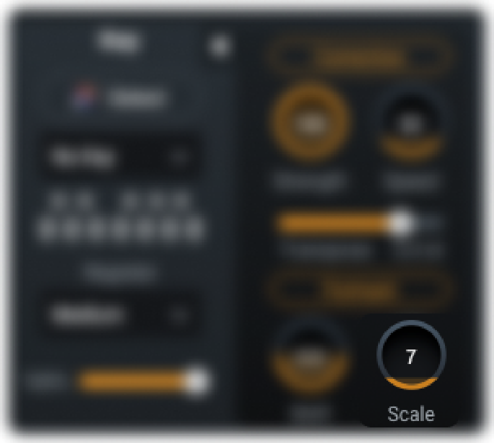
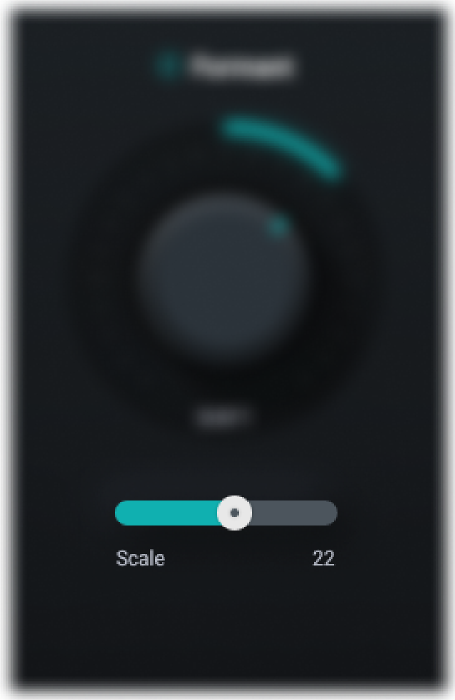
Determines the amount of formant transposition applied to the corrected vocal. This control can be adjusted in increments of semitones (st). In general, the default value of 0 is suitable for most material.

mini-module	component plug-in
	

Formant Scale

Determines the amount of scaling applied to formants when shifting pitch. Scale will always adjust formants in the direction of the pitch shift.

Increasing the amount of formant scaling will shift formants in the direction of the pitch shift. When a singer transitions to a higher note, their vocal formants will shift slightly higher in the direction of that note. Adjusting formant scaling can help to achieve more natural sounding results when a vocal is shifting up or down drastically between notes.

mini-module	component plug-in
	

Vocal Unmask

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Overview

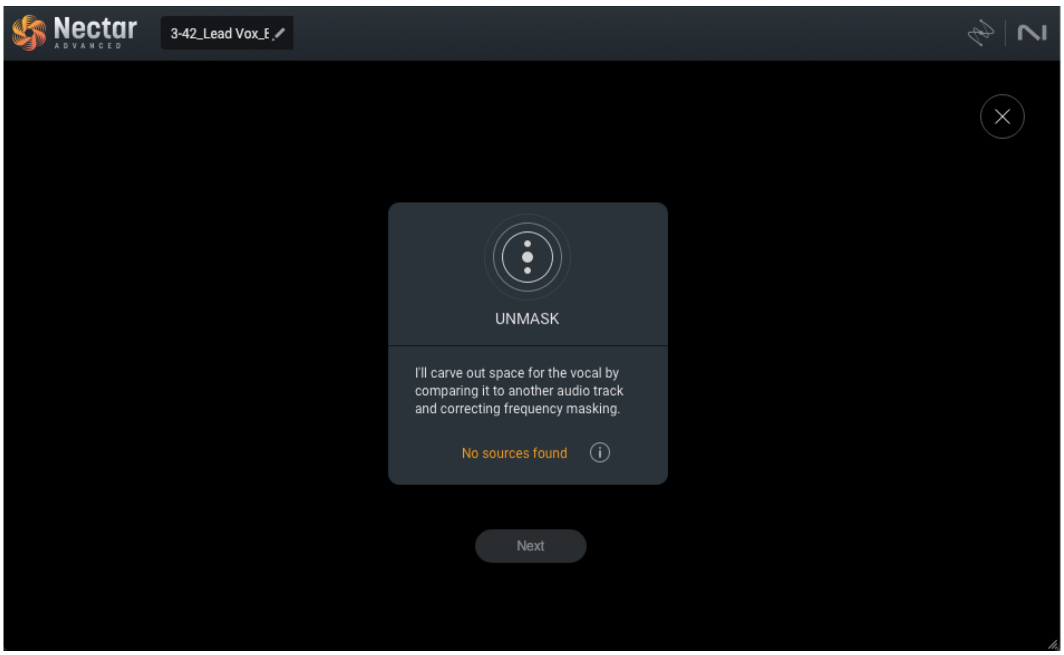
Vocal Unmask will intelligently create an optimal starting point for a vocal. It can also help a vocal stand out in the mix by communicating with IPC compatible iZotope plug-ins to unmask the vocal from other elements in the mix.

Click the “Unmask” button in the header of the plug-in interface to open the Vocal Unmask screen.

Vocal Unmask Configuration

UNMASK REQUIRES AN IPC SOURCE TO WORK

Unmask requires a compatible iZotope plug-in to be inserted on another track in your session in order to work.



Unmask analysis can occur when one of the following plug-ins is present as an insert on the track that is masking your vocal: Relay, Nectar, or Neutron.

After configuring a source plug-in, click Next to continue.

Unmask Source Selection

Select the source IPC plug-in for Unmask analysis.

CHOOSING AN UNMASK SOURCE

1. Select a plug-in in the Unmask dropdown that is inserted on a track that is masking the vocal.
2. For the best results, the source plug-in that is selected in the Unmask dropdown should be after all other inserts on the track it is inserted on.
3. For example, Insert Relay in the last insert slot of a track that is masking the vocal. Select that Relay instance from the Unmask dropdown menu.

Analysis

After configuring the Vocal Unmask Options, ensure audio is playing back and click the “Next” button to continue to the analysis screen.

NOTE

1. **Vocal Unmask needs audio input**: Make sure you are playing audio in your DAW.
2. **Vocal Unmask needs time to analyze your vocal**. If you are trying to run Vocal Unmask on a clip shorter than 20 seconds, you should enable loop playback before starting Vocal Unmask.

Unmask analysis listens to the selected source and the current vocal track and compares them to detect the presence of masking.

If no significant masking is detected between the vocal and the Unmask source, the final Unmask step will state “No significant masking detected” and no EQ curve will be applied to the selected source plug-in.

If masking is detected, an EQ curve will be applied to the instance you selected in the Unmask dropdown. Plug-ins that are unmasking Nectar will update to display an “Unmask” indicator and power button above the I/O level meters.

You can enable/disable Unmask by opening the source plug-in and toggling the power button to the left of the “Unmask” indicator.

Advanced Unmask Controls

Extended Unmask options are available in the selected source plug-in when they are unmasking Nectar.

A down arrow will appear on the right side of the Unmask indicator of a source plugin when advanced unmask controls are available. Click on the Unmask indicator to expand the advanced unmask panel.

The advanced unmask panel includes:

1. **NAME**: The Nectar instance that is being unmasked.
2. **UNMASK EQ CURVE DISPLAY**: Displays the unmask EQ curve being applied to the Nectar instance. Hover your cursor over the EQ curve to display EQ frequency and gain values.
3. **AMOUNT**: Scales the intensity of unmask EQ cuts.
4. **DYNAMIC**: Enables/disables dynamic unmask EQ.

    1. When enabled, unmask EQ cuts will be applied when the masking signal exceeds a fixed threshold. The EQ curve display will animate to reflect changes to the EQ curve when Dynamic mode is enabled.

5. **SIDECHAIN**: Only available when Dynamic mode is enabled. Sets the dynamic EQ detection input to the sidechain input source configured for the plug-in.

Vocal Unmask in **Neutron**	Vocal Unmask in **Relay**	Vocal Unmask in **Nectar**

Apply Settings

To apply the results of Vocal Unmask, click the “Accept” button.

To re-run Vocal Unmask processing before applying settings, click the back button.

To exit the Vocal Unmask screen without applying any changes, click the x button.

Global Controls

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Overview

The Module Chain and I/O section include a number of parameters for customizing the IPC instance name, signal flow, levels, and stereo image of Nectar 4.

Resizable Window

Nectar features a resizable main window. The window can be resized by clicking and dragging the bottom right corner of the plug-in window.

Plug-in Instance Name

Determines the name of the current instance when it appears in IPC lists in supported iZotope plug-ins.

Module Chain

The Module Chain allows for highly customizable vocal processing chains.

The following modules can be added to the module chain:

1. **Backer**
2. **Compressor (x2)**
3. **De-Esser**
4. **Delay**
5. **Dimension**
6. **EQ (x2)**
7. **Gate**
8. **Reverb**
9. **Saturation**
10. **Voices**

The following functions and controls are available in the module chain: **Add**, **Reorder**, **Enable**, **Solo**, **Remove**, and **Mix**.

Add

Clicking the **+** button in the rightmost position of the module chain will open the module menu. Select a module from the list to add it to the last slot in the module chain.

> **MODULE LIMITS**
>
> Most modules can only be added to the module chain once. If a module has been added to the module chain already, the option in the module list will be greyed out. **Only the EQ and Compressor modules can be added to the module chain twice.**

Reorder

Click and drag a module panel left or right within the Module Chain to change its order in the signal flow.

Enable

Click the power button the upper left corner of a module tile to bypass processing of that module.

Solo

Click the **S** button on the left side of a module tile to bypass processing in all *other* modules.

Remove

Click the **X** button on the left side of a module tile to remove the associated module from the module chain.

Mix

The slider on the right side of a module tile adjusts the balance between the dry (unprocessed) and wet (processed) signals.

Limiter

Enables a brickwall, zero latency limiter on the output signal. The Limiter is applied to the signal after output gain.

Limiter Ceiling

Determines the maximum output level of the plug-in when the limiter is enabled. The hard limiter uses the Ceiling as an absolute guide, and the final output level will not exceed this point.

1. Click and drag the slider handle up or down to adjust the Ceiling value.
2. Click on the Ceiling slider handle, then click on the text readout display that appears. Manually enter a value in the inline edit field.

Gain Reduction Meter

Displays the amount of gain reduction applied to the output signal by the Limiter.

Input and Output

Meters

The Input and Output meters display Peak and RMS metering information. The Peak level meter is displayed in white. The RMS level meter is displayed in grey.

The text readouts directly below the meters display the current Peak and RMS values. The current Peak value is displayed in white. The current RMS value is displayed in light grey.

Gain

The Input and Output Gain can be adjusted using the sliders to the left of the input and output meters.

TIPS: ADJUSTING THE I/O GAIN

1. Click and drag the slider handle up (increase input or output gain) or down (decrease input or output gain).
2. Double-click on the text readout directly below the gain sliders and manually enter a gain value in the inline edit field.

Bypass

Bypasses all processing applied by Nectar.

Match

When Nectar is bypassed and Match is enabled, the bypassed signal level will be adjusted to match the processed output level.

Width

Adjusts the amount of stereo widening. Decreasing this control results in a narrowing effect (-100% is equivalent to mono), increasing this control widens the apparent stereo field.

> **STEREO INSTANCES ONLY**
>
> Width is only functional in stereo instances of Nectar.

Pan

Pans the output signal to the left or right channel.

> **STEREO INSTANCES ONLY**
>
> Pan is only functional in stereo instances of Nectar.

Compressor

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Overview

A compressor can be used to reduce dynamic range, maintain consistent levels, and shape the tone and character of a vocal track. Nectar includes **two** Compressor modules that can be placed in series within the **module chain**, allowing for a variety of vocal compression configurations. The Compressor modules include **controls** for adjusting the amount, speed, and character of processing and robust **metering** for visualizing the effect of processing.

Controls

The controls in the Compressor module work together to influence the character, amount, and speed of the compression applied to the input signal. The following controls are available in the Compressor module:

1. **Mode**
2. **Level Detection Mode**
3. **Threshold**
4. **Ratio**
5. **Attack**
6. **Release**
7. **Makeup Gain**
8. **Auto Gain**

Mode

Determines the processing algorithm used by the Compressor. There are four unique mode options available: **Digital**, **Vintage**, **Optical**, and **Solid-state**. The Mode selection influences the character of compression and will also determine the ranges of the **Ratio**, **Attack**, and **Release** controls.

The following table describes the sonic characteristics and associated control ranges for each of the compressor modes.

Mode	Description
Digital	**Modern, surgical compression.** Ideal for achieving precise, clean sounding linear compression with minimal coloration. 1. **Ratio:** 1:1 to 50:1. 2. **Attack:** 0.1ms to 300ms. 3. **Release:** 1ms to 1200ms.
Vintage	**Classic, analog compression.** Emulates the program-dependent response and non-linear release characteristics of classic analog compressors. 1. **Ratio:** 1:1 to 50:1. 2. **Attack:** 0.1ms to 300ms 3. **Release:** 1ms to 1200ms.
Optical	**Smooth and transparent compression with subtle coloration.** Emulates the subtle harmonic coloration and non-linear attack and release characteristics of classic hardware optical compressors. 1. **Ratio:** 4:1 (Ratio is fixed to 4:1 when Optical mode is selected.) 2. **Attack:** 1ms to 100ms. 3. **Release:** 40ms to 200ms. ★ **TIP** Use **RMS** **level detection mode** (rather than Peak level detection) to more faithfully emulate the sound of hardware optical compressors, which utilized RMS level detection.

Mode	Description
Solid-State	**Clear but aggressive compression with unique harmonic coloration.** Emulates the fast attack, non-linear release times and harmonic characteristics of early VCA transistor-based hardware compressors. Solid-state mode can be useful for accentuating vocal transients. 1. **Ratio:** 4:1 to 12:1. 2. **Attack:** 0.2ms to 80ms. 3. **Release:** 50ms to 1200ms. **TIP** Use **Peak** **level detection mode** (rather than RMS level detection mode) to best highlight the pleasing harmonic coloration effect of the Solid-state algorithm.

Level Detection Mode

The Level Detection Mode determines how input levels are calculated by the compressor. Adjusting the level detection mode will alter the level that is considered by the threshold, which will affect when or how often the input level will trigger compression.

The Compressor includes two level detection modes: **Peak** and **RMS**.

1. **Peak:** Determines input level to the compressor using instantaneous peak levels of the incoming signal.
2. **RMS (Root Mean Square):** Determines input level to the compressor by averaging levels of the incoming signal.

Threshold

Determines the signal level at which the compressor begins processing.

When the input level exceeds the threshold level, the compressor will be triggered. Signals that exceed the threshold level will be reduced according to the **Ratio**. The attack phase of the compressor begins when the input level exceeds the threshold level. The release phase of the compressor begins when the input level falls below the threshold level.

THRESHOLD TIPS

1. Click and drag the Threshold slider handle up (to increase the Threshold) or down (to decrease the Threshold).
2. Click on the Threshold value readout text and manually enter a value in theinline edit field.
3. Use the **waveform displays** as a visual guide when setting the threshold level for the compressor.

Ratio

Determines how much gain reduction will be applied to signals that exceed the **threshold** level. When ratio is set to a value of 1:1, no attenuation will be applied to signals that exceed the threshold. Ratio settings of 10:1 or greater allow the compressor to function as a limiter. Limiting ratios can be used to ensure that the output signal level does not exceed the threshold level.

OPTICAL MODE RATIO SETTING

When Optical mode is selected, the Ratio control is fixed to 4:1.

Attack

Adjusts the amount of time it takes for the compressor to apply gain reduction when the input signal exceeds the threshold. Attack time can be adjusted in increments of milliseconds (ms).

Release

Adjusts the amount of time it takes for the compressor to stop processing when the input signal falls below the threshold. Release time can be adjusted in increments of milliseconds (ms) and are typically longer than attack times.

Makeup Gain

Determines the amount of static gain applied to the output signal after compression.

Auto Gain

Automatically adjusts the gain of the compressed signal in order to match the level of the input signal.

Global Module Controls

The module chain features common controls for each module, including: Bypass, Solo, Remove, Reorder, and Wet/Dry Mix.

GLOBAL CONTROLS CHAPTER

To learn more about the module chain and other global controls, visit the **Global Controls** chapter.

Meters

The following meters illustrate how the Compressor is responding to and processing the input signal:

1. **Waveform Displays**
2. **Gain Reduction Trace**
3. **Gain Reduction Meter**

Waveform Displays

The scrolling waveform meters display the amplitude of the input (uncompressed) and output (compressed) signals over time. The meters scroll from right to left, with the most recent information on the right.

The **compressed output signal** waveform is displayed in **light grey**, in front of the input signal waveform. The **uncompressed input signal** waveform is displayed in **dark grey**, behind the output signal waveform. When the signal is being compressed, the gain reduction applied to the output signal can be observed in the difference between the two waveforms.

> ★ **NOTE: AUTO GAIN IN THE OUTPUT WAVEFORM**
>
> Auto gain adjusts the level of the compressed signal to compensate for any level difference between the uncompressed and compressed signals. The gain change introduced by auto gain is reflected in the output waveform, which may make it more difficult to differentiate between the input and output waveforms. **The gain reduction trace meter can be useful for monitoring gain reduction over time when auto gain is enabled.**

Gain Reduction Trace

The yellow trace line indicates the gain reduction applied by the compressor over time. The trace can be used to monitor the response times (attack and release phases) and gain reduction applied over time.

Gain Reduction Meter

Displays the current average amount of gain reduction applied to the signal. This meter displays gain reduction in decibels (dB).

De-esser

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Overview

A de-esser can be used to control sibilance and tame other high frequency issues in a vocal track. Traditionally, a de-esser dynamically reduces loud sibilant content using a threshold and ratio. The Nectar De-esser module is a hardware-modeled level independent processer, allowing for consistent and transparent reduction of sibilance in signals with variable levels, like a vocal track. The De-esser works by analyzing the current level above a specified **frequency cutoff** and comparing that level against the level of the full frequency bandwidth of the signal. When large differences in level are detected, gain reduction will be applied to the entire incoming vocal signal.

Controls

The De-esser module includes controls for adjusting the detection and reduction of sibilant frequency content. The controls are positioned on the right side (upper frequency range) of the De-esser module panel.

Detection Filter Cutoff

Determines the lower frequency boundary for the detection filter frequency band. Any frequency content that is above the cutoff frequency will be used for De-esser detection.

The detection filter cutoff can be set to frequency values ranging from 800 Hz (Hertz) to 8 kHz (kilohertz).

ADJUSTING THE DETECTION CUTOFF FREQUENCY

The detection filter cutoff frequency can be adjusted using the following methods:

1. Click and drag the node at the top of the De-esser module panel to the left (lower frequency value) or right (higher frequency value).
2. Click on the cutoff frequency text readout display and enter a value manually in the inline edit field. The current frequency readout is located on the bottom edge of the De-esser module panel, to the right of the Detection Cutoff Frequency line.
3. Double-click on the cutoff node handle to reset the Detection Cutoff Frequency to the factory default value.

Threshold

Determines the threshold level for ess reduction and the amount of gain reduction applied to the incoming signal when sibilance is detected.

When sibilance is detected and its level exceeds the Threshold level, the De-esser will apply reduction to the entire bandwidth of the incoming signal.

The amount of reduction applied to the signal depends on how much the sibilance level exceeds the Threshold. More gain reduction will be applied as the sibilance level increases farther above the Threshold level.

ADJUSTING THE THRESHOLD

The Threshold can be adjusted using the following methods:

1. Click and drag the Threshold handle up and down to adjust the level.
2. Click on the Threshold level readout text and enter a value manually in the inline edit field. The Threshold level readout is located on the right edge of the De-esser module panel, directly above the Threshold line.
3. Double-click on the Threshold slider handle to reset the Threshold to the factory default value.

Listen

When enabled, the signal content that is being reduced by the De-esser is played back in isolation.

TIP

Engage Listen to monitor the difference between the unprocessed and processed signals. This outputs only the signal content being reduced by the De-esser. Try adjusting the Threshold and Detection Filter Cutoff until the Listen output only contains the ess sounds you are trying to reduce.

Global Module Controls

The module chain features common controls for each module, including Bypass, Solo, Remove, Reorder, and Wet/Dry Mix.

GLOBAL CONTROLS CHAPTER

To learn more about the module chain and other global controls, visit the **Global Controls** chapter.

Meters

The De-esser module features two spectrum analyzers (pre- and post-processing) and a gain reduction meter for monitoring the effect of the De-esser processing.

Spectrum Analyzer

Displays the magnitude of a signal across the frequency spectrum in real-time. The vertical ruler on the left edge of the module panel measures the amplitude of the signal in decibels (dB). The horizontal ruler along the bottom edge of the module panel measures frequency in Hertz (Hz).

Two spectrum analyzers are displayed in the De-esser module: the input to the De-esser module (displayed in dark grey with no border) and the output of the De-esser module (displayed in light grey with a white border).

Gain Reduction Meter

Displays the current average amount of gain reduction applied to the signal. This meter displays gain reduction in decibels (dB).

Delay

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Overview

Delay can be used to thicken and add a sense of space to a vocal track. The Delay module features stereo delay processing with five distinct saturation modes and modulation options.

Controls

The Delay module controls panel includes parameters for customizing the timing and character of the delay effect.

The Delay controls panel includes the following groups of parameters:

1. **Delay**
2. **Saturation**
3. **Modulate**

Delay Controls

The Delay module offers different controls depending on the channel count of the track that Nectar is inserted on.

1. **MONO:** When Nectar is inserted on a mono track, a single set of delay controls (Delay, Tempo Sync, and Feedback) are available in the Delay module controls panel.

2. **STEREO:** When Nectar is inserted on a stereo track, two sets of delay controls (Delay, Tempo Sync, and Feedback) are available in the Delay module controls panel, allowing for independent or linked adjustment of left and right channel

delays.

Delay

Determines the amount of time between each successive repeat of the incoming signal.

When **Tempo Sync** is disabled for the associated channel, the Delay control will be adjusted in increments of milliseconds (ms), ranging from 1ms to 3500ms.

Tempo Sync

When enabled, the Delay control will be adjusted in increments of musical note values. The duration of the musical note value is based on the tempo of the session.

When Tempo Sync is enabled, the Delay control supports note values ranging from **1/64T** (fast - shortest delay between repeats) to **8.** (slow - longest delay between repeats).

EXAMPLE: TEMPO SYNC NOTE DURATIONS

1. **When the session Tempo is set to 120 BPM:** The duration of a quarter note is 500 milliseconds, meaning the input signal will be delayed for 500ms before repeating.
2. **When the session Tempo is set to 220 BPM:** The duration of a quarter note is 273 milliseconds, meaning the input signal will be delayed for 273ms before repeating.

Feedback

Determines the amount of delayed signal that is fed back into the input of the delay.

Link

Enables channel linking of delay control values when the Delay module is instantiated as a stereo effect.

NOTE: RIGHT CHANNEL LINK BEHAVIOR

When Link is enabled, the right channel Delay and Feedback values will snap to match the left channel values. When Link is disabled, the right channel controls will snap back to the values set before enabling Link.

Saturation

The Delay control panel includes parameters for adjusting the saturation mode and amount.

Mode

Determines the type of saturation effect applied to the delayed signal.

The following table outlines the sonic characteristics of the different saturation modes.

Mode	Characteristics
Digital	**Transparent, clean delay.** Provides exact copies of the incoming signal with minimal coloration.
Tape	**Classic, colorful drive.** Filters and distorts each progressive repeat in the way a signal would be degraded over time on magnetic tape.
Analog	**Gritty, distorted edge.** Based on circuit distortion, can lend gritty edge to the delayed signal.
Grunge	**Dirty, degraded character.** Adds post-feedback crunchy drive to the delayed signal.
Echo	**Pronounced presence boost.** Repeats and fades out tape saturated copies for adding creative depth and impactful presence.

Amount

Adjusts the amount of drive applied to the saturation.

Modulate

Enables Rate and Depth parameters for adjusting LFO modulation of delay time.

1. **Rate**: Determines the speed of the LFO modulation, i.e *how often* the delay time is modified by the LFO.
2. **Depth**: Determines the intensity of LFO modulation, i.e. *how much* the delay time is modified from the baseline Delay time value.

Post Filter

The highpass and lowpass filters in the Delay module allow for undesirable low and high frequency content to be filtered out of the processed signal.

POST FILTER

1. The high and lowpass filters in the Delay module are **only** applied to the wet (processed) output of the module.
2. Click and drag the filter node left and right to adjust frequency.

Global Module Controls

The module chain features common controls for each module, including: Bypass, Solo, Remove, Reorder, and Mix.

GLOBAL CONTROLS CHAPTER

To learn more about module chain and other global controls, visit the **Global Controls** chapter.

Meters

The Delay module features two spectrum analyzers for monitoring the effect of processing.

Spectrum Analyzer

Displays the magnitude of a signal across the frequency spectrum in real-time. The vertical ruler on the left edge of the module panel measures the amplitude of the signal in decibels (dB). The horizontal ruler along the bottom edge of the module panel measures frequency in Hertz (Hz).

Two spectrum analyzers are displayed in the Delay module: the input to the Delay module (displayed in dark grey, with no border) and the output of the Delay module (displayed in light grey with a white border).

Dimension

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Overview

A modulation effect (e.g. chorus, flanger, or phaser) can be used to add depth and movement to a vocal track. Modulation effects work by duplicating an input signal, modifying an aspect of the duplicated signal using an LFO (low frequency oscillator) and then mixing the original and modified signals together. The **effect modes** in the Dimension module allow for a range of subtle to wacky vocal modulation effects.

Controls

The Dimension module controls panel includes parameters for customizing the different modulation effects.

The following parameters are available in the Dimension controls panel:

1. **Effect Modes**
2. **Modulation Controls**
3. **Width**
4. **Freq** *(Only available in Phaser effect mode)*

Effect Modes

The Dimension module includes three different modulation effects, Chorus, Flanger and Phaser. The effect mode can be changed using the buttons on the left side of the Dimension controls panel.

The following table describes how the different effect modes apply processing to an input signal.

Mode	Description
Chorus	Chorus effects work by creating a delayed copy of a signal and modulating the delay with an LFO (low frequency oscillator). This results in gradual changes to the pitch of the copied signal. The modified signal and the input signal are mixed back together to create a pitch-modulated doubling effect, simulating the effect of two or more performers singing in unison.
Flanger	Similar to a Chorus, Flangers work by creating a delayed copy of a signal and modulating the delay with an LFO (low frequency oscillator). The modified signal and the input signal are then mixed together, resulting in a sweeping comb filter effect. *Flanger delay times are shorter than chorus delay times.*
Phaser	Phasers work by applying an all-pass filter to a copy of the input signal, creating a series of notches across the frequency spectrum. The positions of the frequency notches are modulated with an LFO (low frequency oscillator). The input signal and modulated signal are mixed together to create a sweeping effect. *Unlike flanger and chorus effects, Phasers do not use delay to modify the signal.*

Modulation Controls

The controls in the center of the Dimension module controls panel adjust the rate and depth of LFO modulation, as well as the amount of processed signal that is fed back into the input of the effect.

Rate

Determines the speed of the LFO modulation. When Chorus or Flanger **effect modes** are selected, the Rate control determines how often the delay time is modified. When the Phaser **effect mode** is selected, the rate control determines how often the filter positions are modified.

When **Tempo Sync** is disabled, the Rate control range extends from 0.01 Hz (slow - longest modulation cycle) to 4.00 Hz (fast - shortest modulation cycle).

Tempo Sync

When enabled, the **Rate** control can be adjusted in increments of musical note values. The duration of the musical note value is based on the tempo of the session.

When Tempo Sync is enabled, the Rate control supports note values ranging from **1/64T** *(fast - shortest modulation cycle)* to **8.** *(slow - longest modulation cycle).*

EXAMPLE: TEMPO SYNC NOTE DURATIONS

1. **When the session Tempo is set to 120 BPM:** The duration of a quarter note is 500 milliseconds, meaning the LFO will complete a cycle once every 500ms.
2. **When the session Tempo is set to 220 BPM:** The duration of a quarter note is 273 milliseconds, meaning the LFO will complete a cycle once every 273ms.

Depth

Determines the intensity of LFO modulation. When Chorus or Flanger **effect modes** are selected, the Depth control determines how much the delay time between signals is modified from the default delay value. When the Phaser **effect mode** is selected, the

Depth control determines how much the filter positions are modified from the default filter positions.

The Depth control range extends from 0% (effectively no modulation) to 100%.

Width

Determines the amount of stereo widening applied to the output of the modulation effect.

The Width control range extends from 0% (no stereo widening) to 100%.

Freq

Determines the starting frequency of the all-pass filter applied by the Phaser.

> **NOTE**
>
> The Freq control is only enabled when the Phaser effect mode is selected.

Global Module Controls

The module chain features common controls for each module, including Bypass, Solo, Remove, Reorder, and Wet/Dry Mix.

> **GLOBAL CONTROLS CHAPTER**
>
> To learn more about the module chain and other global controls, visit the **Global Controls** chapter.

Meters

The Dimension module features two spectrum analyzer displays for monitoring the effect of processing.

Spectrum Analyzer

Displays the magnitude (level) of a signal along the frequency spectrum in real-time. The vertical ruler on the left edge of the module panel measures the amplitude of the signal in decibels (dB). The horizontal ruler along the bottom edge of the module panel measures frequency in Hertz (Hz).

Two spectrum analyzers are displayed in the Dimension module: the input to the Dimension module (displayed in dark grey, with no border) and the output of the Dimension module (displayed in light grey, with a white border).

EQ

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Overview

An equalizer can be used to enhance the tone and character of a vocal track by adjusting the level of specific frequencies. Nectar includes **two** EQ modules with support for adding up to 24 highly customizable bands. Each band includes 16 different filter shapes and two program dependent processing modes: **Dynamic Frequency** mode for dynamically updating the frequency of a filter in response to harmonic frequency content changing over time, and **Dynamic Gain mode** mode for dynamically adjusting the gain of a filter in response to the input signal.

Interactions

1. **Add Bands**
2. **Remove Bands**
3. **Alt-Solo**

Add Bands

Bands can be added to the EQ using the following methods:

1. Click on the **+** button that appears when hovering the mouse cursor over the white **composite curve** line in the EQ display.

    1. Single-click to place a new node in the location of the click.
    2. Click and drag on the **+** button to quickly add a node and drag it to a new position.

2. Double-click anywhere in the EQ spectrum display to add and position a new node in the location of the click.
3. Double-click and drag to quickly position the node in the location of the click, and drag it to a new position.
4. Use Command+Return (Mac) or Control+Return (Windows) to add a new node to the center of the EQ spectrum.

Remove Bands

Bands can be removed from the EQ using the following methods:

1. Select a node and click the **x** button in the band controls panel to remove it.
2. Click and drag over nodes in the EQ spectrum panel to select multiple nodes. Use the delete or backspace key to remove all selected nodes.
3. Shift-click to select multiple nodes. Use the delete or backspace key to remove all selected nodes.

Alt-Solo

The EQ module includes an Alt-Solo feature that allows frequency bands to be played back in isolation without affecting any EQ settings.

Alt-Solo can be enabled by holding the Alt/option key when clicking anywhere in the EQ spectrum to solo the frequencies surrounding the location of the click.

> **ALT-SOLO Q**
>
> The Q (bandwidth) of the Alt-Solo filter can be adjusted in the **Options** window.

Controls

To access the band controls panel for a given band, click on the corresponding node in the EQ module panel.

The following controls are included in the band controls panel:

1. **General Band Controls**
2. **Frequency**
3. **Gain**
4. **Bandwidth**
5. **Filter Shape**
6. **Dynamic Processing Modes**

General Band Controls

The left side of the band controls panel includes parameters for enabling/disabling, soloing, and removing the currently selected band.

Enable

Enables or disables processing of the currently selected band.

NOTE

When a band is disabled, the filter will still be displayed in the EQ module panel. Disabled bands are displayed in grey to clearly differentiate them from enabled bands.

Solo

Enables temporary cutoff filters that allow frequencies within the bandwidth of the selected filter to be listened to in isolation.

TIP: SOLO BAND SHORTCUT

Hold the Alt and click on an EQ node to quickly engage the band **Solo** functionality.

Remove

Removes the currently selected band.

Frequency

Determines the center frequency (or cutoff frequency) (Hz) of the currently selected node. The EQ supports frequency values ranging from 20 Hz to 20 kHz.

TIPS: ADJUSTING EQ FILTER FREQUENCY

The Frequency of an EQ band can be modified using the following methods:

1. Click and drag an EQ node to the left (to reduce the frequency value) or right (to increase the frequency value).
    1. Hold Shift while dragging a node to the left or right to lock the movement of the node to the horizontal axis.
2. Double-click on the Frequency value readout and manually enter a Frequency value in the inline edit field.
3. Select a node and use the Left arrow key to decrease Frequency or use the Right arrow key to increase Frequency.
    1. Hold the **Shift** modifier key to make **coarse** value adjustments.
    2. Hold the **Command** (Mac) or **Ctrl** (Windows) modifier key to make **fine** value adjustments.

Gain

Determines the amount of gain (dB) applied by the selected filter. The EQ supports gain adjustments ranging from -30 dB to +15 dB.

TIPS: ADJUSTING EQ FILTER GAIN

When available for adjustment, the Gain of an EQ band can be modified using the following methods:

1. Click and drag an EQ node up (to increase gain) or down (to decrease gain).
    1. Hold Shift while dragging a node up or down to lock the movement of the node to the vertical axis.
2. Double-click on the Gain value readout and manually enter a Gain value in the inline edit field.
3. Select a node and use the Up arrow key to increase Gain or use the Down arrow key to decrease Gain.
    1. Hold the **Shift** modifier key to make **coarse** value adjustments.
    2. Hold the **Command** (Mac) or **Ctrl** (Windows) modifier key to make **fine** value adjustments.
4. Double-click on a node to reset the Gain and Q/Slope to their default values.

EQ GAIN SCALE

The vertical dB magnitude scale on the **right** edge of the EQ module panel measures the EQ filter Gain.

Bandwidth

Determines the width or slope (dB/octave) of the selected filter. Bandwidth is labeled as either **Q** or **Slope** in the band controls panel, depending on the selected **filter shape**.

TIPS: ADJUSTING EQ FILTER Q/SLOPE

The Q/Slope of an EQ band can be adjusted using the following methods:

1. Click and drag the node handles toward the node to decrease the bandwidth.
2. Click and drag the node handles away from the node to increase the bandwidth.
3. Double-click on the Q/Slope value readout and manually enter a Q/Slope value in the inline edit field.
4. Hover the cursor over the node and use the mousewheel to adjust the Q/Slope value.
5. Select a node and use **Alt/Option+Left Arrow Key** to **decrease** the bandwidth or use **Alt/Option+Right Arrow Key** to **increase** the bandwidth.
    1. Hold the **Shift** modifier key to make **coarse** value adjustments.
    2. Hold the **Command** (Mac) or **Ctrl** (Windows) modifier key to make **fine** value adjustments.
6. Double-click on a node to reset the Gain and Q/Slope to their default values.

Filter Types

The EQ module features 16 different **filter shapes**, each one belonging to one of the following filter type categories:

Pass Filter

Pass type filters are used to attenuate frequency content that is below (in the case of a high pass) or above (in the case of a low pass) a specified cutoff frequency. The degree of attenuation applied to content above or below the cutoff is determined by the slope of the filter. **The EQ module includes four Pass type filter shapes in the Highpass and Lowpass sub-menus.**

Peak Filter

Peak type filters are used to boost or cut the level of a specific center frequency. The amount of boost or cut applied to frequencies surrounding the center frequency is determined by the Q (or “Bandwidth”) of the filter. **The EQ module includes four Peak type filter shapes in the Bell sub-menu.**

Shelf Filter

Shelf type filters are used to boost or cut the frequency content above or below a specified frequency by the same amount. **The EQ module includes Shelf type filter shapes in the Low Shelf and High Shelf sub-menus.**

Filter Shape

Determines the filter shape of the currently selected band.

The filter shape dropdown menu organizes the filter shapes into the following sub-menus: **Bell**, **Low Shelf**, **High Shelf**, **Lowpass**, and **Highpass**.

Bell

The Bell sub-menu includes the following **Peak** type filters:

	Filter Shape	Description
	Bell	Smoothly boosts or cuts an adjustable region around a specific frequency. Looks like a bell, come on what do you want from me.
	Proportional Q	Unique filter that varies shape in proportion to the amount of boost or cut applied. As a cut or boost is increased further away from center of the EQ curve, the shape tightens for more precision.
	Vintage	Asymmetrical bell filter that is more narrow when cutting frequencies than when boosting frequencies.
	Band Shelf	Bell filter with wide, flat top. Useful for boosting or attenuating a block of frequencies.

Low Shelf

The Low Shelf sub-menu includes the following **Shelf** type filters:

	Filter Shape	Description
	Analog	Efficient shelf filter for simple boosts and cuts.
	Resonant	Exhibits a complimentary resonance at both ends of the filter slope creating a complex shape with one node.
	Vintage	Modeled after the renowned Pultec analog equalizer. Exhibits a complimentary frequency dip, creating a complex slope with one node.
	Baxandall Bass	Gentle low frequency shelf. Modeled after the Baxandall EQ, with the addition of freely adjustable frequency.

High Shelf

The High Shelf sub-menu includes the following **Shelf** type filters:

	Filter Shape	Description
	Analog	Efficient shelf filter for simple boosts and cuts.
	Resonant	Exhibits a complimentary resonance at both ends of the filter slope creating a complex shape with one node.
	Vintage	Modeled after the renowned Pultec analog equalizer. Exhibits a complimentary frequency dip, creating a complex slope with one node.
	Baxandall Treble	Gentle high frequency shelf. Modeled after the Baxandall EQ, with the addition of freely adjustable frequency.

Highpass

The Highpass sub-menu includes the following **Pass** type filters:

	Filter Shape	Description
	Flat	Butterworth filter; optimized for maximum flatness without ripple or resonance in the passband or stopband.
	Resonant	Filter equipped with a resonance control to emphasize the cutoff frequency with positive gain.

Lowpass

The Lowpass sub-menu includes the following **Pass** type filters:

	Filter Shape	Description
	Flat	Butterworth filter; optimized for maximum flatness without ripple or resonance in the passband or stopband.
	Resonant	Filter equipped with a resonance control to emphasize the cutoff frequency with positive gain.

Dynamic Processing Modes

The dynamic processing modes can be accessed by selecting a node and clicking the arrow button on the right hand side of the band controls panel.

Dynamic gain or frequency processing can be enabled by clicking the **Gain** or **Frequency** buttons in the dynamic controls panel. Only one dynamic processing mode can be selected at a time for a selected band (i.e. Gain and Frequency mode cannot be enabled simultaneously for a selected band).

Gain Mode

Enables the ability to dynamically update the Gain of the selected EQ band when a signal exceeds the **threshold** level.

Clicking the “Gain” button in the EQ node controls panel will enable Dynamic Gain Mode for the selected node.

> **NOTE: DYNAMIC GAIN MODE AVAILABILITY**
>
> Dynamic Gain Mode is not available for **Lowpass** and **Highpass** filter shapes.

Boost or Cut Mode

When Gain mode is enabled, small triangle icons will appear directly above and below the associated EQ node. These icons indicate the expected direction of the dynamic gain change. Clicking on either triangle icon will determine the direction of gain change.

1. **BOOST**: Selecting the upward facing triangle will *increase* the gain of the filter when signals exceed the **threshold** level.
    1. When the EQ node is placed above the zero line, the filter will be set to 0 dB of gain until it is triggered. When it is triggered by incoming signals exceeding the threshold, the gain of the filter will be increased in the direction of the node.
    2. When the EQ node is placed below the zero line, the gain of the filter will be increased toward the zero line when it is triggered by incoming signals exceeding the threshold.
2. **CUT**: Selecting the downward facing triangle will *decrease* the gain of the filter when signals exceed the **threshold** level.
    1. When the EQ node is placed above the zero line, the gain of the filter will be reduced toward the zero line when triggered.
    2. When the EQ node is placed below the zero line, the filter will be set to 0 dB of gain until it is triggered. When it is triggered by incoming signals exceeding the threshold, the gain of the filter will be reduced in the direction of the node.

Threshold

Determines the signal level at which dynamic gain adjustments will be triggered for the selected EQ node.

> **TIP: USING DYNAMIC EQ TO REDUCE SIBILANCE**
>
> A dynamic EQ node can be used as an alternative to the **De-esser module** when attempting to reduce sibilant frequency content.

Frequency Mode

Enables the ability to dynamically update the Frequency of the selected EQ band to follow harmonics as they change.

Frequency mode works by identifying the fundamental frequency of the input signal when Frequency mode is enabled for an EQ band. It then identifies the harmonic frequency that is closest to the frequency of the selected EQ band. When the fundamental of the incoming vocal changes over time, the frequency of the EQ band will dynamically update to follow that harmonic as it changes with the fundamental. When Frequency mode is enabled and the frequency of the EQ band is manually adjusted, the EQ band will update to track the harmonic closest to the new frequency value.

★ **TIPS: WHEN TO USE DYNAMIC FREQUENCY MODE**

1. Use a Highpass filter with Dynamic Frequency Mode enabled to track the first harmonic of a vocal. This will dynamically cut out unwanted low end rumble while maintaining the character of the vocal.
2. Use multiple EQ bands with Dynamic Frequency Mode enabled to change the character of a vocal by following harmonics over time without the need for automation.
3. Use a Bell filter with Dynamic Frequency Mode enabled to follow and cut harsh vocal resonances.

Vocal Register

The Frequency mode is connected to the Vocal Register to track the input signal and apply the dynamic frequency mode. You can change the Vocal Register by accessing it in the Pitch Module Mini Module if you’re not getting the results you need.

Additionally, the EQ component plugin will toggle the Vocal Register in the component plugin’s header when Frequency mode is activated by the plugin. This will be available only when this mode is active.

Global Module Controls

The module chain features common controls for each module, including: Bypass, Solo, Remove, Reorder, and Wet/Dry Mix.

GLOBAL CONTROLS CHAPTER

To learn more about the module chain and other global controls in Nectar, visit the **Global Controls** chapter.

Meters

The following meters and displays are included in the EQ: **Spectrum Analyzer**, **Composite Curve**, and the **Filter Response Curve**.

Spectrum Analyzer

Displays the magnitude (amplitude, in decibels) of the input signal across the frequency spectrum in real-time. Two spectrum analyzers are displayed in the EQ module to compare the effect of processing. The input signal to the EQ module is shown as a dark grey spectrum with no border. The processed output spectrum is drawn in the foreground with a white border.

EQ Spectrum Rulers

1. The vertical ruler on the left side of the EQ module panel measures the amplitude of the spectrum.
2. The vertical ruler on the right side of the EQ module panel measures the gain of the EQ nodes.
3. The horizontal ruler along the bottom of the EQ module panel measures the frequency.

Composite Curve

The composite curve is the thick white line drawn across the EQ module panel. When EQ filters are added and modified they contribute to the overall shape of the composite curve. This curve represents the combined filter response of all enabled bands.

Filter Response Curve

The semi-transparent yellow fill color represents the filter response of a single band in isolation. This curve will be hidden by the white composite curve in cases where the selected filter shape is not affected by other enabled filters.

Gate

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Overview

A gate can be used to attenuate undesirable signal content in the breaks between phrases of a vocal track. The Gate module provides an efficient and flexible alternative to manually editing or inserting silence between vocal phrases. The Gate includes **controls** for adjusting how and when processing is applied to the input, as well as robust **metering** to assist with determining the best control values and visualizing the effect of processing.

Controls

The controls in the Gate module work together to influence how the gate responds to the incoming signal. The following controls are available in the Gate:

1. **Open and Close Threshold**
2. **Ratio**
3. **Attack**
4. **Release**

Threshold

The Gate module features two threshold controls: the **Open Threshold** and **Close Threshold**.

Open Threshold

Determines the level above which the gate will open, allowing the signal to pass through. When the input signal falls above the Open threshold level, attenuation will stop.

Close Threshold

Determines the level at which the gate will close. When the input signal falls below the close threshold, it will be attenuated.

Move the close threshold down in order to pass more of the decay without affecting the trigger threshold. In some situations, undesirable signals that are near the level of the open threshold can cause the gate to “chatter” by crossing the threshold level too often. The Close threshold (also referred to as hysteresis) can be set a few dB below the Open threshold to help eliminate this chattering effect.

When a signal has dropped below the Close threshold, it will not trigger the gate to open again until it exceeds the level of the Open threshold.

Ratio

Determines the amount of gain reduction applied to signals that fall below the Close threshold.

Attack

Determines the amount of time (in milliseconds) it takes for the gate to transition from closed to open when a signal exceeds the Open threshold.

Release

Determines the amount of time (in milliseconds) it takes to transition from open to closed when a signal falls below the Close threshold.

Global Module Controls

The module chain features common controls for each module, including: Bypass, Solo, Remove, Reorder, and Wet/Dry Mix.

GLOBAL CONTROLS CHAPTER

To learn more about the module chain and other global controls in Nectar, visit the **Global Controls** chapter.

Meters

The Gate module features scrolling waveform displays and a gain reduction meter for monitoring the difference between the processed and unprocessed signals.

Waveform Displays

The scrolling waveform meters display the amplitude of the input (ungated) and output (gated) signals over time. The meters scroll from right to left, with the most recent information on the right.

The **processed output signal** waveform is displayed in **light grey**, in front of the input signal waveform. The **unprocessed input signal** waveform is displayed in **dark grey**, behind the output signal waveform. When the signal is being gated, the gain reduction applied to the output signal can be observed in the difference between the two waveforms.

Gain Reduction Meter

Displays the current average amount of gain reduction in decibels (dB) applied to the signal.

Reverb

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Overview

Reverb can be used to add space and character to a vocal. The Nectar Reverb module is modeled after the classic EMT 140ST Stereo Plate Reverb with added controls for adjusting Pre-delay and Width not included in the EMT 140ST hardware unit.

Controls

The Reverb module controls panel includes parameters for customizing the timing, width, and character of the reverb.

Pre-Delay

Determines the amount of time the processed (wet) signal is delayed from the output.

The Pre-Delay control ranges from 0ms (milliseconds) to 200ms (milliseconds).

> **NOTE**
>
> The EMT 140ST did not include a Pre-Delay control, it is included in the Nectar Reverb module for added control of the plate reverb sound. Pre-Delay is helpful for better clarity and separation between the wet and dry vocal signals.

Decay

Determines the amount of time it takes for reflections to dissipate.

The Decay control ranges from 1.00s (seconds) to 5.00s (seconds).

> **NOTE**
>
> The Reverb module’s Decay time is not constant across all frequencies.

Width

Determines the amount of stereo widening applied to the output of the Reverb.

The Width control ranges from 0% (no stereo widening) to 100%.

Saturation

Determines the amount of subtle harmonic distortion added to the wet (processed) signal.

The Saturation control ranges from 0% (no saturation) to 100%.

Post Filter

The Reverb module features three customizable post filters. The Highpass and Lowpass filters can be used to reduce undesirable low frequency rumble or harsh high frequency content in the wet (processed) signal. The Bell filter can be used to add a creative or utilitarian boost or cut to the wet (processed) signal.

POST FILTER

1. The filters in the Reverb module are **only** applied to the wet (processed) output signal.
2. **Frequency Adjustments**: Click and drag a filter node left and right to adjust frequency.
3. **Gain Adjustments**: Click and drag a filter node up and down to adjust gain.
   1. The vertical ruler on the right side of the Reverb module panel display measures the gain of the Bell filter.
4. **Q (Bandwidth) Adjustments**: Select a filter node and drag the handles that appear towards the node (narrow Q) or away from the node (wide Q).

NOTE: Gain and Q can only be adjusted for the Bell filter. The Highpass and Lowpass filters have fixed slope values.

Global Module Controls

The module chain features common controls for each module, including: Bypass, Solo, Remove, Reorder, and Mix.

GLOBAL CONTROLS CHAPTER

To learn more about module chain and other global controls in Nectar, visit the **Global Controls** chapter.

Meters

The Reverb module features two spectrum analyzers for monitoring the effect of processing.

Spectrum Analyzer

Displays the magnitude (level, in dB) of a signal across the frequency spectrum in real-time. The vertical ruler on the left edge of the module panel measures the amplitude of the signal in decibels (dB). The horizontal ruler along the bottom edge of the module panel measures frequency in Hertz (Hz). The vertical ruler on the right edge of the module panel measures post-filter node gain (dB).

Two spectrum analyzers are displayed in the Reverb module: the input to the Reverb module (displayed in dark grey) and the output of the Reverb module (displayed in light grey).

Saturation

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Overview

Saturation can be used to add subtle warmth or intense distortion to a vocal track. The Saturation module features seven different **modes** for accentuating harmonics and adding unique color and character to a vocal.

Controls

The following parameters are available in the Saturation module panel:

1. **Mode**
2. **Amount**
3. **Post Filter**

Mode

Determines the type of saturation applied to the signal.

The following table outlines the sonic characteristics of the different saturation modes.

Mode	Characteristics
Analog	**Mild, gritty distortion.** Transistor type emulation with emphasis on odd harmonics.
Retro	**Sharp, aggressive distortion.** Based on a row of odd harmonics that are characteristic of transistors.
Tape	**Classic, analog tape saturation.** Emphasizes odd harmonics but with a shorter harmonic slope than that of a transistor-type saturation.
Tube	**Rich, tonal excitation.** Emulates tube saturation with a mix of odd and even harmonics.
Warm	**Subtle, harmonic warmth.** Generates even harmonics with a steep slope.
Decimate	**Unique, digital decimation.** Reduces sample rate to introduce distortion artifacts introduced by aliasing.
Distort	**Aggressive, dirty saturation.**

Amount

Determines the amount of drive applied to the selected saturation mode.

Post Filter

Allows for the reduction of undesirable high frequency content in the processed signal.

POST FILTER NOTES

1. The high shelf filter in the Saturation module is **only** applied to the wet (processed) output of the module.
2. Click and drag the filter node up and down to adjust gain.
3. Click and drag the filter node left and right to adjust frequency.

Global Module Controls

The module chain features common controls for each module, including Bypass, Solo, Remove, Reorder, and Wet/Dry Mix.

GLOBAL CONTROLS CHAPTER

To learn more about the module chain and other global controls in Nectar, visit the **Global Controls** chapter.

Meters

The Saturation module features two spectrum analyzers and a harmonic highlight display for monitoring the effect of processing.

Spectrum Analyzer

Displays the magnitude (level) of a signal across the frequency spectrum in real-time. The vertical ruler on the left edge of the module panel measures the amplitude of the signal in decibels (dB). The horizontal ruler along the bottom edge of the module panel measures frequency in Hertz (Hz).

Two spectrum analyzers are displayed in the Saturation module: the input to the Saturation module (displayed in dark grey) and the output of the Saturation module (displayed in light grey).

Harmonic Highlights

The harmonic content that is generated and added by the Saturation module is represented by the yellow fill color that appears when playing back audio.

Presets

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Overview

Choose from a wide variety of factory presets and save your own custom vocal processing chains in the Nectar Presets window. You can open the preset manager by clicking the button labeled **Presets** in the header area of Nectar.

TIP

You can quickly cycle through presets in the preset list by clicking the **previous** or **next** buttons to the right of the Presets button.

You can load, save, and manage presets in the Nectar Presets window. You can load the settings associated with a preset by selecting it from the list.

The Presets window is divided into two tabs:

1. **iZotope**: lists all factory presets installed with Nectar 4.
2. **Custom**: lists all custom presets you have saved or modified in Nectar 4.

There are two global options that appear in the **iZotope** and **Custom** tabs at the top of the preset list:

1. **Working Settings**: Loads your most recent changes that are not otherwise associated with a preset.
2. **Default**: Loads the factory default settings.

DIRTY STATE INDICATOR

When you make changes to a preset an asterisk (*) will be shown at the beginning of the preset name to indicate that it has been modified. You can add a new preset to save your settings or update the preset to dismiss the dirty state indicator.

Modify Custom Preset Names and Comments

You can modify preset file/folder names and preset comments when the Custom tab is selected.

1. **Edit custom preset name:** Single-click *twice* on a preset name in the **Custom** tab to open an inline edit field. Press return to dismiss the inline edit field and save your changes.
2. **Edit custom preset comment:** The area below the preset list displays descriptive text about the currently selected preset. Single-click the comment text box to open an inline edit field, press return to save changes to the comment.

Presets Window Controls

The following buttons are located in the footer of the Presets window:

1. Deletes the currently selected custom preset or preset folder.
2. **Update:** Saves changes to a modified custom preset.
3. **Folder**: Adds a new custom preset folder.
4. **New**: Creates a new preset based on the current settings.
5. **Close**: Dismiss the Presets window. *Note*: Double-clicking on a preset or preset folder will also close the Presets window.

ORGANIZE CUSTOM PRESETS

In the Custom tab, you can click and drag presets or folders over other folders in the list to move them into that folder.

Preset Locations

Factory presets are *installed* to the following locations:

1. **Windows:** `C:\Program Files\iZotope\Nectar Pro\Presets\`
2. **Mac:** `/Library/Application Support/iZotope/Nectar Pro/Presets/`

Custom presets are *saved* to the following locations:

1. **Windows:** `C:\Users\Username\iZotope\Nectar Pro\Presets\`
2. **Mac:** `/Users/Username/Documents/iZotope/Nectar Pro/Presets/`

Options

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Overview

The Options window includes parameters for managing authorization, updates, and module specific controls. The Options window can be opened by clicking on the gear button in the upper right corner of the Nectar interface.

The following categories are available in the Options window:

1. **General**
2. **EQ**
3. **Pitch & Voices**
4. **Licenses**

General

1. **Show tooltips:** Enables or disables the tooltip display text.
2. **Send anonymous usage data:** Enables sending anonymous usage data to iZotope to help improve Nectar.

EQ

Alt Solo Q: Sets the default bandwidth of the Alt-Solo Feature. The default is 5.02.

> **EQ: ALT-SOLO**
>
> See the **EQ chapter** for more information about the Alt-Solo feature.

Pitch and Voices

Calibration (A): sets the audio frequency reference to calibrate the Vocal Register and Scale in the Pitch Module. The default is 440 Hz.

Licenses

This section allows you to view your current license status, manage your license, and access more information about licensing the product.

> **ADDITIONAL HELP WITH LICENSING AND INSTALLATION**
>
> For additional information about licensing, authorization and installation, see the **Install and auth help** section of our Support Portal.

Apply Changes

To apply changes and close the Options window, click anywhere within the Nectar interface and outside of the Options window.

Any changes made to the Options window will be applied once the window is closed. You cannot cancel changes made, but you can reset to the factory default.

Reset

Press the Reset Button in the lower left-hand corner of the Global Options Window to reset any changes made within the Global Options window to the default setting.

TIP

The Reset button in the Options window will reset the plug-in window size to the factory default window size.

License Information

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Anti-Grain Geometry

Version 2.4

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base64

v0.4.0

Better Enums

Version 0.11.1

Bravura

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C++ Rest SDK

Version 2.10.15

Main Library:

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Bundled Libraries:

****** Base 64 Library (base64/base64.hpp) ******
base64.hpp is a repackaging of the base64.cpp and base64.h files into a single header suitable for use as a header only library. This conversion was done by Peter Thorson (**webmaster@zaphoyd.com**) in 2012. All modifications to the code are redistributed under the same license as the original, which is listed below.

base64.cpp and base64.h

René Nyffenegger **rene.nyffenegger@adp-gmbh.ch**

****** SHA1 Library (sha1/sha1.hpp) ******

sha1.hpp is a repackaging of the sha1.cpp and sha1.h files from the shallsha1 library (**http://code.google.com/p/smallsha1/**) into a single header suitable for use as a header only library. This conversion was done by Peter Thorson (**webmaster@zaphoyd.com**) in 2013. All modifications to the code are redistributed under the same license as the original, which is listed below.

****** MD5 Library (common/md5.hpp) ******
md5.hpp is a reformulation of the md5.h and md5.c code from **http://www.opensource.apple.com/source/cups/cups-59/cups/md5.c** to allow it to function as a component of a header only library. This conversion was done by Peter Thorson (**webmaster@zaphoyd.com**) in 2012 for the WebSocket++ project. The changes are released under the same license as the original (listed below)

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L. Peter Deutsch
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****** UTF8 Validation logic (utf8_validation.hpp) ******
utf8_validation.hpp is adapted from code originally written by Bjoern Hoehrmann **bjoern@hoehrmann.de**. See **http://bjoern.hoehrmann.de/utf-8/decoder/dfa/** for details.

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Eigen

Version 3.4.99

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FLAC

libFLAC and libFLAC++

Version 1.3.2

FreeType

Version 2.4.6

GLEW

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gsl

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Version 2019.0.5

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JsonCpp

Version 1.2.1

LibXML2

Version 2.7.8

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Material Docs Theme

NLohmann JSON

v3.10.4

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Netlib numeralgo na10 Aberth's method

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OGG / Vorbis

libogg and libvorbis

Version 1.3.2

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Roboto font family

Skia

TagLib

Version 1.9.1

TinyXML

Tipue Search

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vectorize

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xsimd

Yoga

Version 1.9.0

ZeroMQ

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