YAMAHA

mLAN Mixer Owner's Manual

Table of Contents

This owner's manual assumes that you are already familiar with basic Windows/Macintosh operation. If you are not, please refer to the owner's manual which came with your Windows/Mac OS software before using mLAN Mixer.
For information about hardware requirements, the interconnection of devices and the installation of the mLAN Mixer software, refer to the separate Installation Guide as well as the owner's manual for the respective mLAN device.
This owner's manual is applicable to the mLAN Mixer for both Windows and Macintosh. The screen illustrations are mainly taken from the mLAN Mixer for Macintosh.

What is mLAN Mixer?

mLAN Mixer is a software application that enables you to control the mixer functions of the mLAN8P/mLAN8E from the computer as if you were controlling a normal mixing console. mLAN Mixer offers independent EQ and dynamics settings for each channel, making it possible to perform detailed mixing operation.

Starting mLAN Mixer

After installing mLAN Mixer and making the necessary connections, follow the steps below to start the software.

NOTE Use mLAN Patchbay to set up the COM port for Windows.

Windows98/95 environment

In Windows98/95, select [Start | Program (P) | YAMAHA mLAN Mixer], then select "mLAN Mixer."

Macintosh environment

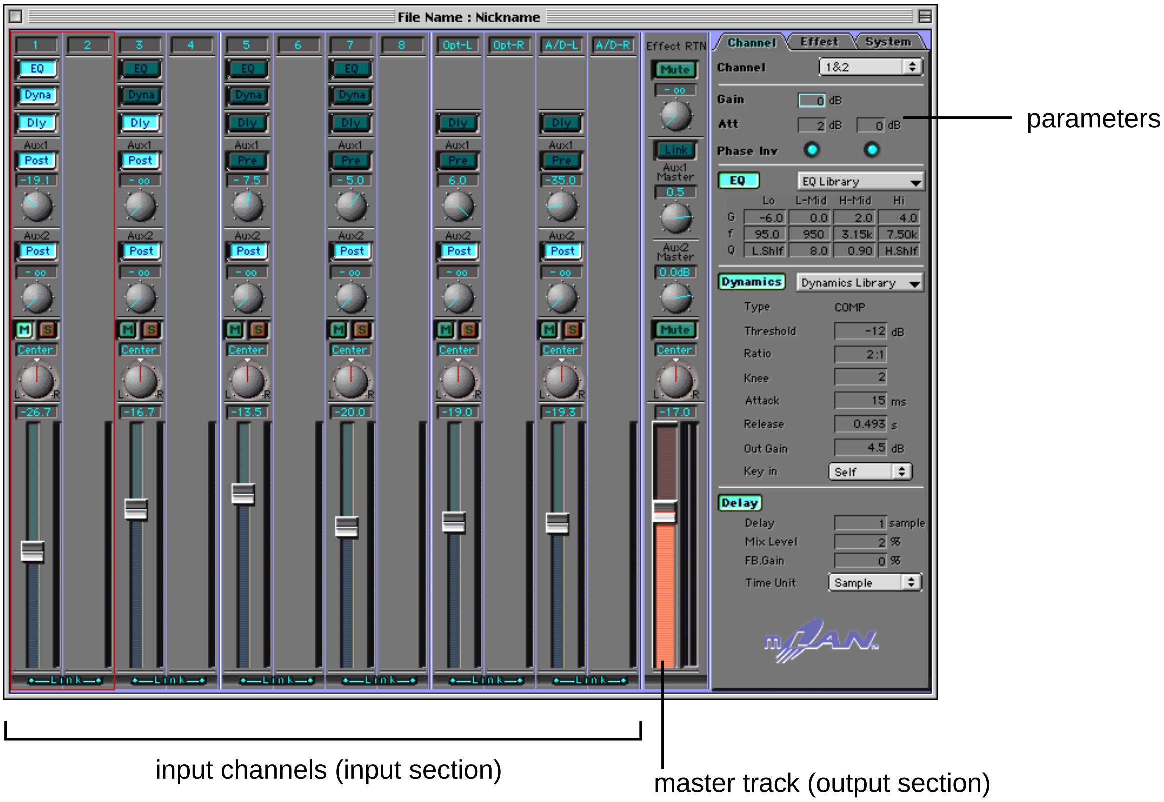
Open the "YAMAHA mLAN Mixer" folder on the computer and double-click the "mLAN Mixer" icon.

Top Panel Screen

The following screen appears when you start mLAN Mixer.

NOTE If there are multiple mLAN8P/mLAN8Es connected to the bus (system) of a Macintosh computer, a window appears, prompting you to select the device you wish to control with the mLAN Mixer. You can select multiple devices to control simultaneously from the mLAN Mixer. If you are using a PC that runs Windows98/95, an mLAN8P/mLAN8E connected serially to the computer will be automatically selected.

NOTE The mLAN8P controlling screen and the mLAN8E controlling screen are slightly different. This manual references the mLAN8P controlling screen. However, functions that differ are also explained.



Menu Bar

File menu

New: .... Creates a new Mixer file.
Open: .... Opens an existing Mixer file.
Save: .... Overwrites an existing file with the currently-edited Mixer file.
Save As: .... Saves the currently-edited Mixer file with a different name.
Quit: .... Closes mLAN Mixer.

NOTE Save and Save As do not save the Level, Meter Source, Peak Hold, Fall Time, and Word Clock settings.

NOTE The edited settings for the mLAN mixer will be lost when the power to the mLAN8P/mLAN8E is turned off. (This does not apply to the parameters accessible from the control panel of the unit.) Be sure to store the necessary settings using the File menu.

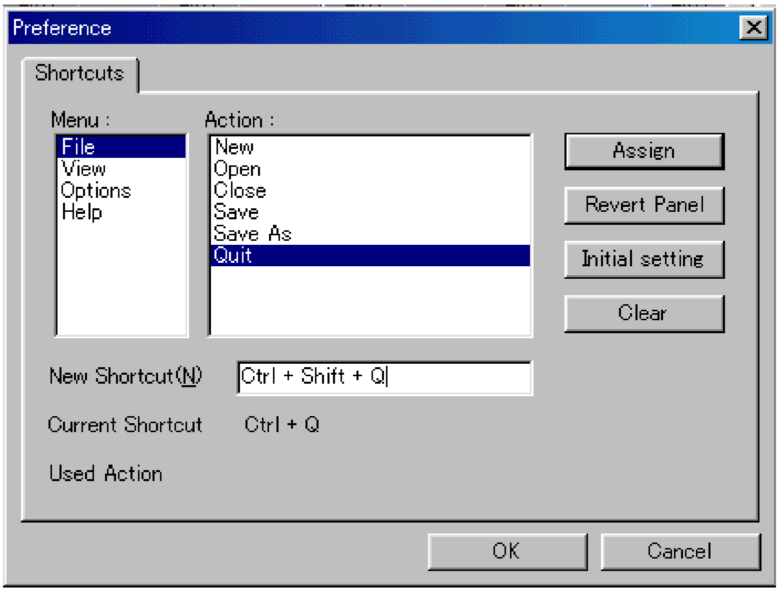
View menu

Hide/Show Parameter: .... Displays or hides various parameters in the tab screens.

Option

Init Channel (Initialize Channel): ...... Opens the Init Channel dialog that is used to initialize the settings for selected channels or all channels.
Preference (Windows only): .... Opens the Preference dialog that is used to customize the menu shortcut keys.

Preference dialog (Setting the shortcuts) (Windows only)



Shortcuts

1. Select a menu and action from the Menu and Action columns for which you wish to set a shortcut. In the example above, [New] in the File menu is selected.
2. Click [New Shortcuts] and press the desired shortcut key. In the example above, the [Q] key is pressed while the [Ctrl] and [Shift] keys are held down. If the shortcut key has already been assigned to another "Action," the "Used Action" column displays the name of the "Action."
3. Click [Assign]. The "Current Shortcut" column displays the selected shortcut.
4. Press the [OK] button to confirm the selected shortcut setting. If you do not wish to change the setting, click [Cancel].

   Revert Panel: .... Returns to the previous setting.
   Initial setting:........ The shortcut is set to the [Ctrl] key plus the first letter of the selected Action.
   Clear:...................... Clears the setting.

Direct Mode: .... You can temporarily place mLAN Mixer in Direct mode while you are controlling the mLAN8E. This function is not available with the mLAN8P.

NOTE Refer to the Owner's manual for the mLAN8E for more information on Direct mode.

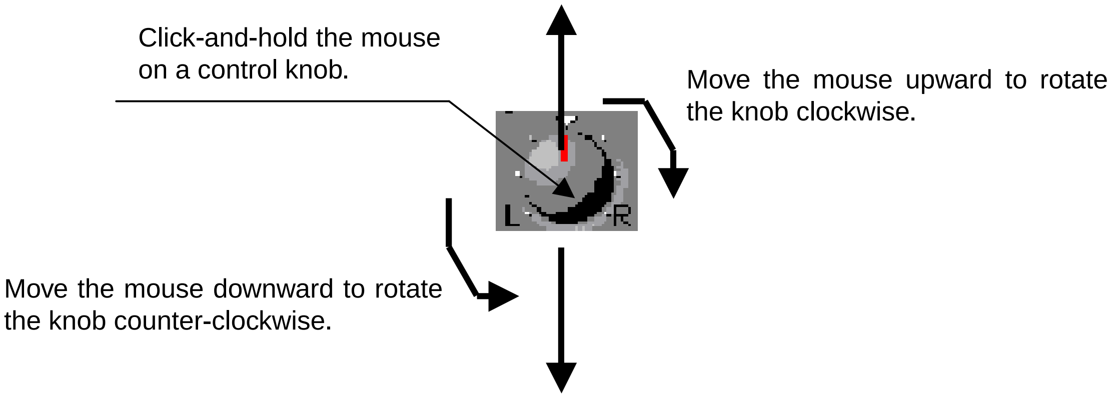
Help menu

About mLAN Mixer (A): .... Displays the version information for the mLAN Mixer (Windows).

Basic Controls

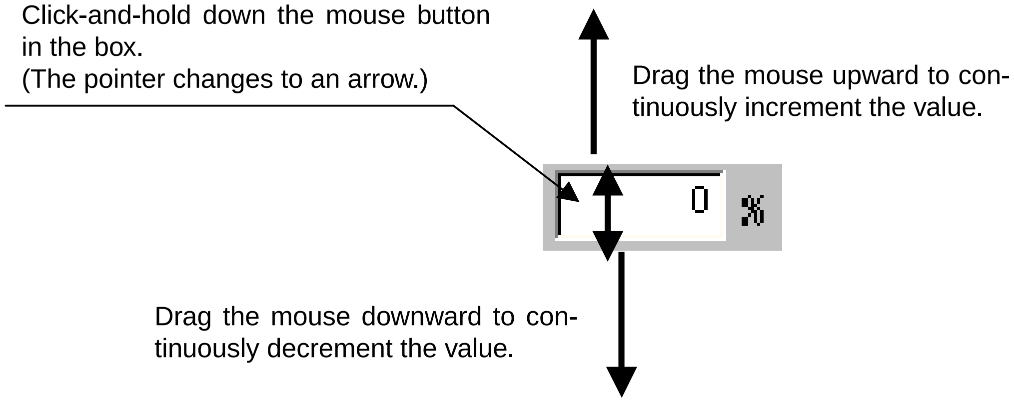
Control knobs

To rotate a control knob clockwise, click-and-hold the mouse on the knob while dragging upward. Drag down to rotate the knob counter-clockwise.



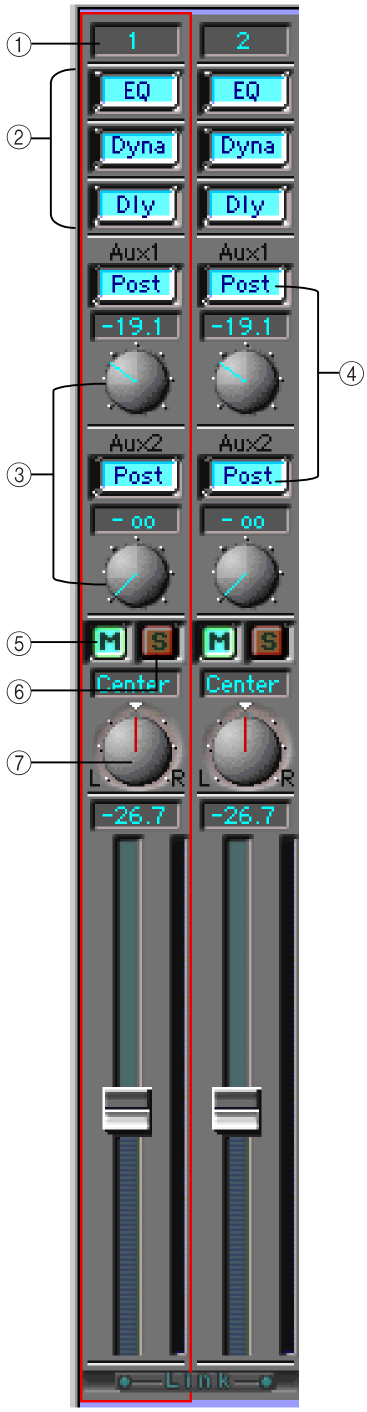
Numeric field box

Click-and-hold the mouse and drag upward in a number box to continuously increment the value. Drag the mouse downward to decrement the value.
You can also enter the value directly by clicking (double-clicking) the box.



Input Channel Settings (Input Section)

You can adjust the input signal level and effect on/off for each channel.



① Channel number

Channels 1-8 are used for input signals via mLAN. Channels 9 and 10 are used for Digital input signals, and channels 11 and 12 are used for A/D input signals.

② EQ/Dyna (Dynamics)/Dly (Delay) button

These buttons are used to switch EQ, Dynamics, and Delay on and off for each channel. The settings are linked with the buttons in the Channel tab.

③ Aux 1/2 send knobs

These knobs are used to adjust the level of each channel signal sent to AUX 1/2.
Use these knobs when you are sending signals to an external effect processor or when you are monitoring signals with a balance that is different from the stereo output balance.
If you are using the mLAN8P: Signals sent to AUX2 are also sent to the internal multi-effect processor.

NOTE When AUX Layer is turned on, you can set AUX 3-6 parameters (mLAN8E).

④ Pre/Post buttons

These buttons enable you to select a pre-fader (Pre) signal or a post-fader (Post) signal as the AUX send signal. With the "Pre" setting, a signal is sent to the AUX bus before it reaches the channel fader; thus the signal is not affected by the channel fader setting. With the "Post" setting, a signal is sent to the AUX bus after it passes through the channel fader; thus the signal is affected by the channel fader.

⑤ M (Mute) button

Press this button to mute the corresponding channel signal.

⑥ S (Solo) button

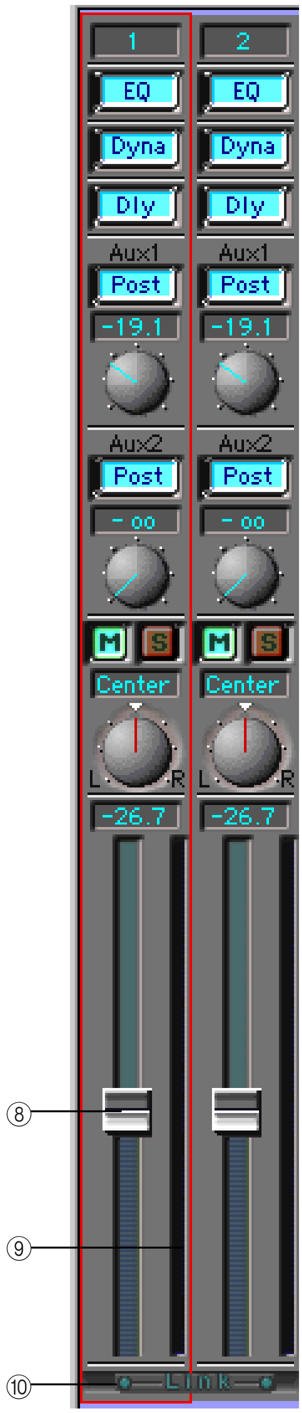
Press this button to monitor the corresponding channel signal.
Press this button while holding down the Shift key to monitor multiple channel signals.

⑦ Panpot knob

This knob enables you to set the stereo position of the corresponding channel. If the channel's Link switch is set to "ON," the knob adjusts the signal balance.

NOTE You can set the attenuation and phase independently for each channel, even when the Link switch is turned on.

NOTE When adjacent channels are linked, the panpot knob enables you to adjust the level balance between the odd and even channels.



⑧ Channel fader

The channel fader enables you to adjust the channel level. Click and hold down the mouse button on the fader and slide the mouse up and down, or click a desired point on the fader to move the fader to that point.

⑨ Channel level meter

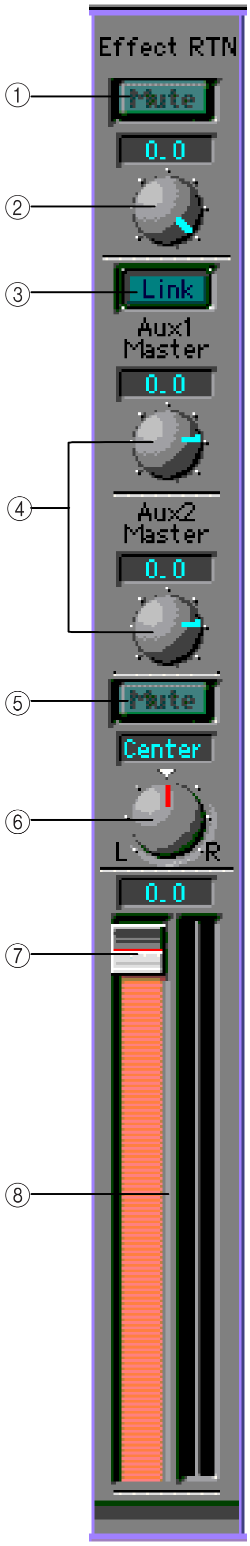
The channel level meter is displayed on the right side of the channel fader. The channel meter source is always the signal just after the gain, so the channel level meter displays the level of the input signal and is not affected by the EQ, Dynamics or the fader.

⑩ Link (Link switch)

When this switch is turned on, adjacent channels (1 and 2, 3 and 4, etc.) are linked to each other for stereo operation.

Master Track Settings (Output Section)

In this section, you can set the stereo out fader, AUX master level, and effect return level.



① Mute button (mLAN8P only)

Press this button to mute the return signal from the internal multi-effect processor.

② Effect RTN (mLAN8P only)

This knob adjusts the amount of signal processed by the effect processor and routed to Stereo Out.

NOTE The mLAN8E features an AUX Layer button instead of this button. When you turn this button on, you can set AUX 3-6 parameters for both input channels and the master track.

③ Link button

Set this button to ON to link the AUX 1 and 2 settings. When they are linked, the panpot setting of the corresponding input channel is used. This is useful when you wish to use AUX 1 and 2 as individual stereo outputs.

④ AUX 1/2 Master

These are used to adjust the level of the AUX 1 and 2 return signals.

NOTE When AUX Layer is turned on, you can set AUX 3-6 parameters (mLAN8E).

⑤ Mute button

Press this button to mute the stereo mix output.

⑥ Balance

This knob adjusts the left and right balance of the stereo mix output.

⑦ Stereo mix fader

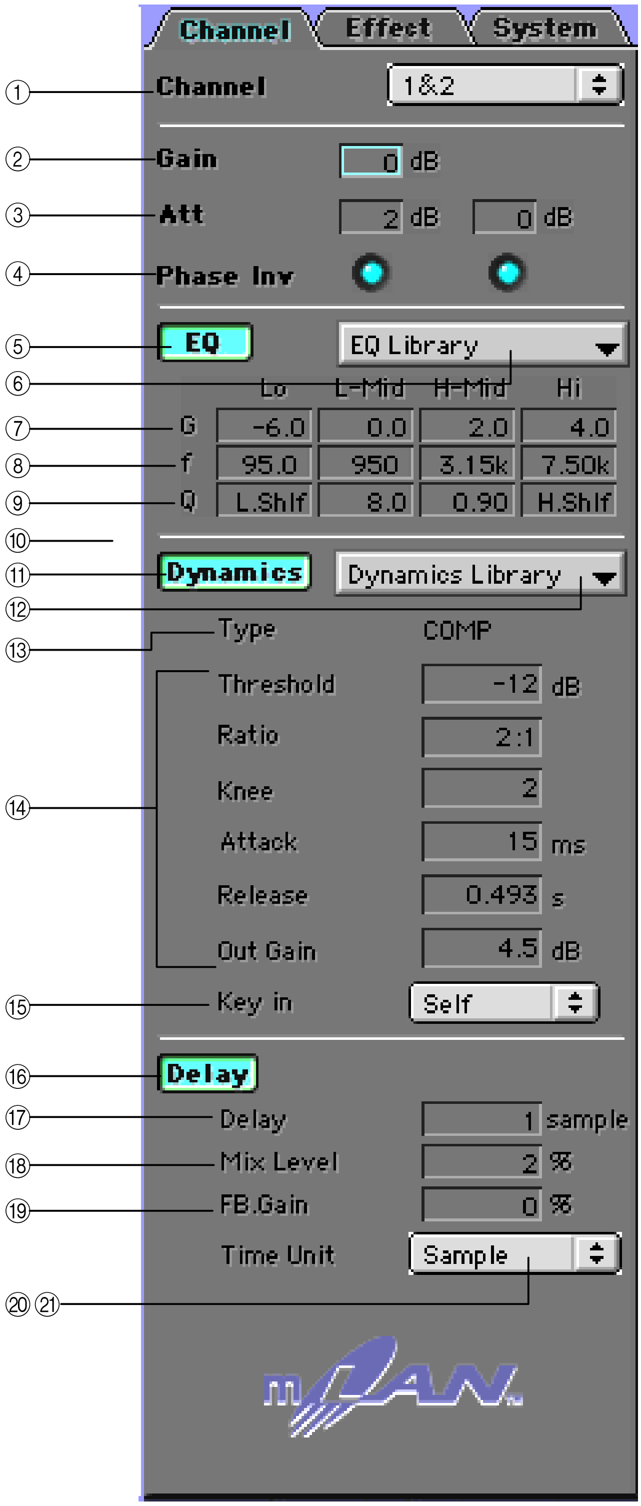
This fader adjusts the level of the stereo mix signal. Click-and-hold the mouse on the fader and slide the mouse up and down, or click the desired point on the fader to move the fader to that point.

⑧ Level meter

The level meter is displayed on the right side of the fader. The meter source signal is specified by the Mix Meter Source parameter.

Channel Tab Window

mLAN Mixer offers the parameters independently for each channel, making detailed mixing operations possible.



① Channel (Target Channel)

Select the channel you wish to set here. Click the desired input channel module to display the corresponding channel number here automatically.

② Gain

This parameter enables you to adjust the input level in steps of 6dB.
Internally, mLAN Mixer performs a Bit Shift process.
Settings: –12 – +24dB

③ Att (Attenuator)

This parameter enables you to set the input level in steps of 1dB. In this way, you can fine tune the Gain value.
Settings: –96–+24dB

④ Phase Inv (Inverted)

Click the check box to reverse the phase of the input signal.

EQ (Equalizer)

⑤ EQ button

This button turns the selected channel EQ on or off. It is linked to each channel's EQ button.

⑥ EQ Library

This parameter enables you to select a preset EQ type.

⑦ G (dB)

This parameter sets the amount of boost (+) or cut (–) of the signal at the specified frequency (F (Hz)).
Settings: –18–+18dB (in steps of 0.5dB), ON/OFF (only when [HPF] or [LPF] is selected for the "Type" parameter with the Macintosh version.)

⑧ F (Hz)

This parameter sets the equalizing frequency.
LO: .... 21 (Hz) – 20.0 k(Hz)
L-MID: .... 21 (Hz) – 20.0 k(Hz)
H-MID: .... 21 (Hz) – 20.0 k(Hz)
HI: .... 21 (Hz) – 20.0 k(Hz)

⑨ Q

This parameter sets the frequency range for boost and cut. The higher the value is, the smoother the changes in the specified frequency range. With the Macintosh version, use this parameter to make the setting for "Type."
Settings: 10.0–0.10

⑩ Type (Windows only)

This parameter enables you to select a type of EQ.

NOTE With the Macintosh version, set the Q value (⑨) to the maximum or minimum to select a Type.

[LO]
PEAK (peaking): A normal parametric equalizer
LSLV (low-shelving): A shelving-type equalizer that boosts and cuts the signal in the low range.
LPF (low pass filter): A filter that cuts the frequency range above the threshold specified by the "F (frequency)" parameter.

[HI]
PEAK (peaking): A normal parametric equalizer
HSLV (high-shelving): A shelving-type equalizer that boosts and cuts the signal in the high range.
HPF (high pass filter): A filter that cuts the frequency range below the threshold specified by the "F (frequency)" parameter.

NOTE mLAN Mixer uses a 4-band full parametric equalizer. Basically, a parametric equalizer changes a tone by boosting (+) or cutting (–) the signal at the specified frequency. Boosting emphasizes the corresponding frequency range, and cutting attenuates the range. mLAN Mixer offers many presets suitable for various applications. Select one that suits your application, and fine tune it for quick operation.

Dynamics

⑪ Dynamics button

This button turns the dynamics processor of the selected channel on or off. This button is linked with each channel's Dynamics button.

⑫ Dyna (Dynamics) Library

Select the desired Dynamics type (page 34).

⑬ **Type**

This field displays the type of dynamics selected in the Dyna Library field.

⑭ **Dynamics Parameters**

These parameters are for the dynamics of the type indicated in the "Type" field (⑬). For more information, refer to Dynamics in the Data List on page 29.

⑮ **Key in (Key in Source)**

You can set the ducking effect. Ducking is used for voice-over applications in which the background music level is reduced automatically when the announcer speaks. For dynamics processing, match the Key in the source parameter of the Target Channel (music channel) with that of the voice channel.
The input channel processors can be self-triggered (triggered by the signal to be processed), or they can be triggered by a signal from another channel.

NOTE Dynamics processors are generally used to correct or control signal levels (like a compressor) and psycho-acoustically extend the sustain. mLAN Mixer provides not only a compressor but gates, limiters, and other types of processors for various occasions.
mLAN Mixer offers many presets suitable for various applications. Select one that suits your application, and fine tune it for quick operation.

Delay

NOTE This function is not available on the mLAN8E.

⑯ **Delay button**

This button turns the selected channel Delay on or off. It is liked to each channel's Dly (Delay) button.

⑰ **Delay/Time (Delay Time)**

This parameter sets the delay time.
Settings (sample): 1–9600 samples
Settings (millisecond): 0.023–217.687ms (44.1kHz), 0.021–200.00ms (48kHz)

⑱ **Mix Level**

This parameter sets the mix balance between the dry sound and the delay sound.
With a setting of "0," the ratio of dry signal to delay signal is 1:0. With a setting of "+50," the ratio is 1:1. With a setting of "+100," the ratio is 0:1. If the value is negative, the phase of the delay signal is reversed.

⑲ **FB.Gain/FB. Level**

This parameter sets the delay feedback amount.
This level is the amount of the delay signal fed back to the delay effect. With a setting of "0," there is no feedback. With a setting of "+99," the feedback is at maximum. If the value is negative, the phase of the delay signal is reversed.

⑳ **Sample**

Select this option to display the delay time in sample units.
With 44.1kHz, one sample corresponds to 1/44,100 seconds. With 48kHz, one sample corresponds to 1/48,000 seconds.

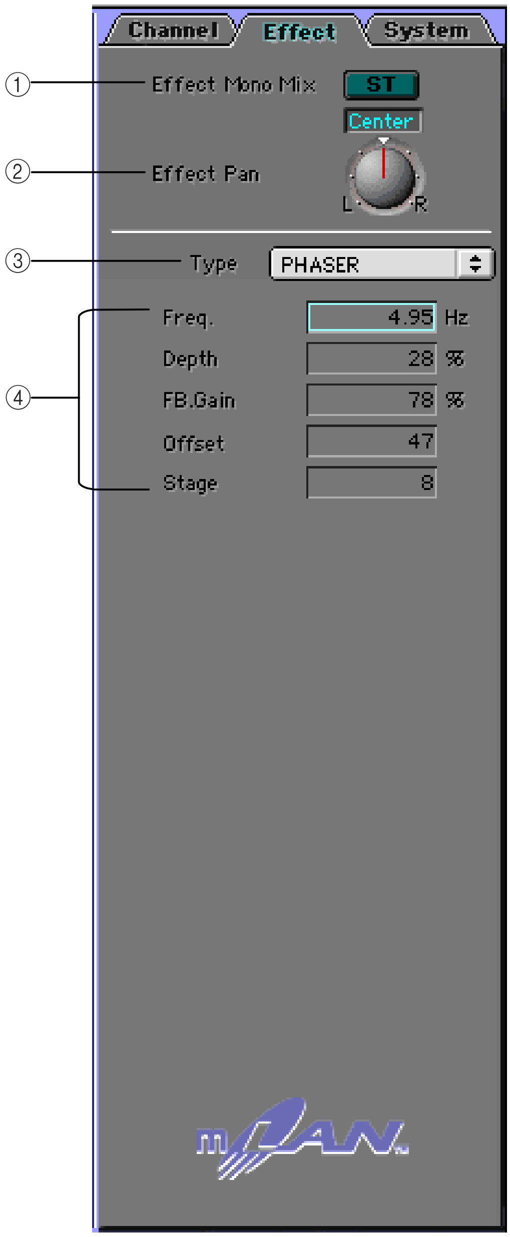
㉑ **MS (millisecond)**

Select this option to display the delay time in ms (milliseconds).

NOTE Both options are based on either 1/44,100 seconds or 1/48,000 seconds, and feature the same upper limit value. Select the desired option depending on your application.

Effect Tab Screen

You can set the internal effect parameters on this screen. (mLAN8P only)



① **Effect Mono Mix (Mono)**
This parameter enables you to mix the L/R effect return signals into a monaural signal.

② **Effect Pan (Balance/Return Pan)**
This parameter sets the L/R balance of the effect return signals.

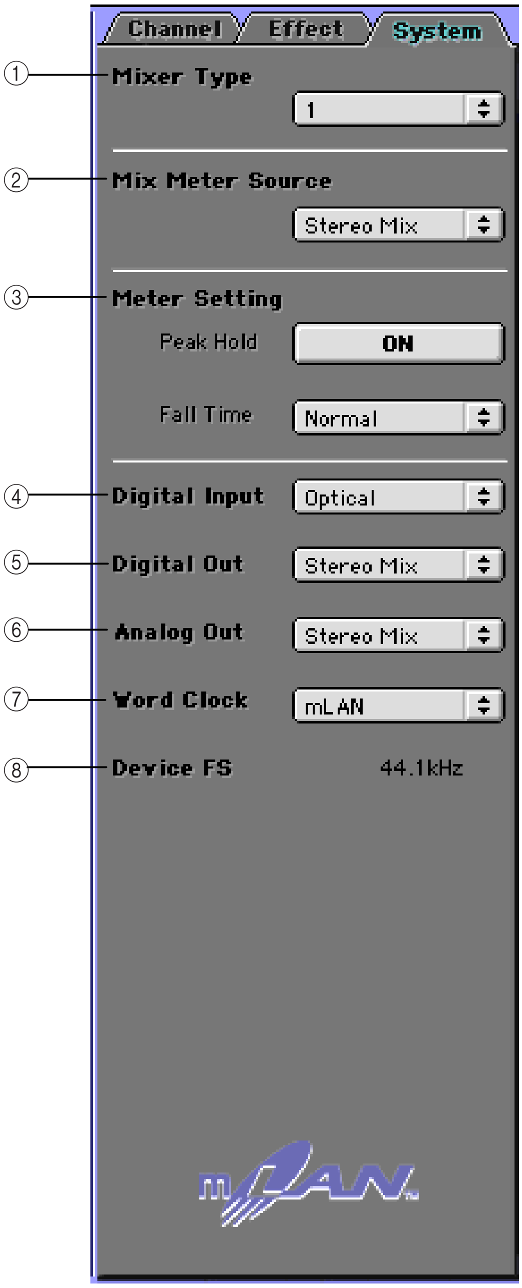
③ **Type**
This parameter enables you to select the desired effect type (page 19).

④ Use these parameters to set up the effect specified by the Type parameter. The displayed parameters differ depending on the selected effect (page 20).
mLAN Mixer offers many presets suitable for various applications. Select one best suits your application, and fine tune it for quick operation.

System Tab Screen

You can make settings related to the mLAN8P/mLAN8E on this screen.

mLAN8P



① Mixer Type

You can assign EQ and Dyna for up to eight channels. The assignments for channels 1-4 are fixed. You can assign EQ and Dyna for the other four channels here.

1: EQ and Dyna are assigned to channels 5-8.
2: EQ and Dyna are assigned to channels 5-6 and Digital in.
3: EQ and Dyna are assigned to channels 5-6 and A/D in.
4: EQ and Dyna are assigned to Digital in and A/D in.

② Mix Meter Source (Level Meter Source)

This parameter enables you to select a signal to display on the master track level meter.

Stereo Mix: Displays the stereo output signal.
Effect Return: Displays the effect return signal.
AUX1/2: Displays the AUX1/2 master output signal.

③ Meter Setting

This parameter enables you to select the type of level meter display.

Peak Hold ON: The meter holds the peak level until you turn off "Peak Hold."
Peak Hold OFF: The meter does not hold the peak level.
Fall Time: This parameter sets the speed of the meter's response to the decaying sound.

Normal: Normal response speed
Fast: The meter responds quickly to and reflects the attenuating sound. This setting clarifies the attack level on the meter.
OFF: The level meter is not displayed.

④ Digital Input

This parameter enables you to select Optical or Coaxial for the input signal at the Digital In connector on the mLAN8P.

⑤ Digital Output

This parameter specifies the signal routed to the Digital Out connector on the rear panel of the mLAN8P.
Settings: Stereo Mix, AUX1/2, Coaxial/Optical In, A/D In

⑥ **Analog Output**

This parameter specifies the signal routed to the D/A out connector on the rear panel.
Settings: Stereo Mix, AUX1/2, Coaxial/Optical (depending on the Digital Input settings) In, A/D In

⑦ **Word Clock**

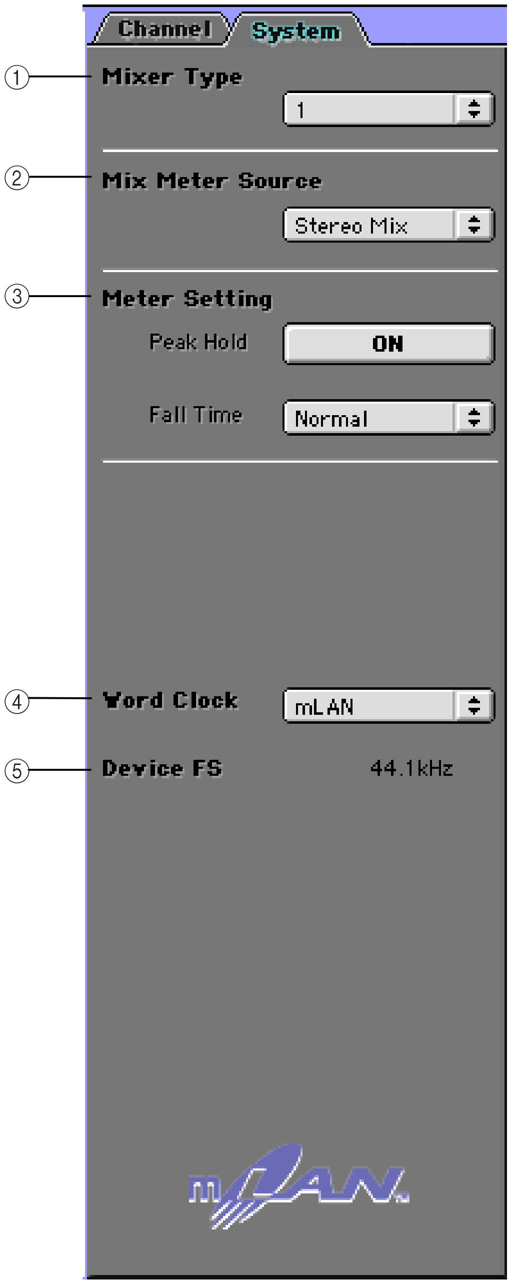
This parameter specifies the word clock for the mLAN8P.
Settings: Internal44.1k, Internal48k, Coaxial/Optical (depending on the Digital Input settings), mLAN (follows the unit's or mLAN Patchbay's setting.)

⑧ **Device Fs**

This field displays the actual sampling frequency.

NOTE The signal specified by the Analog Output source parameter is also output to connected headphones.

mLAN8E



① Mixer Type

You can assign EQ and Dyna to up to eight channels.

1: EQ and Dyna are assigned to channels 1-8.
2: EQ and Dyna are assigned to channels 1-6 and 9-10.
3: EQ and Dyna are assigned to channels 1-4 and 9-12.
4: EQ and Dyna are assigned to channels 9-16.

② Mix Meter Source (Level Meter Source)

This parameter enables you to select a signal to display on the master track level meter.

Stereo Mix: Displays the stereo output signal.
AUX1/2: Displays the AUX1/2 signal.
AUX3/4: Displays the AUX1/2 signal.
AUX5/6: Displays the AUX1/2 signal.

③ Meter Setting:

This parameter enables you to select the type of level meter display.

Peak Hold ON: The meter holds the peak level until you turn off "Peak Hold."
Peak Hold OFF: The meter does not hold the peak level.
Fall Time: This parameter sets the speed of the meter's response to the decaying sound.

Normal: Normal response speed
Fast: The meter responds quickly to and reflects the attenuating sound. This setting clarifies the attack level on the meter.
OFF: The level meter is not displayed.

④ Word Clock

This parameter specifies the word clock for the mLAN8E.

Settings: Internal44.1k, mLAN (follows the unit's or mLAN Patchbay's setting.)

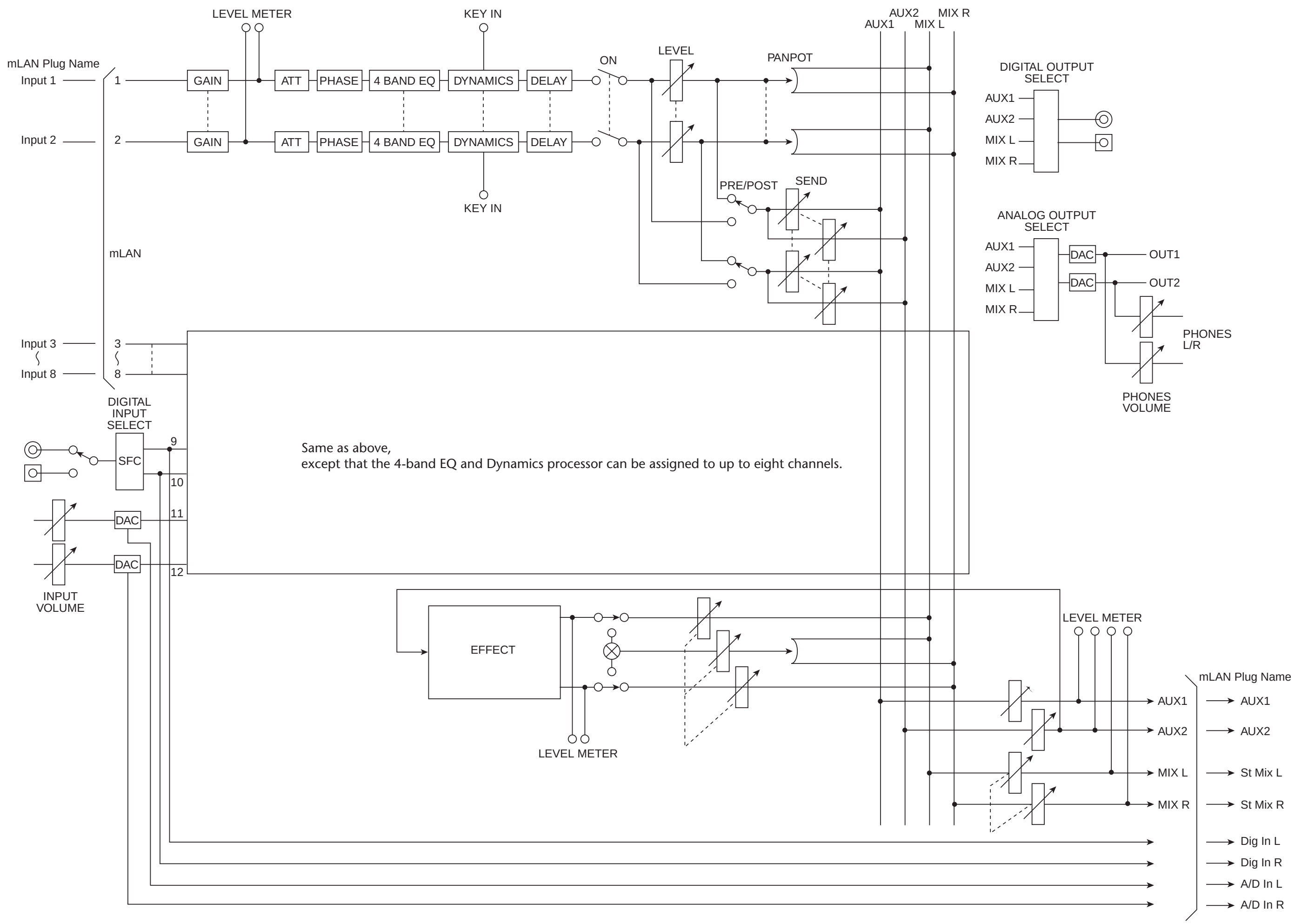
⑤ Device Fs

This field displays the actual sampling frequency.

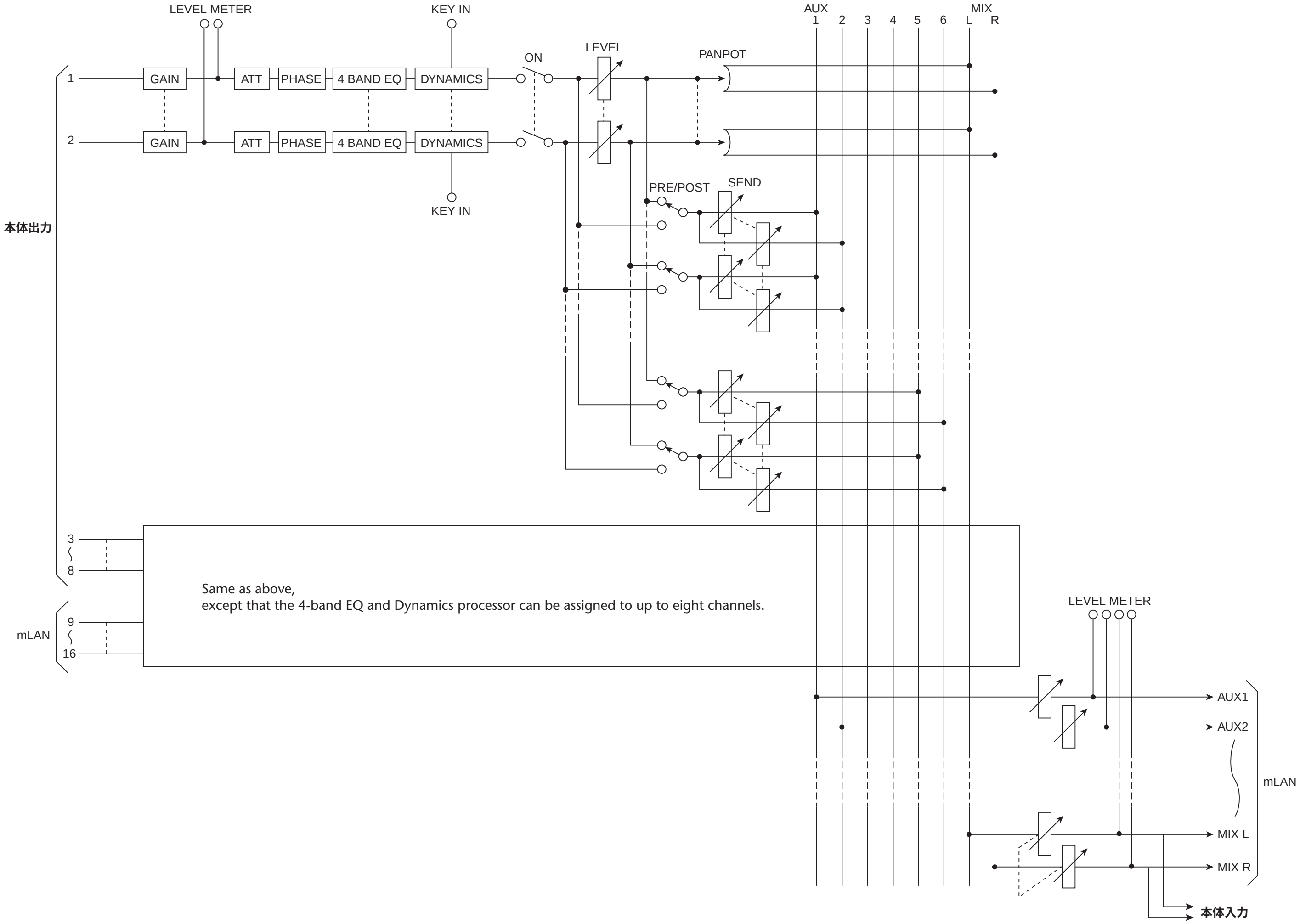
NOTE The signal specified by the Analog Output source parameter is also output to connected headphones.

Block Diagram (Audio)

mLAN8P



mLAN8E


LEVEL METER
KEY IN
AUX
1 2 3 4 5 6
MIX
L R
ON
LEVEL
PANPOT
1
2
GAIN
ATT
PHASE
4 BAND EQ
DYNAMICS
KEY IN
PRE/POST
SEND
本体出力
3
8
9
16
mLAN
Same as above,
except that the 4-band EQ and Dynamics processor can be assigned to up to eight channels.
LEVEL METER
AUX1
AUX2
mLAN
MIX L
MIX R
本体入力

Data List

Effect Type

Reverb-type Effects

#	Type	Description
01	Reverb Hall	Reverb simulating a large space such as a concert hall.
02	Reverb Room	Reverb simulating the acoustics of a smaller space (room) than REVERB HALL.
03	Reverb Stage	Reverb designed with vocals in mind.
04	Reverb Plate	Simulation of a metal-plate reverb unit, producing a feeling of hard-edged reverberation.
05	Early Ref.	An effect which isolates only the early re ection (ER) component from reverberation. A ashier effect than reverb is produced.
06	Gate Reverb	A type of ER designed for use as gated reverb.
07	Reverse Gate	A reverse-playback type ER.

Delays

#	Type	Description
08	Mono Delay	Mono delay with simple operation. Use when you don t need to use complex parameter settings.
09	Stereo Delay	Stereo delay with independent left and right.
10	Mod.delay	Mono delay with modulation.
11	Delay LCR	Three-tap delay (L, C, R).
12	Echo	Stereo delay with additional parameters for more detailed control. The signal can be fed back from left to right, and right to left.

Guitar Effects

#	Type	Description
23	Distortion	Distortion
24	Amp Simulate	Guitar Amp Simulator

Dynamic Effects

#	Type	Description
25	Dyna.Filter	Dynamically controlled filter.
26	Dyna.Flange	Dynamically controlled anger.
27	Dyna.Phaser	Dynamically controlled phase shifter.

Modulation-type Effects

#	Type	Description
13	Chorus	Three-phase stereo chorus.
14	Flange	The well-known anging effect.
15	Symphonic	A Yamaha proprietary effect that produces a richer and more complex modulation than chorus.
16	Phaser	Stereo phaser with 2–16 stages of phase shift.
17	Auto Pan	An effect which cyclically moves the sound between left and right.
18	Tremolo	Tremolo
19	Dual Pitch	Stereo pitch shift with left and right pitches set independently.
20	Rotary	Simulation of a rotary speaker.
21	Ring Mod.	An effect that modifies the pitch by applying amplitude modulation to the frequency of the input. Even the modulation frequency can be controlled by modulation.
22	Mod.Filter	An effect which uses an LFO to modulate the frequency of the filter.

Combined Effects

#	Type	Description
28	Rev+Chorus	Reverb and chorus in parallel
29	Rev->Chorus	Reverb and chorus in series
30	Rev+Flange	Reverb and anger in parallel
31	Rev->Flange	Reverb and anger in series
32	Rev+Sympho.	Reverb and symphonic in parallel
33	Rev->Sympho.	Reverb and symphonic in series
34	Rev->Pan	Reverb and auto-pan in parallel
35	Delay+ER.	Delay and early re ections in parallel
36	Delay->ER.	Delay and early re ections in series
37	Delay+Rev	Delay and reverb in parallel
38	Delay->Rev	Delay and reverb in series
39	Dist->Delay	Distortion and delay in series

Effects Parameters

REVERB HALL, REVERB ROOM, REVERB STAGE, REVERB PLATE

Parameter	Range	Description
REV TIME	0.3–99.9 s	Reverb time
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
HI.RATIO	0.1–1.0	High-frequency reverb time ratio
LO.RATIO	0.1–2.4	Low-frequency reverb time ratio
DIFF.	0–10	Reverb diffusion (left–right reverb spread)
DENSITY	0–100%	Reverb density
E/R DLY	0.0–100.0 ms	Delay between early re ections and reverb
E/R BAL.	0–100%	Balance of early re ections and reverb (0% = ER, 100% = reverb)
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

EARLY REF.

Parameter	Range	Description
TYPE	S-Hall, L-Hall, Random, Revers, Plate, Spring	Type of early re ection simulation
ROOMSIZE	0.1–20.0	Re ection spacing
LIVENESS	0–10	Early re ections decay characteristics (0 = dead, 10 = live)
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
DIFF.	0–10	Reverb diffusion (left–right reverb spread)
DENSITY	0–100%	Reverb density
ER NUM.	1–19	Number of early re ections
FB GAIN	–99 to +99%	Feedback gain
HI.RATIO	0.1–1.0	High-frequency feedback ratio
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

GATE REVERB, REVERSE GATE

Parameter	Range	Description
TYPE	Type-A, Type-B	Type of early re ection simulation
ROOMSIZE	0.1–20.0	Re ection spacing
LIVENESS	0–10	Early re ections decay characteristics (0 = dead, 10 = live)
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
DIFF.	0–10	Reverb diffusion (left–right reverb spread)
DENSITY	0–100%	Reverb density
HI.RATIO	0.1–1.0	High-frequency feedback ratio
ER NUM.	1–19	Number of early re ections
FB GAIN	–99 to +99%	Feedback gain
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

MONO DELAY

Parameter	Range	Description
DELAY	0.0–2730.0 ms	Delay time
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
HI.RATIO	0.1–1.0	High-frequency feedback ratio
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

STEREO DELAY

Parameter	Range	Description
DELAY L	0.0–1350.0 ms	Left channel delay time
FB.G L	–99 to +99%	Left channel feedback (plus values for normal-phase feedback, minus values for reverse-phase feedback)
DELAY R	0.0–1350.0 ms	Right channel delay time
FB.G R	–99 to +99%	Right channel feedback (plus values for normal-phase feedback, minus values for reverse-phase feedback)
HI.RATIO	0.1–1.0	High-frequency feedback ratio
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

MOD.DELAY

Parameter	Range	Description
DELAY	0.0–2725.0 ms	Delay time
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
HI.RATIO	0.1–1.0	High-frequency feedback ratio
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

DELAY LCR

Parameter	Range	Description
DELAY L	0.0–2730.0 ms	Left channel delay time
LEVEL L	–100 to +100%	Left channel delay level
DELAY C	0.0–2730.0 ms	Center channel delay time
LEVEL C	–100 to +100%	Center channel delay level
DELAY R	0.0–2730.0 ms	Right channel delay time
LEVEL R	–100 to +100%	Right channel delay level
FB.DLY	0.0–2730.0 ms	Feedback delay time
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
HI.RATIO	0.1–1.0	High-frequency feedback ratio
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

ECHO

Parameter	Range	Description
DELAY L	0.0–1350.0 ms	Left channel delay time
FB.G L	–99 to +99%	Left channel feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
DELAY R	0.0–1350.0 ms	Right channel delay time
FB.G R	–99 to +99%	Right channel feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
FB.D L	0.0–1350.0 ms	Left channel feedback delay time
L->R FB.G	–99 to +99%	Left to right channel feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
FB.D R	0.0–1350.0 ms	Right channel feedback delay time
R->L FB.G	–99 to +99%	Right to left channel feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
HI.RATIO	0.1–1.0	High-frequency feedback ratio
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

CHORUS

Parameter	Range	Description
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
MOD.DLY	0.0–500.0 ms	Modulation delay time
WAVE	Sine, Tri	Modulation waveform

FLANGE

Parameter	Range	Description
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
MOD.DLY	0.0–500.0 ms	Modulation delay time
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
WAVE	Sine, Tri	Modulation waveform

SYMPHONIC

Parameter	Range	Description
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
MOD.DLY	0.0–500.0 ms	Modulation delay time
WAVE	Sine, Tri	Modulation waveform

PHASER

Parameter	Range	Description
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
OFFSET	0–100	Lowest phase-shifted frequency offset
STAGE	2, 4, 8, 10, 12, 14, 16	Number of phase shift stages

AUTOPAN

Parameter	Range	Description
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
DIR.	[a]	Panning direction
WAVE	Sine, Tri, Square	Modulation waveform

a. L<->R, L >R, L< R, Turn L, Turn R

TREMOLO

Parameter	Range	Description
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
WAVE	Sine, Tri, Square	Modulation waveform

DUAL PITCH

Parameter	Range	Description
PITCH L	–24 to +24 semitones	Left channel pitch shift
FINE L	–50 to +50 cents	Left channel pitch shift fine
LEVEL L	–100 to +100%	Left channel level (plus values for normal phase, minus values for reverse phase)
PITCH R	–24 to +24 semitones	Right channel pitch shift
FINE R	–50 to +50 cents	Right channel pitch shift fine
LEVEL R	–100 to +100%	Right channel level (plus values for normal phase, minus values for reverse phase)
DELAY L	0.0–1000.0 ms	Left channel delay time
FB.G L	–99 to +99%	Left channel feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
DELAY R	0.0–1000.0 ms	Right channel delay time
FB.G R	–99 to +99%	Right channel feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
MODE	1–10	Pitch shift precision

ROTARY

Parameter	Range	Description
ROTATE	STOP, START	Rotation stop, start
SPEED	SLOW, FAST	Rotation speed (see SLOW and FAST parameters)
DRIVE	0–100	Overdrive level
ACCEL	0–10	Accelation at speed changes
LOW	0–100	Low-frequency filter
HIGH	0–100	High-frequency filter
SLOW	0.05–10.00 Hz	SLOW rotation speed
FAST	0.05–10.00 Hz	FAST rotation speed

RING MOD.

Parameter	Range	Description
SOURCE	OSC, SELF	Modulation source: oscillator or input signal
OSC FREQ	0.0–3000.0 Hz	Oscillator frequency
FM FREQ	0.05–40.00 Hz	Oscillator frequency modulation speed
FM DEPTH	0–100%	Oscillator frequency modulation depth

MOD.FILTER

Parameter	Range	Description
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
TYPE	LPF, HPF, BPF	Filter type: low pass, high pass, band pass
OFFSET	0–100	Filter frequency offset
RESO.	0–20	Filter resonance
PHASE	0.00–354.38	Left-channel modulation and right-channel modulation phase difference
LEVEL	0–100	Output level

DISTORTION

Parameter	Range	Description
DST TYPE	DST1, DST2, OVD1, OVD2, CRUNCH	Distortion type (DST = distortion, OVD = overdrive)
DRIVE	0–100	Distortion drive
MASTER	0–100	Master volume
TONE	–10 to +10	Tone

AMP SIMULATE

Parameter	Range	Description
AMP TYPE	[a]	Guitar amp simulation type
DST TYPE	DST1, DST2, OVD1, OVD2, CRUNCH	Distortion type (DST = distortion, OVD = overdrive)
DRIVE	0–100	Distortion drive
MASTER	0–100	Master volume
CAB DEP	0–100%	Speaker cabinet simulation depth
BASS	0–100	Bass tone control
MIDDLE	0–100	Middle tone control
TREBLE	0–100	High tone control
EQ F	99–8.0 kHz	Parametric equalizer frequency
EQ G	–12 to +12 dB	Parametric equalizer gain
EQ Q	10.0–0.10	Parametric equalizer bandwidth

a. STK-M1, STK-M2, THRASH, MIDBST, CMB-PG, CMB-VR, CMB-DX, CMB-TW, MINI, FLAT

DYNA.FILTER

Parameter	Range	Description
SENSE	0–100	Sensitivity
TYPE	LPF, HPF, BPF	Filter type
OFFSET	0–100	Filter frequency offset
RESO.	0–20	Filter resonance
LEVEL	0–100	Output Level
DIR.	UP, DOWN	Upward or downward frequency change
DECAY	[a]	Filter frequency change decay speed

a. 6.0 ms–46.0 s (fs=44.1 kHz), 5.0 ms–42.3 s (fs=48 kHz)

DYNA.FLANGE

Parameter	Range	Description
SENSE	0–100	Sensitivity
FB GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
OFFSET	0–100	Delay time offset
DIR.	UP, DOWN	Upward or downward frequency change
DECAY	[a]	Decay speed

a. 6.0 ms–46.0 s (fs=44.1 kHz), 5.0 ms–42.3 s (fs=48 kHz)

DYNA.PHASER

Parameter	Range	Description
SENSE	0–100	Sensitivity
FB GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
OFFSET	0–100	Lowest phase-shifted frequency offset
DIR.	UP, DOWN	Upward or downward frequency change
STAGE	2, 4, 8, 10, 12, 14, 16	Number of phase shift stages
DECAY	[a]	Decay speed

a. 6.0 ms–46.0 s (fs=44.1 kHz), 5.0 ms–42.3 s (fs=48kHz)

REV+CHORUS

Parameter	Range	Description
REV TIME	0.3–99.9 s	Reverb time
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
HI.RATIO	0.1–1.0	High-frequency reverb time ratio
DENSITY	0–100%	Reverb density
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
MOD.DLY	0.0–500.0 ms	Modulation delay time
WAVE	Sine, Tri	Modulation waveform
REV/CHO	0–100%	Reverb and chorus balance (0% = chorus, 100% = reverb)
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

REV->CHORUS

Parameter	Range	Description
REV TIME	0.3–99.9 s	Reverb time
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
HI.RATIO	0.1–1.0	High-frequency reverb time ratio
DENSITY	0–100%	Reverb density
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
MOD.DLY	0.0–500.0 ms	Modulation delay time
WAVE	Sine, Tri	Modulation waveform
REV BAL.	0–100%	Reverb and chorused reverb balance (0% = chorused reverb, 100% = reverb)
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

REV+FLANGE

Parameter	Range	Description
REV TIME	0.3–99.9 s	Reverb time
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
HI.RATIO	0.1–1.0	High-frequency reverb time ratio
DENSITY	0–100%	Reverb density
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
MOD.DLY	0.0–500.0 ms	Modulation delay time
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
WAVE	Sine, Tri	Modulation waveform
REV/FLG	0–100%	Reverb and ange balance (0% = ange, 100% = reverb)
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

REV->FLANGE

Parameter	Range	Description
REV TIME	0.3–99.9 s	Reverb time
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
HI.RATIO	0.1–1.0	High-frequency reverb time ratio
DENSITY	0–100%	Reverb density
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
MOD.DLY	0.0–500.0 ms	Modulation delay time
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
WAVE	Sine, Tri	Modulation waveform
REV BAL.	0–100%	Reverb and anged reverb balance (0% = anged reverb, 100% = reverb)
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

REV+SYMPHO.

Parameter	Range	Description
REV TIME	0.3–99.9 s	Reverb time
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
HI.RATIO	0.1–1.0	High-frequency reverb time ratio
DENSITY	0–100%	Reverb density
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
MOD.DLY	0.0–500.0 ms	Modulation delay time
WAVE	Sine, Tri	Modulation waveform
REV/SYM	0–100%	Reverb and symphonic balance (0% = symphonic, 100% = reverb)
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

REV->SYMPHO.

Parameter	Range	Description
REV TIME	0.3–99.9 s	Reverb time
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
HI.RATIO	0.1–1.0	High-frequency reverb time ratio
DENSITY	0–100%	Reverb density
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
MOD.DLY	0.0–500.0 ms	Modulation delay time
WAVE	Sine, Tri	Modulation waveform
REV BAL.	0–100%	Reverb and symphonic reverb balance (0% = symphonic reverb, 100% = reverb)
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

REV->PAN

Parameter	Range	Description
REV TIME	0.3–99.9 s	Reverb time
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
HI.RATIO	0.1–1.0	High-frequency reverb time ratio
DENSITY	0–100%	Reverb density
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth
DIR.	[a]	Panning direction
WAVE	Sine, Tri, Square	Modulation waveform
REV BAL.	0–100%	Reverb and panned reverb balance (0% = panned reverb, 100% = reverb)
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

a. L<->R, L >R, L< R, Turn L, Turn R

DELAY+ER.

Parameter	Range	Description
DELAY L	0.0–1000.0 ms	Left channel delay time
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
DELAY R	0.0–1000.0 ms	Right channel delay time
HI.RATIO	0.1–1.0	High-frequency feedback ratio
FB.DLY	0.0–1000.0 ms	Feedback delay time
DLY/ER	0–100%	Delay and early re ections balance (0% = early re ections, 100% = delay)
TYPE	S-Hall, L-Hall, Random, Revers, Plate, Spring	Type of early re ection simulation
ROOMSIZE	0.1–20.0	Re ection spacing
LIVENESS	0–10	Early re ections decay characteristics (0 = dead, 10 = live)
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
DENSITY	0–100%	Reverb density
ER NUM.	1–19	Number of early re ections

DELAY->ER.

Parameter	Range	Description
DELAY L	0.0–1000.0 ms	Left channel delay time
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
DELAY R	0.0–1000.0 ms	Right channel delay time
HI.RATIO	0.1–1.0	High-frequency feedback ratio
FB.DLY	0.0–1000.0 ms	Feedback delay time
DLY BAL.	0–100%	Delay and early re ected delay balance (0% = early re ected delay, 100% = delay)
TYPE	S-Hall, L-Hall, Random, Revers, Plate, Spring	Type of early re ection simulation
ROOMSIZE	0.1–20.0	Re ection spacing
LIVENESS	0–10	Early re ections decay characteristics (0 = dead, 10 = live)
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
DENSITY	0–100%	Reverb density
ER NUM.	1–19	Number of early re ections

DELAY+REV

Parameter	Range	Description
DELAY L	0.0–1000.0 ms	Left channel delay time
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
DELAY R	0.0–1000.0 ms	Right channel delay time
DLY HI	0.1–1.0	Delay high-frequency feedback ratio
FB.DLY	0.0–1000.0 ms	Feedback delay time
DLY/REV	0–100%	Delay and reverb balance (0% = reverb, 100% = delay)
REV TIME	0.3–99.9 s	Reverb time
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
REV HI	0.1–1.0	High-frequency reverb time ratio
DENSITY	0–100%	Reverb density
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

DELAY->REV

Parameter	Range	Description
DELAY L	0.0–1000.0 ms	Left channel delay time
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
DELAY R	0.0–1000.0 ms	Right channel delay time
DLY HI	0.1–1.0	Delay high-frequency feedback ratio
FB.DLY	0.0–1000.0 ms	Feedback delay time
DLY BAL	0–100%	Delay and delayed reverb balance (0% = delayed reverb, 100% = delay)
REV TIME	0.3–99.9 s	Reverb time
INI.DLY	0.0–500.0 ms	Initial delay before reverb begins
REV HI	0.1–1.0	High-frequency reverb time ratio
DENSITY	0–100%	Reverb density
HPF	Thru, 21 Hz–8.0 kHz	High-pass filter cutoff frequency
LPF	50 Hz–16.0 kHz, Thru	Low-pass filter cutoff frequency

DIST->DELAY

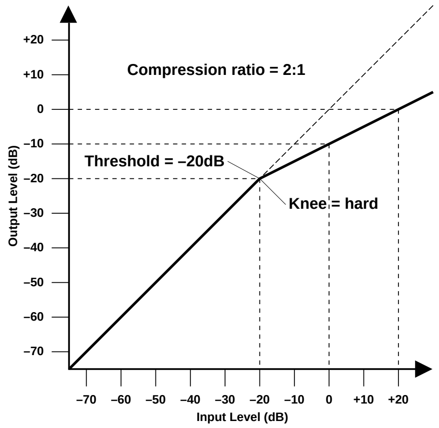
Parameter	Range	Description
DST TYPE	DST1, DST2, OVD1, OVD2, CRUNCH	Distortion type (DST = distortion, OVD = overdrive)
DRIVE	0–100	Distortion drive
MASTER	0–100	Master volume
TONE	–10 to +10	Tone control
DLY BAL	0–100%	Distortion and delay balance (0% = distortion, 100% = delayed distortion)
DELAY	0.0–2725.0 ms	Delay time
FB.GAIN	–99 to +99%	Feedback gain (plus values for normal-phase feedback, minus values for reverse-phase feedback)
HI.RATIO	0.1–1.0	High-frequency feedback ratio
FREQ.	0.05–40.00 Hz	Modulation speed
DEPTH	0–100%	Modulation depth

Dynamics

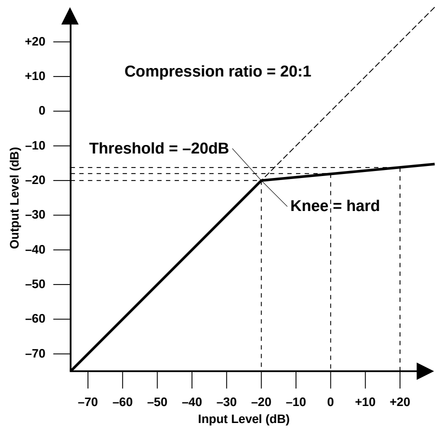
Dynamics processors are generally used to correct or control signal levels, although they can also be used creatively to shape a sound's volume envelope. The following sections explain the COMP, GATE, DUCKING, EXPAND, COMPANDER-(H), and COMPANDER-(S) dynamics processors, their parameters, and general applications.

COMP

The COMP processor is a compressor that attenuates signals above a specified threshold, providing automatic level control. Vocalists that tend to move toward and away from the microphone while singing produce fluctuating signal levels; sometimes loud, sometimes soft. Likewise, acoustic instruments with a large dynamic range produce sound levels from *pianissimo* (very soft) through to *fortissimo* (very loud). In these situations, it is often difficult to set an average fader level that will allow a voice or instrument to be heard clearly throughout a song or piece of music. This is where the compressor comes in with automatic level control. By automatically reducing high levels, thus effectively reducing the dynamic range, the compressor makes it much easier to control signals and set appropriate fader levels. Reducing the dynamic range also means that recording levels can be set higher, therefore improving signal-to-noise performance.



The COMP processor can also be used as a limiter, which is essentially a compressor with a high ratio setting. Compression ratios above 10:1 are considered to limit signals rather than compress them. When an input signal exceeds the specified threshold level, its level is automatically reduced to the threshold level. This means that the limiter's output level never actually exceeds the threshold level. Limiters are often used to prevent signals from overloading amplifiers and tape recorders. A limiter with a relatively high threshold, for example, could be used with the stereo outputs to prevent amplifier and speaker overload.



Parameter	Range
THRESHOLD	–54 dB to 0 dB (55 steps)
OUT GAIN	0.0 dB to +18.0 dB (0.5 dB steps)
KNEE	hard, 1, 2, 3, 4, 5
ATTACK	0–120 ms (1 ms steps)
RELEASE	5 ms–42.3 s (fs = 48 kHz) 6 ms–46 s (fs = 44.1 kHz)
RATIO	1:1, 1.1:1, 1.3:1, 1.5:1, 1.7:1, 2:1, 2.5:1, 3:1, 3.5:1, 4:1, 5:1, 6:1, 8:1, 10:1, 20:1, ∞:1 (16 steps)

THRESHOLD—This determines the level of input signal required to trigger the compressor. Signals at a level below the threshold pass through the compressor unaffected. Signals at and above the threshold level are compressed by the amount specified using the Ratio parameter. The trigger signal is sourced using the KEY IN parameter.

OUT GAIN—This sets the compressor's output signal level, and can be used to compensate for the overall level change caused by the compression process.

KNEE—This determines how compression is applied at the threshold point. When set to hard, compression at the specified ratio is applied as soon as the input signal level exceeds the specified threshold. For knee settings from 1 to 5, however, compression is applied gradually as the signal exceeds the specified threshold, creating a more natural sound. This is called soft-knee compression.

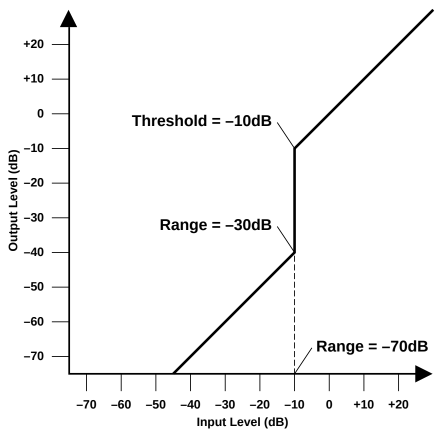
ATTACK—This determines how soon the signal is compressed once the compressor has been triggered. With a fast attack time, the signal is compressed almost immediately. With a slow attack time, however, the initial transient of a sound passes through unaffected. Attack times from 1 to 5 milliseconds are a good place to start.

RELEASE—This determines how soon the compressor returns to its normal gain once the trigger signal level drops below the threshold. If the release time is too short, the gain will recover too quickly causing level pumping (i.e., noticeable gain fluctuations). If it is set too long, the compressor may not have time to recover before the next high level signal appears, and it will be compressed incorrectly. Release times from 0.1 to 0.5 seconds are a good place to start.

RATIO—This determines the amount of compression, that is, the change in output signal level relative to change in input signal level. For a 2:1 ratio, for example, a 10 dB change in input level (above the threshold) results in a 5 dB change in output level. For a 5:1 ratio, a 10 dB change in input level (above the threshold) results in a 2 dB change in output level.

GATE

A gate, or noise gate is essentially an audio switch used to mute signals below a set threshold level. It can be used to cut background noise picked up by open microphones, noise and hiss from guitar valve amps and effects pedals, and leakage between drum microphones. It also has many creative uses too. For example, gating a drum sound with a short decay time tightens up the sound. Also, patching a gate into a droning bass synth channel and then triggering it from the kick drum channel allows the bass synth through only when the kick drum is struck, adding extra "oomph" on the beat.



Parameter	Range
THRESHOLD	–54 dB to 0 dB (55 steps)
RANGE	–70 dB to 0 dB (71 steps)
HOLD	0.02 ms–1.96 s (fs = 48 kHz) 0.02 ms–2.13 s (fs = 44.1 kHz)
ATTACK	0–120 ms (1 ms steps)
DECAY	5 ms–42.3 s (fs = 48 kHz) 6 ms–46 s (fs = 44.1 kHz)

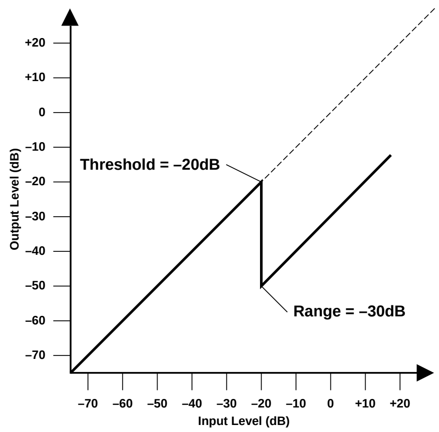
THRESHOLD—This determines the level at which the gate closes, cutting off the signal. Signals above the threshold level pass through unaffected. Signals at or below the threshold, however, cause the gate to close. The trigger signal is sourced using the KEY IN parameter.

RANGE—This determines the level to which the gate closes. Think of it as a brick holding a garden gate open so that a certain amount of signal always flows through. For a setting of –70 dB, the gate closes completely when the input signal falls below the threshold. For a setting of –30 dB, however, the gate half closes. For a setting of 0 dB, the gate has no effect. When signals are gated abruptly, the sudden disappearance can sometimes sound odd. This parameter causes the gate to reduce the signal level rather than cut it completely.

HOLD—This determines how long the gate stays open once the trigger signal has fallen below the threshold level.
ATTACK—This determines how fast the gate opens when the signal exceeds the threshold level. Slow attack times can be used to remove the initial transient edge of percussive sounds. Too slow an attack time makes some sounds appear backwards.
DECAY—This determines how fast the gate closes once the hold time has expired. A longer decay time produces a more natural gating effect, allowing the natural decay of an instrument to pass through. With a maximum decay time of between 42 and 63 seconds, you could even use this for fade-outs.

DUCKING

Ducking is commonly used for voice-over applications in which the background music level is reduced automatically when an announcer speaks. Ducking is achieved by triggering a compressor with a different sound source. For example, a ducker is patched into the background music channel, and the KEY IN signal is sourced from the announcer's microphone channel. When the announcer's microphone level exceeds the specified threshold, the background music level is reduced automatically, allowing the announcer to be heard clearly. The same technique can also be used for vocals in a mix. For example, ducking backing sounds such as rhythm guitar and synth pad during vocal phrases allows the vocals to be heard more clearly. This can also be used to bring solo instruments up in a mix.

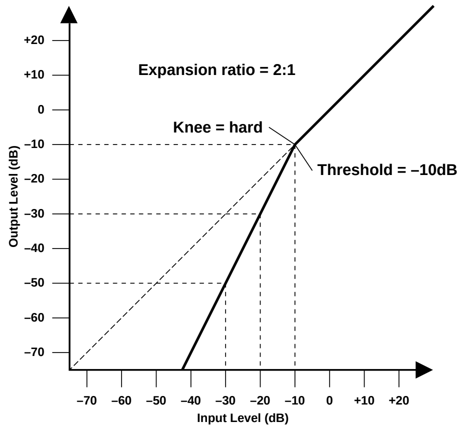


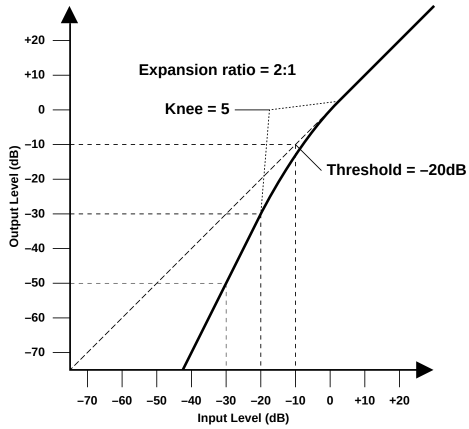
Parameter	Range
THRESHOLD	–54 dB to 0 dB (55 steps)
RANGE	–70 dB to 0 dB (71 steps)
HOLD	0.02 ms–1.96 s (fs = 48 kHz) 0.02 ms–2.13 s (fs = 44.1 kHz)
ATTACK	0–120 ms (1 ms steps)
DECAY	5 ms–42.3 s (fs = 48 kHz) 6 ms–46 s (fs = 44.1 kHz)

THRESHOLD—This determines the level of trigger signal (KEY IN) required to activate ducking. Trigger signal levels below the threshold do not activate ducking. Trigger signals at and above the threshold level, however, activate ducking, and the signal level is reduced to a level set by the Range parameter. The trigger signal is sourced using the KEY IN parameter.
RANGE—This determines the level to which the signal is ducked. For a setting of –80 dB, the signal is virtually cutoff. For a setting of –30 dB, however, the signal is ducked by 30 dB. For a setting of 0 dB, the ducker has no effect.
HOLD— This determines how long ducking remains active once the trigger signal has fallen below the threshold level.
ATTACK—This determines how soon the signal is ducked once the ducker has been triggered. With a fast attack time, the signal is ducked almost immediately. With a slow attack time, however, ducking appears to fade the signal. Too fast an attack time may sound abrupt.
DECAY—This determines how soon the ducker returns to its normal gain once the trigger signal level drops below the threshold.

EXPAND

An expander is similar to a compressor except that it works on signals below the threshold level. By reducing signals below the threshold level, the expander attenuates low-level noise, effectively increasing the dynamic range and improving the signal-to-noise performance. An expander set to an infinite ratio (i.e., ∞:1) is essentially a gate. The following two graphs show typical expander curves. The one on the left shows an expander with an expansion ratio of 2:1 and a hard knee setting. The one on the right shows an expander with an expansion ratio of 2:1 and a soft knee setting of 5.





Parameter	Range
THRESHOLD	–54 dB to 0 dB (55 steps)
OUT GAIN	0.0 dB to +18.0 dB (0.5 dB steps)
KNEE	hard, 1, 2, 3, 4, 5
ATTACK	0–120 ms (1 ms steps)
RELEASE	5 ms–42.3 s (fs = 48 kHz) 6 ms–46 s (fs = 44.1 kHz)
RATIO	1:1, 1.1:1, 1.3:1, 1.5:1, 1.7:1, 2:1, 2.5:1, 3:1, 3.5:1, 4:1, 5:1, 6:1, 8:1, 10:1, 20:1, ∞:1 (16 steps)

THRESHOLD—This determines the level of input signal required to trigger the expander. Signals above the threshold pass through the expander unaffected. Signals at and below the threshold level are attenuated by the amount specified using the Ratio parameter. The trigger signal is sourced using the KEY IN parameter.

OUT GAIN—This sets the expander's output signal level, and can be used to compensate for the overall level change caused by the expansion process.

KNEE—This determines how expansion is applied at the threshold point. When set to hard, expansion at the specified ratio is applied as soon as the input signal level falls below the specified threshold. For knee settings from 1 to 5, however, expansion is applied gradually as the signal falls below the specified threshold, creating a more natural sound.

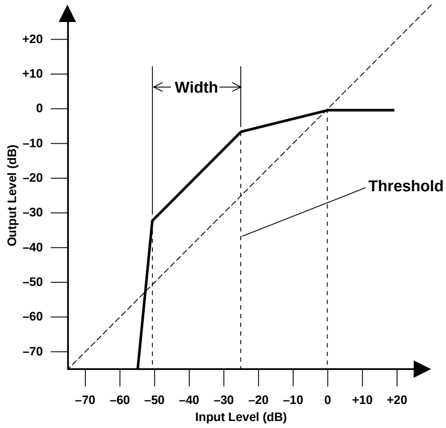
ATTACK—This determines how soon the signal is expanded once the expander has been triggered. With a fast attack time, the signal is expanded almost immediately. With a slow attack time, however, the initial transient of a sound passes through unaffected. Attack times from 1 to 5 milliseconds are a good place to start.

RELEASE—This determines how soon the expander returns to its normal gain once the trigger signal level exceeds the threshold. If the release time is too short, the gain will recover too quickly causing level pumping (i.e., noticeable gain fluctuations). If it is set too long, the expander may not have time to recover before the next low-level signal appears, and it will be expanded incorrectly. Release times from 0.1 to 0.5 seconds are a good place to start.

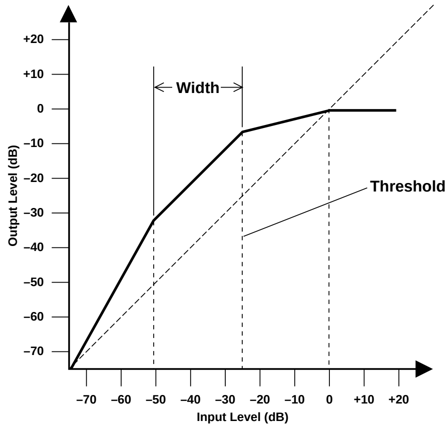
RATIO—This determines the amount of expansion. That is, the change in output signal level relative to change in input signal level. For a 2:1 ratio, for example, a 5 dB change in input level (below the threshold) results in a 10 dB change in output level. For a 5:1 ratio, a 2 dB change in input level (below the threshold) results in a 10 dB change in output level.

COMPANDER (HARD & SOFT)

The hard (H) and soft (S) companders comprise of compressor, expander, and limiter. The limiter prevents output signals from exceeding 0 dB. The compressor compresses signals that exceed the threshold level. The expander attenuates signals below the threshold and width. The soft compander has an expansion ratio of 1.5:1, while the hard compander has an expansion ratio of 5:1. The following two graphs show typical compander curves. The one on the left shows the hard compander. The one on the right, the soft compander.



Hard Compander



Soft Compander

Parameter	Range
THRESHOLD	–54 dB to 0 dB (55 steps)
OUT GAIN	–18 dB to 0 dB (0.5 dB steps)
WIDTH	1 dB–90 dB (1 dB steps)
ATTACK	0–120 ms (1 ms steps)
RELEASE	5 ms–42.3 s (fs = 48 kHz) 6 ms–46 s (fs = 44.1 kHz)
RATIO	1:1, 1.1:1, 1.3:1, 1.5:1, 1.7:1, 2:1, 2.5:1, 3:1, 3.5:1, 4:1, 5:1, 6:1, 8:1, 10:1, 20:1 (15 steps)

THRESHOLD—This determines the input signal level at which compression and expansion are applied. Signals at a level below the sum of the threshold and width are attenuated by the expander. Signals at and above the threshold level are compressed by the amount specified using the Ratio parameter. The trigger signal is sourced using the KEY IN parameter.

OUT GAIN—This sets the compander's output signal level. It can be used to compensate for the overall level change caused by the compression and expansion processes.

WIDTH—This determines how far below the threshold level expansion is applied. The expander is essentially turned off when the width is set to 90 dB.

ATTACK—This determines how soon the signal is compressed and expanded once the compander has been triggered. With a fast attack time, the signal is companded almost immediately. With a slow attack time, however, the initial transient of a sound passes through unaffected. Attack times from 1 to 5 milliseconds are a good place to start.

RELEASE—This determines how soon the compressor and expander return to their normal gains once the trigger signal level drops below the threshold. If the release time is too short, the gain will recover too quickly causing level pumping (i.e., noticeable gain fluctuations). If it is set too long, the compressor may not have time to recover before the next high level signal appears, and it will be compressed incorrectly. Release times from 0.1 to 0.5 seconds are a good place to start.

RATIO—This determines the amount of compression. That is, the change in output signal level relative to change in input signal level. For a 2:1 ratio, for example, a 10 dB change in input level (above the threshold) results in a 5 dB change in output level. For a 5:1 ratio, a 10 dB change in input level (above the threshold) results in a 2 dB change in output level. The expander ratios are fixed: 1.5:1 for the soft compander (S) and 5:1 for the hard compander (H).

Dynamics Library

#	Title	Type	Description
01	Comp	COMP	Compressor intended to reduce the overall volume level. Use it on the stereo output during mixdown. It can also be used with the stereo input.
02	Gate	GATE	Gate template.
03	Expand	EXPAND	Expander template.
04	Ducking	DUCKING	Ducking template.
05	Compander(H)	COMPAND-H	Hard-knee compressor template.
06	Compander(S)	COMPAND-S	Soft-knee compressor template.
07	A.Dr.BD	COMP	Compressor program for use with acoustic kit's bass drum.
08	A.Dr.BD	GATE	Gate program for use with acoustic kit's bass drum.
09	A.Dr.BD	COMPAND-H	COMPAND-H program for use with acoustic kit's bass drum.
10	A.Dr.SN	COMP	Compressor program for use with acoustic kit's snare drum.
11	A.Dr.SN	EXPAND	Expander program for use with acoustic kit's snare drum.
12	A.Dr.SN	GATE	Gate program for use with acoustic kit's snare drum.
13	A.Dr.SN	COMPAND-S	COMPAND-H program for use with acoustic kit's snare drum.
14	A.Dr.Tom	EXPAND	Expander program for use with acoustic kit's tom toms, which automatically reduces the volume when the tom toms are not played, helping to differentiate the bass and snare drums clearly.
15	A.Dr.OverTop	COMPAND-S	Soft-knees compander program to emphasize the attack and ambience of cymbals recorded with overhead microphones. It automatically reduces the volume when the cymbals are not played, helping to differentiate the bass and snare drums clearly.
16	E.B.Finger	COMP	Compressor program to level the attack and volume level of a finger-picked electric bass guitar.
17	E.B.Slap	COMP	Compressor program to level the attack and volume level of a slap electric bass guitar.
18	Syn.Bass	COMP	Compressor program to control or emphasize the level of a synth bass.
19	Piano1	COMP	Compressor program to brighten the tonal color of a piano.
20	Piano2	COMP	A variation on program 19, using a deep threshold to change the entire attack and level.
21	E.Guitar	COMP	Compressor program for electric guitar cutting and arpeggio-style backing performance. The sound color can be varied using different playing styles.
22	A.Guitar	COMP	Compressor program for acoustic guitar stroke and arpeggio-style backing performance.
23	Strings1	COMP	Compressor program for strings.
24	Strings2	COMP	A variation on program 23, intended for violas or cellos.
25	Strings3	COMP	A variation on program 23, intended for string instruments with a very low range, such as cellos or contrabass.
26	BrassSection	COMP	Compressor program intended for brass sounds with a fast and strong attack.
27	Syn.Pad	COMP	Compressor program for synth pad, intended to prevent diffusion of the sound.
28	SamplingPerc	COMPAND-S	Compressor program for sampled sounds, making them as powerful as real acoustic drums. This program is for percussion sounds.
29	Sampling BD	COMP	A variation on program 28, intended for sampled bass drum sounds.
30	Sampling SN	COMP	A variation on program 28, intended for sampled snare drum sounds.
31	Hip Comp	COMPAND-S	A variation on program 28, intended for sampled sound loops.
32	Solo Vocal1	COMP	Compressor program suited for use with solo vocals.
33	Solo Vocal2	COMP	A variation on program 32.
34	Chorus	COMP	A variation on program 32, intended for chorus vocals.
35	Click Erase	EXPAND	Expander program to remove click track sounds that may bleed out of the musicians monitor headphones.
36	Announcer	COMPAND-H	Hard compander program to reduce the music level when the announcer speaks, making the voice clearer.
37	Limiter1	COMPAND-S	A soft-knee compander program with a slow release.
38	Limiter2	COMP	A compressor program using the peak-stop style.
39	Total Comp1	COMP	Compressor intended to reduce the overall volume level. Use it on the stereo output during mixdown. It can also be used with the stereo input.
40	Total Comp2	COMP	A variation on program 39 with greater compression.

EQ Library

#	Title	Description
01	Bass Drum 1	Emphasizes the low range of a bass drum and the attack created by the beater.
02	Bass Drum 2	Creates a peak around 80Hz, producing a tight, stiff sound.
03	Snare Drum 1	Emphasizes snapping and rimshot sounds.
04	Snare Drum 2	Emphasizes the ranges of that classic rock snare drum sound.
05	Tom-tom 1	Emphasizes the attack of tom-toms, and creates a long, leathery decay.
06	Cymbal	Emphasizes the attack of crash cymbals, extending the sparkling decay.
07	High Hat	Use on a tight high-hat, emphasizing the mid to high range.
08	Percussion	Emphasizes the attack and clarifies the high-range of instruments, such as shakers, cabasas, and congas.
09	E.Bass 1	Makes a tight electric bass sound by cutting very low frequencies.
10	E.Bass 2	Unlike program 9, this program emphasizes the low range of an electric bass.
11	Syn.Bass 1	Use on a synth bass with emphasized low range.
12	Syn.Bass 2	Emphasizes the attack that is peculiar to a synth bass.
13	Piano 1	This is used to make a piano sound brighter.
14	Piano 2	Used in conjunction with a compressor, this program emphasizes the attack and low range of a piano sound.
15	E.G.Clean	Use for line-recording an electric guitar or semi-acoustic guitar to get a slightly hard sound.
16	E.G.Crunch 1	Adjusts the tonal quality of a slightly distorted guitar sound.
17	E.G.Crunch 2	A variation on program 16.
18	E.G.Dist. 1	Makes a heavily distorted guitar sound clearer.
19	E.G.Dist. 2	A variation on program 18.
20	A.G.Stroke 1	Emphasizes the bright tones of an acoustic guitar.
21	A.G.Stroke 2	A variation on program 20. You can also use it with a gutsy guitar sound.
22	A.G.Arpeg. 1	Corrects arpeggio technique of an acoustic guitar.
23	A.G.Arpeg. 2	A variation on program 22.
24	Brass Sec.	Use with trumpets, trombones, or sax. With one instrument, adjust the HIGH or H-MID frequency.
25	Male Vocal 1	Use as a template for male vocal. Adjust the HIGH or H-MID setting according to the voice quality.
26	Male Vocal 2	A variation on program 25.
27	Female Vo. 1	Use as a template for female vocal. Adjust the HIGH or H-MID setting according to the voice quality.
28	Female Vo. 2	A variation on program 27.
29	Chorus&Harmo	Use as a template for a chorus. It makes the entire chorus much brighter.
30	Total EQ 1	Use on a stereo mix during mixdown. Sounds even better when used with a compressor.
31	Total EQ 2	A variation on program 30.
32	Total EQ 3	A variation on program 30. Can also be used with stereo inputs or external effect returns.
33	Bass Drum 3	A variation on program 1. The low and mid range is removed.
34	Snare Drum 3	A variation on program 3. It creates a thick sound.
35	Tom-tom 2	A variation on program 5. Emphasizes the mid and high range.
36	Piano 3	A variation on program 13.
37	Piano Low	Use for the low range of a piano sound recorded in stereo.
38	Piano High	Use for the high range of a piano sound recorded in stereo.
39	Fine-EQ Cass	Use when recording to or from cassette tape to make the sound clearer.
40	Narrator	Use when recording narration.