Effect Reference Guide

Introduction

This guide explains the parameters and technical terms that are used in the effects of Yamaha mixers and audio interfaces.
You should use this guide together with the documentation unique to the product.
Read the documentation first and use this reference guide to learn more about parameters and terms.

Information

Some functions and parameters in this guide may not be provided in your product.
The information in this guide is current as of January 2026.

EN

contents

EQ / Channel Section

PEQ (Parametric Equalizer)

A parametric equalizer (PEQ) is a high–precision equalizer that allows you to select specific frequency bands of an audio signal and freely adjust the gain (boost/cut) and bandwidth (Q) of those bands.
It is suitable for a wide range of applications, such as sound quality correction, acoustic space optimization, and feedback suppression.

Parameters

Parameter Name	Description	Example Range
Frequency	Sets the cutoff/center frequency to be boosted or cut according to the gain setting.	20.0 Hz–20.0 kHz
Gain	Sets the amount of boost/cut for the signal level near the frequency set by Frequency.	–18.0 dB–+18.0 dB
Q	Sets the bandwidth to be boosted/cut. The larger the value, the narrower the affected band.	16.0–0.1
Filter Type	Selects the filter type for each band (peak, shelving, HPF/LPF, etc.).	Peak, High Shelf, Low Shelf, HPF, LPF
Bypass	Disables PEQ processing for each band or the entire signal.	On, Off
1–Knob On/Off	Sets the 1–Knob Control On/Off.	On, Off
1–Knob Type	Sets the type of 1–Knob control.	Intensity, Vocal, Loudness
1–Knob Level	Sets the level of 1–Knob control.	0–100

Recommended Settings by Application

Application	Recommended Setting Example
Improve vocal clarity	2.5 kHz at +3 dB, Q = 1.2
Remove muddiness in low frequencies	200 Hz at –4 dB, Q = 1.0
Feedback suppression	6.3 kHz at –12 dB, Q = 10
Speaker correction	80 Hz at +2 dB (low shelf), 10 kHz at +3 dB (high shelf)

SSMCS (Sweet Spot Morphing Channel Strip)

SSMCS is a channel strip effect developed by Yamaha that combines a compressor and equalizer. Based on presets condensed from professional engineers' expertise, it features a "Morphing" function that allows multiple parameters to be changed simultaneously with a single knob, enabling easy and intuitive sound creation.

SSMCS consists of the following two main sections:

- **COMP (Compressor)**
  Controls volume peaks and adjusts dynamics. The Morphing knob allows simultaneous adjustment of multiple compressor parameters (Ratio, Attack, Release, Knee, etc.).
- **EQ (Equalizer)**
  Three–band structure (Low/Mid/High), with Frequency (F), Gain (G), and Q (Mid only) settings for each band. Low/High use shelf filters, Mid uses a peak filter. The Morphing knob allows simultaneous adjustment of multiple equalizer parameters (Frequency, Gain, Q).

Common Parameters for Compressor and Equalizer

Parameter Name	Description	Example Range
Morphing	Sets the intensity of the preset.	Continuous change between presets
Sweet Spot Data	For selecting the Sweet Spot Data.	Varies by setting

Call up Sweet Spot Data suitable for your application and adjust the Morphing parameter to find the optimal point.

Comp Section Parameters

Parameter Name	Description	Example Range
Comp Drive	Adjusts the degree to which the compressor is applied.	0.00–10.0
Knee	Sets how gently compression starts near the threshold.	Soft / Medium / Hard
Ratio	Sets the amount of compressor effect.	1.0:1–20:1–500:1, INF:1
Attack	Sets the time that elapses until the compressor effect starts.	0.092 ms–80.0 ms
Release	Sets the time that elapses until the compressor effect is released.	9.3 ms–999.0 ms
Side Chain	When set to On, the compressor is enabled to affect the side chain level detection.	On, Off

Parameter Name	Description	Example Range
SC–Q	This sets the width of the boost/cut band for the Side Chain that is turned on. Higher values affect narrower bands.	0.5–16
SC–Freq.	Sets the Side Chain cutoff frequency. This determines the frequency that is boosted/cut (increased/decreased) depending on the gain setting.	20 Hz–20 kHz
SC–Gain	Determines how much to boost/cut (increase/decrease) the signal level around the frequency set by Side Chain SC–Freq.	–18 dB–+18 dB

EQ Section Parameters

Parameter Name	Description	Example Range
Q	Sets the width of the band to be boosted/cut. Higher values affect narrower bands.	0.5–16.0
Frequency	Sets the cutoff frequency. This determines the frequency that will be boosted/cut (increased/decreased) depending on the gain setting.	20 Hz–20 kHz
Gain	Determines how much the signal level around the frequency set by the Frequency control will be boosted or cut.	–18 dB–+18 dB

Band	Frequency Range	Gain	Q	Filter Type
Low	20 Hz–1 kHz	–18 dB–+18 dB	None	Low Shelf
Mid	20 Hz–20 kHz	–18 dB–+18 dB	0.50–16.00	Peak
High	500 Hz–20 kHz	–18 dB–+18 dB	None	High Shelf

Dynamics Section

GATE

Gate is a dynamics effect that closes a gate (mutes the audio signal) when the input signal is below the set threshold, attenuating or cutting off the signal to remove unwanted noise and ambient sounds. It is particularly effective at cutting out quiet parts and ambient noise when using a microphone, improving the quality of your streaming and recordings.

Parameters

Parameter Name	Description	Example Range
Threshold	Sets the threshold level at which the gate effect is applied.	–72.0 dB–0.0 dB
Range	Sets the amount of gate closure (signal attenuation) when the input signal falls below the threshold.	–∞ dB–0.0 dB
Attack	Sets how quickly the gate opens after the input signal level exceeds the Threshold.	0.092 ms–80 ms
Hold	Sets the time that elapses before the gate begins to close after the input signal level drops below the Threshold.	0.02 ms–1960.0 ms
Decay	Sets how quickly the gate closes after the input signal has passed the Hold wait time.	9.3 ms–999.0 ms

Recommended Settings by Application

Application	Recommended Setting Example
Noise removal during vocal recording	Threshold: –45 dB / Range: –∞ dB / Attack: 10 ms / Hold: 300 ms Decay: 200 ms
Ambient noise cut during live streaming	Threshold: –50 dB / Range: –12 dB / Attack: 5 ms / Hold: 500 ms Decay: 150 ms
Silence processing during instrument recording	Threshold: –40 dB / Range: –6 dB / Attack: 20 ms / Hold: 250 ms Decay: 100 ms

COMP (Compressor)

Compressor is a dynamics effect that compresses the parts of the input signal that exceed a set threshold, smoothing out volume changes. By suppressing peaks, it stabilizes the overall volume balance for a professional finish.

Parameters

Parameter Name	Description	Example Range
Threshold	Sets the threshold level for the compressor effect.	–54.0 dB–0.0 dB
Ratio	Sets the amount of compressor effect.	1.0:1–20:1–500:1, INF:1
Gain	Sets the output level of the compressor.	0.0–18.0 dB
Auto Makeup	Automatically compensates for gain loss due to compression.	On, Off
Attack	Sets the speed at which the compressor effect reaches maximum after the input signal exceeds the threshold.	0.092 ms–80.00 ms
Release	Sets the time that elapses until the compressor effect stops after the input signal falls below the threshold.	9.3–999.0 ms
Knee	Sets how gently compression starts near the threshold. Soft: natural volume change. Medium: intermediate volume change. Hard: clear, abrupt volume change.	Soft / Medium / Hard

Recommended Settings by Application

Application	Recommended Setting Example
Vocal	Threshold: –40 dB / Ratio: 3:1 / Attack: 20 ms / Release: 80 ms / Knee: Medium
Guitar	Threshold: –35 dB / Ratio: 2:1 / Attack: 10 ms / Release: 100 ms / Knee: Soft
Drums	Threshold: –30 dB / Ratio: 5:1 / Attack: 5 ms / Release: 150 ms / Knee: Hard

COMPANDER–H / COMPANDER–S

COMPANDER–H / COMPANDER–S are composite dynamics effects combining compressor, expander, and limiter. They suppress volume changes while simultaneously reducing ambient noise, providing stable sound quality for streaming, recording, and live performance.

COMPANDER–H	Large noise attenuation, excellent for removing ambient sounds. Effective for suppressing noise in quiet scenes.
COMPANDER–S	Moderate noise attenuation, processes noise slightly while maintaining natural sound quality. Suitable for situations emphasizing musical nuance.

Parameters

Parameter Name	Description	Example Range
Threshold	Sets the threshold level for the compressor effect.	–54.0 dB–0.0 dB
Ratio	Sets the amount of compressor effect.	1.0:1–20:1
Attack	Sets the time that elapses until the set level is reached after the input signal exceeds three boundary levels.	0–120 ms
Release	Sets the time that elapses until the set level is released after the input signal exceeds three boundary levels.	5.0 ms–42.3 s
OutGain	Sets the output level.	0.0 dB––18 dB
Width	Sets the width of the boundary levels between compressor and expander.	1–90 dB

Recommended Settings by Application

Application	Type	Recommended Setting Example
Vocal recording	COMPANDER–S	Threshold: –40 dB / Ratio: 2:1 / Attack: 20 ms / Release: 80 ms / OutGain: –6 dB
Streaming with emphasis on noise suppression	COMPANDER–H	Threshold: –50 dB / Ratio: 5:1 / Attack: 10 ms / Release: 120 ms / OutGain: –12 dB
Instrument recording (acoustic)	COMPANDER–S	Threshold: –35 dB / Ratio: 1.5:1 / Attack: 40 ms / Release: 100 ms / OutGain: –3 dB

DUCKER

A ducker is an effect that detects a trigger input signal and automatically attenuates (reduces) the volume. It is mainly used to automatically lower background music when narration or vocals are present, achieving a natural mix balance without manual fader operation.

Parameters

Parameter Name	Description	Example Range
Ducker Source	Sets the signal used to determine the strength of the ducker.	Any input channel
Threshold	Sets the threshold level at which the ducker effect is applied.	–60 dB–0 dB
Range	Sets the amount of attenuation when the ducker effect is applied.	–70 dB–0 dB –∞ dB–0.0 dB
Attack	Sets how quickly the volume lowers after the input signal level exceeds the “Threshould”.	0.092 ms–80.00 ms
Decay	Sets how quickly the volume returns after the input signal level falls below the “Threshould”.	1.3 ms–5.0 s 9.3 ms–999.0 ms

Recommended Settings by Application

Application	Recommended Setting Example
Background music with narration	Source: Mic IN / Threshold: –40 dB / Range: –12 dB Attack: 10 ms / Decay: 300 ms
Automatic mute for conference audio	Source: Voice IN / Threshold: –50 dB / Range: –∞ dB Attack: 5 ms / Decay: 500 ms
Prioritize talk during live performance	Source: Vocal Mic / Threshold: –45 dB / Range: –6 dB Attack: 20 ms / Decay: 200 ms

M.B.COMP (Multi-band Compressor)

A multi–band compressor divides the audio signal into three frequency bands (LOW, MID, and HIGH) and applies independent compression to each, enabling more precise dynamics control. By suppressing volume changes in each band, it improves level stability during streaming and recording, resulting in a balanced sound.

Parameters

Parameter Name	Description	Example Range
1–knob On/Off	This function controls the effect of Multi–Band Compressor with a single slider. When this is set to On, Threshold, Ratio, and Gain for each band are adjusted according to the 1–knob Level. ATTACK, RELEASE and XOVER frequencies are fixed values.	On, Off
1–Knob Level	Sets the level of 1–Knob control.	0–48
Gain	Sets the volume for the target band.	–∞, –60 dB–+18 dB
L–M XOVER	Sets the crossover frequency between the LOW band and MID band.	21.2 Hz–4.00 kHz
M–H XOVER	Sets the crossover frequency between the MID band and HIGH band.	42.5 Hz–8.00 kHz
Bypass	Enables/disables bypass for the compressor of the target band.	On, Off
Attack Time	Sets the attack time for the compressor of the target band. This adjusts the time it takes for the compressor to reach its maximum effect after the input signal exceeds the threshold.	1 ms–200 ms
Release Time	Sets the release time for the compressor (all bands). This sets the time it takes for the compressor effect to stop after the input signal falls below the threshold.	10 ms–3000 ms
Threshold	Sets the threshold for the compressor of the target band. This sets the level at which the compressor effect is applied.	–54 dB––6 dB
Ratio	Sets the ratio for the compressor of the target band. This sets the amount of compression applied.	1.0:1–20.0:1
Out Gain	Sets the overall output level.	–12 dB–+12 dB

Recommended Settings by Application

Application	Recommended Setting Example
Live streaming	LOW: Threshold –30 dB / Ratio 3:1 / Gain +2 dB MID: Threshold –25 dB / Ratio 2.5:1 / Gain +1 dB HIGH: Threshold –20 dB / Ratio 2:1 / Gain 0 dB
Vocal processing	LOW: Light compression / MID: Emphasis on clarity / HIGH: Suppress harsh sibilance
Mastering	Apply even compression to all bands to stabilize loudness

Pitch / Vocal Processing Section

PITCH FIX

PITCH FIX is an effect that corrects the pitch of input audio in real time. It automatically adjusts the pitch of vocals or narration, resulting in stable phrasing. Additionally, flexible settings are available, such as formant (voice quality) adjustment and specifying the target notes for correction via MIDI control.

Combines the following two main processes for pitch correction:
Manual pitch correction: Corrects pitch and formant to manually set values.
Automatic pitch correction: Automatically corrects input audio pitch to match the specified key and scale.

Parameters

Parameter Name	Description	Example Range
Coarse	Sets pitch deviation in semitone units.	–12–+12
Fine	Sets pitch deviation in cent units.	–50–+50
Formant	Sets the formant (voice quality).	–62–+62
Mix	Sets the volume balance before and after pitch correction.	0–126
Key	Selects the key for automatic correction.	C–B
Scale	Selects the scale for automatic correction.	Chromatic, Major, Minor, Pentatonic, etc.
Note Low Limit	Sets the lower limit of the range for automatic correction.	C–2–G8
Note High Limit	Sets the upper limit of the range for automatic correction.	C–2–G8
Correction	Enables/disables automatic correction.	On, Off
MIDI Control	Selects the mode for MIDI control.	Off, Setting, Real Time
Speed	Sets the speed of correction tracking.	0–100
Tolerance	Sets sensitivity to pitch changes.	0–100

Recommended Settings by Application

Application	Recommended Setting Example
Natural vocal correction	Key: C / Scale: Major / Speed: 60 / Tolerance: 40 / Mix: 100
Robot voice style	Key: C / Scale: Chromatic / Speed: 100 / Tolerance: 0 / Formant: +40
Stabilize narration	Correction: On / Speed: 30 / Tolerance: 60 / Note Limit: C2–C5

Amp / Guitar Section

GUITAR AMP CLASSICS

Guitar Amp Classics are guitar amp simulations that make extensive use of advanced Yamaha modeling technology. Four types with different sound characters are available: CLEAN, CRUNCH, LEAD, and DRIVE, suitable for a wide range of genres and playing styles.

CLEAN	Clear sound characteristic of transistor amps. Ideal for clean tones.
CRUNCH	Lightly distorted sound for blues and rock. Vintage tube amp style.
LEAD	High–gain, rich harmonic tube amp sound. Ideal for lead guitar.
DRIVE	Strongly distorted sound for hard rock/metal. Wide range of distortion expression.

Parameters

Common Parameters for CLEAN/CRUNCH/DRIVE/LEAD

Parameter Name	Description	Example Range
Treble, Middle, Bass	Adjusts the level of high/mid/low frequencies.	0–10
Presence	Emphasizes high–frequency overtones.	0–10
Off / Gate	Switches the noise gate On/Off.	On, Off
Gate level	Adjusts the threshold for the gate.	0–10
SP Type	Selects the SP TYPE. For details, see “Cabinet Types and Features.”	0–8
Mic Position	Sets the microphone position.	Center or Edge
Output	Adjusts the final output level.	–∞ dB–0 dB

CLEAN Only

Parameter Name	Description	Example Range
Volume	Adjusts input level.	0–10
Distortion	Adjusts amount of distortion.	0–10
Cho/OFF/Vib:	Switches effect On/Off. “Cho” enables chorus, while “Vib” enables vibrato.	
Speed/Depth	Sets vibrato speed/depth when “Vib” is On.	0–10
Blend	Adjusts balance between original and effect sound (CLEAN only).	0.0–10.0

CRUNCH Only

Parameter Name	Description	Example Range
Normal/Bright	Switches sound character.	
Gain	Sets input level. Higher values increase distortion.	0–10

LEAD Only

Parameter Name	Description	Example Range
High/Low	Switches sound character.	
Gain	Sets input level. Higher values increase distortion.	0–10
Master	Sets preamp output level.	0–10

DRIVE Only

Parameter Name	Description	Example Range
Amp Type	Switches sound character.	1–6
Gain	Sets input level. Higher values increase distortion.	0–10
Master	Sets preamp output level.	0–10

Cabinet types and characteristics

The following table shows the cabinet characteristics that are common to each of the four types: CLEAN, CRUNCH, DRIVE, and LEAD.

SP TYPE	Characteristics	Speaker configuration
BS 4×12	British flat stack type with rich cabinet resonance.	4×12"
AC 2×12	American combo type cabinet, featuring a clear tone for versatile use in various music genres.	2×12"
AC 1×12	American combo type cabinet, featuring a clear tone for ensemble use.	1×12"
AC 4×10	American combo type cabinet, featuring a bright tone reminiscent of more traditional guitar sounds.	4×10"
BC 2×12	British combo type cabinet, ideal for distortion sounds and featuring a wide range with broad treble response.	2×12"
AM 4×12	American stack type cabinet, ideal for matching with high–power amplifiers and featuring a clear sound contour.	4×12"
YC 4×12	Yamaha F series combo type cabinet, featuring a rich midrange and a mild high range.	4×12"
JC 2×12	Japanese combo type cabinet, ideal for clean sounds, and featuring a rich mid–high range plus modulation effects.	2×12"

Recommended Settings by Application

Application	Type	Recommended Setting Example
Clean backing	CLEAN	DISTORTION: 0 / BLEND: 50 / TREBLE: 6 / BASS: 5
Blues solo	CRUNCH	GAIN: 5 / TREBLE: 7 / MIDDLE: 6 / BASS: 5
Rock lead	LEAD	GAIN: 8 / MASTER: 7 / PRESENCE: 6 / SP TYPE: AC 4×10
Metal riff	DRIVE	GAIN: 10 / TREBLE: 8 / BASS: 6 / GATE LEVEL: 7

Reverb Section

REV–X

REV–X is a high–quality digital reverb effect developed by Yamaha for professional audio equipment. Inheriting the legacy of the classic REV5 and REV7 models, it is designed as an algorithm and widely adopted in DSP–equipped devices and VST plugins. It excels at reproducing a sense of air and is loved by engineers worldwide. You can select from three types: REV–X HALL, REV–X ROOM, and REV–X PLATE.

REV–X HALL	Reproduces the spaciousness and depth of a concert hall. Creates a grand and rich spatial feel.
REV–X ROOM	Simulates the natural reverberation of a studio or small room. Adds thickness to the space while maintaining instrument and vocal localization.
REV–X PLATE	Simulates classic plate reverb using metal plates. Features clear and smooth reverberation, ideal for vocal processing.

Parameters

Parameter Name	Description	Example Range
Reverb Time	Adjusts the reverb time. This parameter links to Room Size. The adjustable range varies depending on the REV–X type.	Hall: 0.103 sec–31.0 sec Room: 0.152 sec–45.3 sec Plate: 0.176 sec–52.0 sec *Minimum value when Room Size is 0 (min) to Maximum value when Room Size is 31 (max)
Initial Delay	Adjusts the time that elapses between the direct, original sound and the initial reflections that follow it.	0.1 msec–200.0 msec
Decay	Adjusts the characteristic of the envelope from the moment the reverberation starts to the moment it attenuates and stops.	0–63
Room Size	Adjusts the size of the simulated room. This parameter links to Reverb Time.	0–31
Diffusion	Adjusts the spread of the reverberation.	0–10
HPF	Adjusts the cutoff frequency of the high pass filter.	20 Hz–8 kHz
LPF	Adjusts the cutoff frequency of the low pass filter.	1 kHz–20 kHz
Hi Ratio	Sets the length of high–frequency reverberation.	0.1–1.0

Parameter Name	Description	Example Range
Low Ratio	Sets the length of low–frequency reverberation.	0.1–1.4
Low Freq	Adjusts the frequency of the Low Ratio.	22 Hz–18 kHz

Recommended Settings by Application

Application	Type	Recommended Setting Example
Vocal	Plate	Reverb Time: 1.8 s / High Ratio: High
Acoustic guitar	Plate	Reverb Time: 1.8 s / High Ratio: High
Drums	Room	Room Size: Small / Initial Delay: 20 ms
Strings	Hall	Reverb Time: 3.0 s / Diffusion: High

REV R3

REV R3 is a next–generation reverb effect that inherits the technology of the REV–X series, enabling more flexible parameter settings and high–precision spatial expression. It features three types: Hall, Room, and Plate. Widely used in professional audio equipment and VST plugins, it offers natural, smooth reverberation and high spatial expressiveness for music production and live sound.

REV R3 HALL	Reproduces the spaciousness and depth of a concert hall. Creates a grand and rich spatial feel.
REV R3 ROOM	Simulates the natural reverberation of a studio or small room. Adds thickness to the space while maintaining instrument and vocal localization.
REV R3 PLATE	Simulates classic plate reverb using metal plates. Features clear and smooth reverberation, ideal for vocal processing.

Parameters

Parameter Name	Description	Example Range
Reverb Time	Adjusts the length of the reverberation.	0.3s–30.0s
Initial Delay	Adjusts the time that elapses before reverberation begins.	0.1 msec–200.0 msec
Hi Ratio	Sets the ratio of high–frequency reverberation time to REV TIME.	0.1–1.0
Diffusion	Adjusts the density and spread of the reverberation.	0–10
Density	Sets the density of the reverberation.	0–4
HPF	Sets the cutoff frequency for the high–pass filter.	Thru, 21.2 Hz–8.00 kHz
LPF	Sets the cutoff frequency for the low–pass filter.	50.0 Hz–16.0 kHz, Thru
ER/ Reverb Delay	Sets the delay time from early reflections to reverb sound.	0.1 ms–200 ms
ER/Rev Balance	Sets the level balance between early reflections and reverb sound.	E63>R–E<R63
Feedback Gain	Sets the amount of feedback for the initial delay.	–99%–+99%

Recommended Settings by Application

Application	Type	Recommended Setting Example
Vocal	Plate	Reverb Time: 1.5 s / Diffusion: 7 / High Ratio: 0.8 / LPF: 12 kHz
Acoustic guitar	Room	Reverb Time: 1.2 s / Density: 3 / HPF: 100 Hz / LPF: 10 kHz
Drums	Hall	Reverb Time: 2.5 s / Initial Delay: 50 ms / Diffusion: 10 / ER/Rev Balance: E=R
Cinematic (movie style)	Hall	Reverb Time: 4.0 s / Density: 4 / High Ratio: 1.0 / LPF: 16 kHz

Delay Section

MONO DELAY

Mono Delay is a simple delay effect that adds a delayed signal to the input, creating spatial width and rhythmic feel. Its monaural configuration makes it easy to use for vocals, guitar, synth, and more. It supports tempo synchronization, allowing delay time settings to match the song's BPM.

Parameters

Parameter Name	Description	Example Range
Delay	Sets the delay time.	0.1ms–2700.0ms 1.0ms–1350.0ms
Feedback Gain	Sets the amount of repeated delayed sound.	–99%–+99%
High Ratio	Sets the amount of high–frequency feedback.	0.1–1.0
HPF	Sets the cutoff frequency for the high–pass filter.	Thr, 21.2 Hz–8 kHz
LPF	Sets the cutoff frequency for the low–pass filter.	50 Hz–16 kHz, Thru
Sync	Enables/disables tempo synchronization.	On, Off
Note	Sets the delay in note units when tempo sync is enabled.	*1
BPM	Sets the tempo.	25–300

*1 Note values are calculated as follows, with the maximum value depending on the tempo setting.

= 1/48 = 1/24 = 1/16 = 1/12 = 3/32 = 1/8 = 1/6

= 3/16 = 1/4 = 3/8 = 1/2 = 3/4 = 1/1 = 2/1

Recommended Settings by Application

Application	Recommended Setting Example
Vocal	Delay: 240 ms / Feedback Gain: 15 / LPF: 8.5 kHz / HPF: 150 Hz
Guitar	Delay: 500 ms / Feedback Gain: 20 / LPF: 10 kHz / HPF: 100 Hz
Synthesizer	Sync: On / Note: ♩Quarter note / High Ratio: 0.6

PING PONG DELAY

Ping Pong Delay is a stereo delay effect where the delayed sound alternates between the left and right channels. It creates spatial width and movement, adding rhythmic depth to vocals, synths, and guitars.

Parameters

Parameter Name	Description	Example Range
Delay Time	Sets the delay time in milliseconds. When tempo sync is enabled, this depends on the Note and BPM settings.	1.0 ms–1350.0 ms
Feedback Gain	Sets the amount of feedback.	–99%–+99%
High Ratio	Sets the amount of high–frequency feedback.	0.1–1.0
HPF	Sets the cutoff frequency for the high–pass filter.	Thru, 21.2 Hz–8.00 kHz
LPF	Sets the cutoff frequency for the low–pass filter.	50.0 Hz–16.0 kHz, Thru
Sync	Enables/disables tempo synchronization.	On, Off
Note	Sets the delay in note units when tempo sync is enabled.	*1
BPM	Sets the tempo.	25–300

*1 Note values are calculated as follows, with the maximum value depending on the tempo setting.

= 1/48	= 1/24	= 1/16	= 1/12	= 3/32	= 1/8	= 1/6
= 3/16	= 1/4	= 3/8	= 1/2	= 3/4	= 1/1	= 2/1

Recommended Settings by Application

Application	Recommended Setting Example
Vocal	Delay: 240 ms / Feedback Gain: 15 / LPF: 8.5 kHz / HPF: 150 Hz
Synthesizer	Sync: On / Note: ♩Quarter note / High Ratio: 0.6
Guitar	Delay: 500 ms / Feedback Gain: 20 / LPF: 10 kHz / HPF: 100 Hz

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