VM-3100 Preset Patch List

Reverb + Gate (21 presets)

No.	Patch Name	Comment
P00	RV:LrgHall	Large concert hall reverberation.
P01	RV:SmlHall	Small hall reverberation.
P02	RV:Strings	Reverberation optimized for delicate highs of strings.
P03	RV:Pf.Hall	Rich and warm reverberation optimized for pianos.
P04	RV:LrgClub	Simulated reverberation of large dance floor.
P05	RV:ClubFlr	Simulated reverberation of small dance floor.
P06	RV:LrgRoom	Simulated acoustics of wide rooms with lots of reverberation.
P07	RV:MidRoom	Warm and naturally spacious room reverb.
P08	RV:Orch.Rm	Reverberation of large-capacity rooms such as big banquet halls.
P09	RV:VocalRm	Room reverb suitable for vocals and chorus.
P10	RV:Cool Pt	Distinctive bright plate reverb.
P11	RV:ShortPt	Shorter plate reverb.
P12	RV:VocalPt	Crystal-clear reverb optimized for vocals.
P13	RV:SoftAmb	Simulated reverberation of a room with minimal wall reflections.
P14	RV:RoomAmb	Natural reverberation of rooms with good acoustics, suitable for drums and guitars.
P15	RV:Cathdrl	Acoustics of a very large, high-ceilinged church.
P16	RV:LngCave	Simulated reverberation of deep caves.
P17	RV:Garage	Natural reverb that enhances unique drum sounds.
P18	RV:RockK	Reverb with many low-frequency components, suitable for rock kicks.
P19	RV:RockSn	Rich and thick sounding reverb suitable for rock snares.
P20	RV:Gated	Distinctive gate reverb.

EZ Delay (6 presets)

No.	Patch Name	Comment
P21	DL:Short	An ambience effect that adds depth to the sound by doubling.
P22	DL:Medium	Natural echo optimized for vocals.
P23	DL:Long	Long delay suited for brass and analog synth solos.
P24	DL:Analog	Analog sound with gradually diminishing feedbacking highs.
P25	DL:Hi-Pass	Delay applied only to the high end.
P26	DL:KickDub	Delay applied only to the low end.

Vocal Multi (9 presets)

No.	Patch Name	Comment
P27	VO:VocalFx	Basic setup for recording/mixdown of vocals.
P28	VO:JazzVo	A natural sounding jazz club-like ambience for warm reverb well-suited for vocals.
P29	VO:RockVo	Sound featuring limiter/enhancer processing as well as a unison effect.
P30	VO:Naratin	An effect with heavy compression, used for narration.
P31	VO:BigChrs	A spacious-sounding stereo effect similar to increasing the number of vocalists.
P32	VO:AMRadio	Sound featuring hard compression and narrower frequency range.
P33	GT:DiClean	Superclean sound like line recording directly into the console.
P34	GT:ElecAc.	Optimized for electroacoustic guitars.
P35	BS:DIedBs	Slight limiting and equalization optimized, ideal for line recording applications.

Guitar Multi (9 presets)

No.	Patch Name	Comment
P36	GT:RockLed	Straight distortion sound with delay.
P37	GT:LA Lead	Lead guitar sound with tasty compression and chorus applied.
P38	GT:MetalLd	Metal sound with dynamic, ultrahigh gain distortion.
P39	GT:RhythmC	Clean sound with compression and chorus applied.
P40	GT:DlyRiff	Delay sounds at dotted eighth note intervals when a 120 BPM riff is played.
P41	GT:BluesDv	Crunchy overdrive sound suited to blues and R&R.
P42	GT:LivPool	Crunchy sound often heard on '60s British rock.
P43	GT:Country	Clean sound featuring distinctive compression and delay.
P44	BS:AutoWah	Synth bass like sound added with auto wah essential for '70s funk.

Keyboard Multi (5 presets)

No.	Patch Name	Comment
P45	KB:SpacePh	Phaser sound with spatial expanses, for synthesizer pads.
P46	KB:RingNz	Ring modulated Noisy sound with high frequency oscillator, for guitar, keyboards.
P47	KB:DeepRng	Ring modulated Noisy sound with low frequency oscillator, for drums loops, etc.
P48	KB:Tremolo	Tremolo sounds.
P49	KB:EchoBk	Delay and Chorus sounds.

Reverb 2 (5 presets)

No.	Patch Name	Comment
P50	R2:LrgHall	Large concert hall reverberation.
P51	R2:SmlHall	Small hall reverberation.
P52	R2:Cool Pt	Distinctive bright plate reverb.
P53	R2:ShortPt	Shorter plate reverb.
P54	R2:Cathdrl	Acoustics of a very large, high-ceilinged church.

Stereo Delay Chorus (2 presets)

No.	Patch Name	Comment
P55	CH:Lt Cho	Natural stereo chorus with shallow depth for spacious, crystal-clear sound.
P56	CH:DeepCho	Intense stereo chorus that adds depth and spaciousness to the sound.

Stereo Pitch Shifter (2 presets)

No.	Patch Name	Comment
P57	PT:ST Dtun	Stereo detune: heavy sound
P58	PT:OctDown	Pitch shift: down one octave

Chorus RSS (2 presets)

No.	Patch Name	Comment
P59	CR:3D Cho1	Deep chorus sound with spatial expanses produced by RSS.
P60	CR:3D Cho2	Light chorus sound with spatial expanses produced by RSS.

Delay RSS (2 presets)

No.	Patch Name	Comment
P61	DR:RSS Alt	RSS delay with alternate panning.
P62	DR:RSSrund	Surround effect by RSS for monaural sources.

Panner RSS(2 presets)

No.	Patch Name	Comment
P63	PN:Around	Sound rotates horizontally around the listener at slow speed.
P64	PN:SpedPan	Sound goes and returns between L/R at high speed.

Mic Simulator (4 presets)

No.	Patch Name	Comment
P65	MS:57-58	Converts a general-purpose D. mic to a vocal D. mic. Rich mid/low range.
P66	MS:57-421	Converts a general-purpose D. mic to a large D. mic. For drums and guitar amp.
P67	MS:57-Line	Cancels the characteristics of D.mic, giving the sound a flat frequency response.
P68	MS:DR20-87	Converts a Roland DR-20 to a large C. mic. For vocals and acoustic inst.

Guitar Amp. Sumulator (5 presets)

No.	Patch Name	Comment
P69	GA:JC-120	Roland JC-120 amp. Sounds more authentic when used with chorus for mixdown.
P70	GA:ClnTwin	U.S. tube combo amp circa "black panel."
P71	GA:MatchLd	Hot-rodded British combo amp.
P72	GA:JMP-Stk	Late '60s British stacks.
P73	GA:5150Ld	Big tube amp standard for American heavy metal.

Stereo Dynamics Proccessor (6 presets)

No.	Patch Name	Comment
P74	DN:DanceEQ	Equalization for dancing
P75	DN:Loudnes	Loudness
P76	DN:Hard+GT	Hard comp/gate for dancing
P77	DN:TotalCp	Total Compression for broadcast mixes and similar applications.
P78	DN:Limiter	Stereo Limiter
P79	DN:Enhance	Stereo Enhancer

@2Dynamics x 2 (1 presets)

No.	Patch Name	Comment
P80	D2:DynaX2	Dynamic processor that allows independent left and right settings.

Parametric Equalizer (5 presets)

No.	Patch Name	Comment
P81	PE:Overhed	For drum kit: overhead mic; collects entire drum kit.
P82	PE:Bass	For electric bass: tight, wide-range bass sound.
P83	PE:Sax	For alto and soprano sax: high-range gain control mellows sound.
P84	PE:ElecGtr	Setting keeps lead guitar from being buried in overall mix
P85	PE:NylonGt	Transparent fret sound; gain control on fret sound's high range

Graphic Equalizer (3 presets)

No.	Patch Name	Comment
P86	GE:TotalE1	Lows and highs boosted; "just right" sound
P87	GE:TotalE2	Narrowed range of low and high cut, consistent overall sound
P88	GE:SpaceEQ	Special effect: stereo output from mono input.

Stereo Chorus (1 presets)

No.	Patch Name	Comment
P89	CH:SDD/3+4	Roland SDD-320 (button 3+4)

Stereo Flanger (2 presets)

No.	Patch Name	Comment
P90	FL:HardJet	Typically brash short flanger.
P91	FL:Hi-Band	Two BOSS HF-2s connected in stereo.

Stereo Phase (2 presets)

No.	Patch Name	Comment
P92	PH:4stage	Vintage four-stage type phaser.
P93	PH:See-Saw	See-Saw phasing between L/R (eight-stage type).

Hum Canceller (2 presets)

No.	Patch Name	Comment
P94	HC:Quiet60	Cancels 60Hz hum noise.
P95	HC:Quiet50	Cancels 50Hz hum noise.

Center Canceller (1 presets)

No.	Patch Name	Comment
P96	VC:VoCancl	Vocal canceller (removes sounds in middle of stereo field).

Isolator & Filter (1 presets)

No.	Patch Name	Comment
P97	IS:HiCancl	Isolator high-range cancel.
P98	IS:Low-Phs	Stereo anti-phase effect for low range.

Speaker Modeling (1 presets)

No.	Patch Name	Comment
P99	SPM:SpFlat	Connecting Roland's DS-90 speakers provides optimum frequency characteristics.

NOTE

Speaker Modeling can be used in FX2 only.

VM-3100 Library List

Compressor Library List

NOTE

To apply the channel compressor to the sounds output from the MASTER OUT jack, use the channel compressors as a "stereo pair" and select the channel numbers for the "INS MASTER" setting (User's Manual; p. 53).

No.	Name	Description
01	KickDrum1	Applied to the bass drum, this gives a tight, fat sound.
02	SnareDrum1	Used with the snare drum, this restrains peaks for a softer snare sound.
03	Overheads	This compression provides greater presence in situations such as when distance mikes are used to record the entire drum set.
04	DI Bass	This lends a natural softness to sounds like the solid bass sounds that are input with a direct box.
05	Mic Gt Cab	This compression produces a fat, rich midrange in the recorded sounds from a mic'ed guitar amp.
06	Breakbeats	This can be used to add a punchy lo-fi feel to sampled data and other similar sounds.
07	Rap Vocals	Used for rap and similar vocals, this reduces differences in vocal volume levels and provides greater presence.
08	Narration	Used in narration and similar applications, this reduces differences in vocal volume levels and provides greater presence.
09	StereoComp	This compresses the overall differences in volume levels in sounds mixed in stereo.
10	StereoLmtr	This compresses local peaking occurring in stereo mixed sounds.
11	RockVocals	This reduces the differences in volume levels produced by strong shouted vocals, making it easier to listen to these vocals.
12	JazzVocals	This reduces the differences in volume levels occurring in general vocals, making the vocals easier to hear.
13	AcousticGt	Used with acoustic guitar sounds, this reduces the differences in volume levels, making the sound easier to hear.
14	DI Guitar	This lends a natural softness to sounds such as the solid guitar sounds input with a direct box.
15	KeyboardLd	Added to keyboard solos, this compression reduces the differences in volume levels, making the sound easier to hear.
16	Kb Rhythm	Added to keyboard backing and similar sounds, this compression reduces the differences in volume levels, making the sound more pleasing to the ear.

Equalizer Library List

No.	Name	Description
01	Rock BD	For bass drum. A sound suitable for rock with mid-lows emphasized.
02	Rock SD	For snare drum. Drops the mid-lows and emphasizes the attack and snares.
03	Rimshot	For rim shot. Emphasizes the feeling of attack unique to a rim shot.
04	Toms	For toms. Adjust LowF and LowMidF.
05	Hi Hat	For the crisper hi-hat. Adjust bell sound with HiMidG.
06	Cymbals	For cymbals. Emphasizes the difference in tone between cymbals and their clarity.
07	Overhead	For drum kit. Use when miking the sound of the entire kit.
08	RockBass	For electric bass. For rock.
09	ElecGtr	Settings that keep the lead guitar from being buried in the mix.
10	NylonGtr	Emphasize the tone of nylon strings. Adjust fret sound with HiG.
11	BluesGtr	Adds a delicate nuance suitable when playing blues on an acoustic guitar.
12	SlideGtr	Adds a rich feel to acoustic slide guitar. Adjust HiF.
13	LineGtr	For piezo pickups. Adjust brightness with HiG.
14	Male	Improves the tone quality of a male vocal. Adjust HiG.
15	Female	Improves the tone quality of a female vocal. Adjust LoMidG.
16	Narrator	Standard equalizer for male narration. Brings out the character of the voice.

VM-3100 EZ Routing Set List

Preset Memo Screen

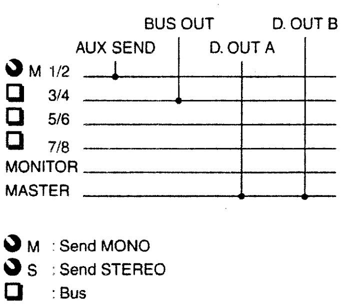
The EZ Routing Set memo screen displays the following symbols when the VM-3100 is shipped from the factory. Descriptions for each symbol are provided.

```
EZR SEL  01[BASIC MIX ]
   Memo..[AuBsDaDb  ]
         [Sd34MxMx  ]
```

Upper Row	Output Jacks
Au:	AUX SEND Jack
Bs:	BUS OUT Jack
Da:	DIGITAL OUT Connector A
Db:	DIGITAL OUT Connector B
Lower Row	Bus Assigned to Each Output Jack
Sd:	"SND-M" (Send Monaural: sends two monaural signals)
St:	"SND-S" (Send Stereo: sends one stereo signal)
Mx:	"MASTER" (Same signal as that output from the MASTER OUT jack)
Mo:	"MONITOR" (Same signal as that output from the MONITOR jack)
12:	Output of the internal "1/2" bus
34:	Output of the internal "3/4" bus
56:	Output of the internal "5/6" bus
78:	Output of the internal "7/8" bus

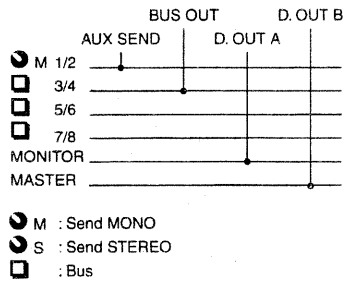
01 BASIC MIX

This is the mix using the most basic settings (shown in the connection diagram on p. 16 of the User's Manual). The monitor speakers are connected to the MASTER OUT jack. The sounds from the DIGITAL OUT A and B connectors are the same as those output from the MASTER OUT jack. Digital recordings can be made using a DAT, MD, or similar device.



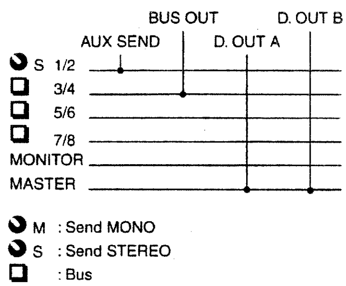
02 D MONITOR

In addition to the basics of EZ Routing Set 01, a monitor speaker that is equipped with a digital input connector (such as the optional DS-90) can be connected to the DIGITAL OUT A connector.



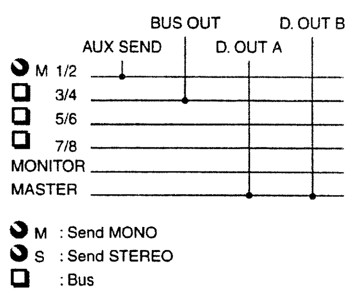
03 STEREO SND

With this setting, you can use the AUX SEND jacks 1 and 2 to connect an external stereo effects device. This allows placement of effects exactly as they are processed by the external effects device.



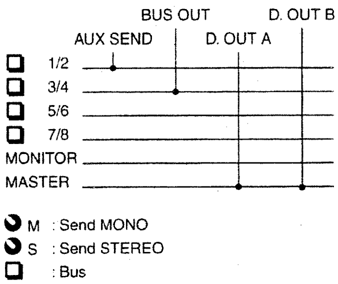
04 SUB MIXER

This features the most basic PA mixer or sub mixer settings (shown in the connection diagram on p. 35 of the User's Manual). The MASTER OUT jacks are connected the input section of the PA amp or main mixer.



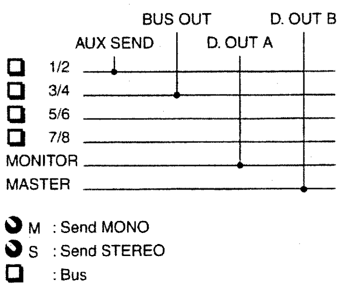
05 4Tr AnaMTR

This selects the settings used for recording to a four-track recorder (shown in the connection diagram on p. 28 of the User's Manual). The AUX SEND and BUS OUT jacks are used as "busses," with each output connected to a track input on the multitrack recorder.



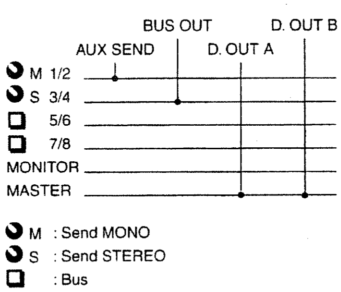
06 4Tr D-MONI

Along with the basic Routing Set 05, a monitor speaker that is equipped with a digital input connector (such as the optional DS-90) can be connected to the DIGITAL OUT A connector.



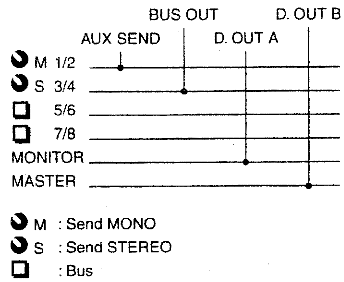
07 MIX DOWN

This contains the settings used when the VM-3100 is connected to a multitrack recorder for mixdown. All AUX SEND and BUS OUT jacks are used for sending signals to external effects. Signals can be sent both two monaural signals and one stereo signal.



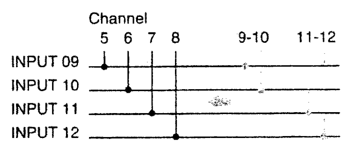
08 M.D.D-MONI

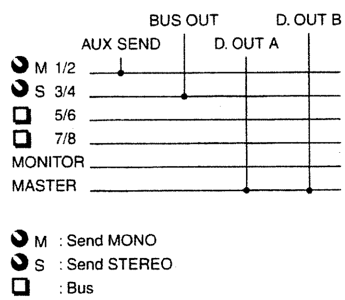
Along with the basic settings of Routing Set 07, a monitor speaker that is equipped with a digital input connector (such as the optional DS-90) can be connected to the DIGITAL OUT A connector.



09 SWAP MIX

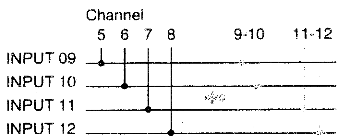
This sets up the VM-3100 so that, with the outputs of a four-track recorder connected to Inputs 9-10 and 11-12, Channels 5-8 can be used to balance the mix. When recording to the multitrack recorder, you can use Channels 9-10 and 11-12 to monitor two channels at a time, and without changing any connections, adjust the volume balance independently on Channels 5-8.

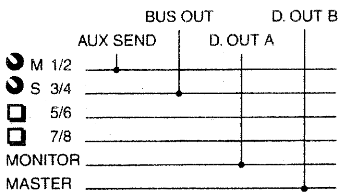




10 SWAP D-MON

In addition to the basic settings in Routing Set 09, a monitor speaker that is equipped with a digital input connector (such as the optional DS-90) can be connected to the DIGITAL OUT A connector.

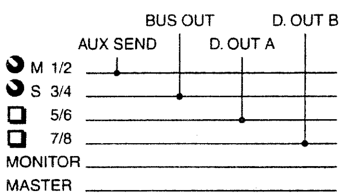




M : Send MONO
S : Send STEREO
: Bus

11 4TrDigiMTR

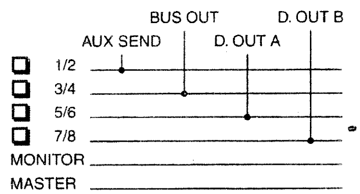
This contains the settings used for recording to a digital four-track recorder. The DIGITAL OUT A and B connectors are used as "busses" connected to the inputs of the digital multitrack recorder.



M : Send MONO
S : Send STEREO
: Bus

12 8Tr MTR

This contains the settings used for recording to a digital eight-track recorder. The AUX SEND and BUS OUT jacks and the DIGITAL OUT A and B connectors are all used as "busses" that are connected to the inputs of the digital multitrack recorder.



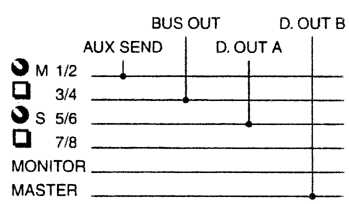
M : Send MONO
S : Send STEREO
: Bus

13 EXT DigiFX

This selects the settings needed to keep signals in digital form for effects processing when the VM-3100 is connected to an effects device featuring digital input capabilities. The DIGITAL OUT A connector is connected to the digital input of the external effects device, and the output of the effects device is connected to either the DIGITAL IN A or B connector.

NOTE

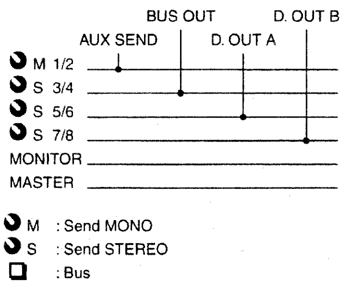
To input digital signals from an external effects device, set the Master Clock to "DIN-A" (DIGITAL IN A) or "DIN-B" (DIGITAL IN B) (User's Manual; p.52).



M : Send MONO
S : Send STEREO
: Bus

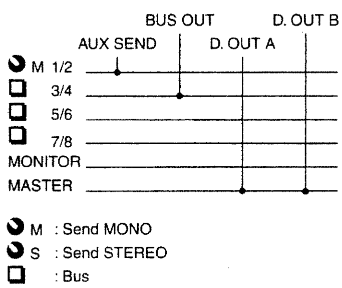
14 ALL SENDs

In this configuration, the AUX SEND and BUS OUT jacks and the DIGITAL OUT A and B connectors are all used for sending signals either to external effects or to performers' monitors (five signals, two monaural and three stereo, can be sent).



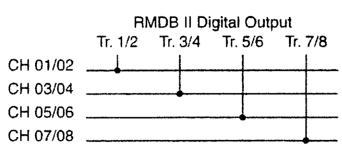
15 PA MIXER

This contains the settings for using the VM-3100 as a PA mixer equipped with sends for the performers' monitors (shown in the connection diagram on p. 38 of the User's Manual). The AUX SEND jacks send two monaural signals. MASTER OUT jacks are connected the input section of the PA amp or main mixer.



16 RMDB DIRCT

This selects the settings needed to send the output of Channels 1-8 directly through the eight-channel digital output of the RMDB II connector (VM-3100Pro only).

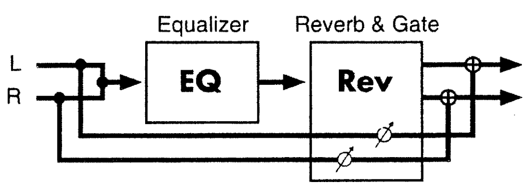


VM-3100 List of Algoriyhms

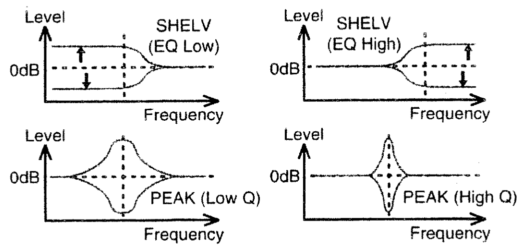
01 REVERB & GATE (p.10)
02 EZ DELAY (p.12)
03 VOCAL MULTI (p.13)
04 GUITAR MULTI 1 (p.15)
05 KEYBOARD MULTI (p.16)
06 REVERB 2 (p.18)
07 STEREO DELAY CHORUS (p.19)
08 STEREO PITCH SHIFTER (p.19)
09 CHORUS RSS (p.20)
10 DELAY RSS (P.20)
11 PANNER RSS (P.22)
12 MIC SIMULATOR (p.22)
13 GUITAR AMP. SIMULATOR (p.23)
14 STEREO DYNAMICS PROCESSOR (p.25)
15 DYNAMICS x 2 (p.25)
16 PARAMETRIC EQ (p.26)
17 GRAPHIC EQ (p.27)
18 SPACE CHORUS (p.27)
19 STEREO FLANGER (p.28)
20 VINTAGE PHASER (p.28)
21 HUM CANCELER (p.29)
22 CENTER CANCELER (p.29)
23 ISOLATOR & FILTER (p.30)
24 SPEAKER MODELING (p.31)

01 REVERB & GATE

This digital reverb creates a variety of room and hall reverberation sounds. A 3-band equalizer is connected in series before the reverb. Furthermore, the gate function provides additional special effects. This is normally used with the send/return method.



EQ (3-Band Equalizer).



Low Type [LType]SHELV, PEAK
This switches the Low EQ curve characteristics (peaking-type/shelving-type).

Low Freq [LFreq]20 - 2000 Hz
This sets the reference for the frequency range to be boost or cut. With the peaking-type equalizer, this means the center frequency; with the shelving-type equalizer, this becomes the cutoff frequency.

Low Gain [LGain]-12 - +12 dB
This sets the gain (boost or cut) of the equalizer.

Low Q [LQ]0.3 - 10.0
This sets the bandwidth, or "Q" of the sound that is boost or cut when the low-frequency equalizer is set to the peaking type. This is disabled when the shelving-type equalizer is in effect.

Mid Freq [MFreq]200 - 8000 Hz
Mid Gain [MGain]-12 - +12 dB
Mid Q [MQ]0.3 - 10.0
Just as with the low-frequency equalizer, these set the gain, center frequency, and Q for the midrange equalizer. This equalizer is peaking-type only.

High Type [HType]SHELV, PEAK
High Freq [HFreq]1.4 - 20.0 kHz
High Gain [HGain]-12 - +12 dB
High Q [HQ]0.3 - 10.0
Just as with the low-frequency equalizer, this sets the gain, center frequency, and Q for the high-frequency equalizer.

Out Level [OutLv]-60 - +12 dB
This sets the output volume.

REVERB

This is a high-quality digital reverb. It is equipped with a gate function to cut the reverb sound as it is produced, providing you with gated reverb, reverse reverb, ducking reverb, and other particular effects.

Type(Size) [Type] 5 - 40 m
This sets the size of the room. For example, the setting "10m" gives you reverb as it would sound in a space with 10-meter walls.

Rev Time [Time] (Reverb Time)0.1 - 32.0 sec
This sets the length (in seconds) of the reverb sound.

FX Level [EffLv]0 - 100
This sets the volume of the reverb sound. When using the insert method, first get a rough balance between the reverb and the dry sounds, then lower the level a little.

Dry Level [DryLv]0 - 100
This sets the volume of the source sound. Set this to 0 when using the send/return method. Increase the value when using the insert method to mix the source sound into the output.

Pre Delay [PreDL]0 - 200 msec
This sets the delay between the time source sound is first played and the point at which the reverb sound is played. This indicates distance from the source of the sound.

Diffusion [Diff.]0 - 100
Increasing this value intensifies the sense of spatial width. This is effective when playing back sounds in stereo.

Density [Dens.]0 - 100
Increasing this value makes the reverb sound denser. Reduce the density when using hall and garage reverbs.

Early Ref. [E.Ref] (Early Reflection)0 - 100
Raising the value for this setting increases the volume of the initial reflections. Early reflections are the direct reflections of the walls. You can hear these sounds as they spread out in the initial reverb sound.

LoDampFreq [LDFrq] (Low Damp Frequency) 50 - 4000 Hz
This sets the upper frequency limiting the range to be dampened. The Low Damp effect rapidly dampens the low frequency range of the reverb sound, resulting in a cleaner reverb effect.

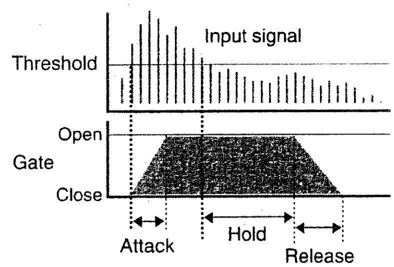
LoDampGain [LD-G] (Low Damp Gain) -36 - 0dB
This sets the amount of the Low Damp effect.

HiDampFreq [HDFrq] (High Damp Frequency) 1.0 - 20.0 kHz
In the natural world, the high frequencies in reverberation die out very quickly. High Damp, by attenuating the higher frequencies first, makes the reverb sound more natural. This sets the lower frequency limiting the range to be dampened.

HiDampGain [HD-G] (High Damp Gain) -36 - 0dB
This sets the amount of the High Damp effect. By conbining low damp and high damp, you can indicate the qualities of the room such as surface material (or the sound absorption properties thereof).

HiCutFreq [HiCut] (High Cut Frequency) 0.2 - 20.0 kHz
This gently cuts the upper frequencies of the reverb sound, making the reverberation more stable. This alteration does not change with time.

GATE



Mode [Mode] (Gate Mode)GATE, DUCK
GATE (Gate Reverb)

When the source volume falls below a certain level, the gate closes, resulting in an effect similar to a gated reverb cutting off the reverb sound.

DUCK (DuckingReverb)

When the source volume gets high enough, the gate closes, which gives a ducking-type reverb effect. Stop the reverb sound only the input loud sound so that prevent the play sound become unclear.

Threshold [Thre] (Threshold Level)0 - 100
This sets volume level of the source sound needed to close the gate and cut the reverb sound.

Attack [Atk] (Attack Time)0 - 100
This sets the time it takes for the gate to fully open after being triggered.

Hold [Hold] (Hold Time)0 - 100
This sets the time it takes for cutting of the reverb sound to start from the instant the source sound reaches the threshold level.

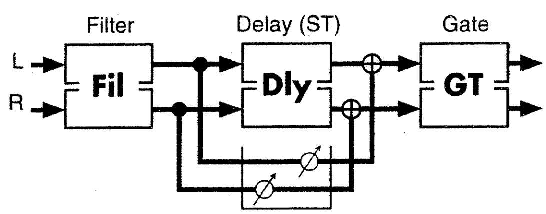
Release [Rele] (Release Time)0 - 100
This sets the elapsed time between moment the gate begins to close to when it is fully close after the hold time has elapsed.

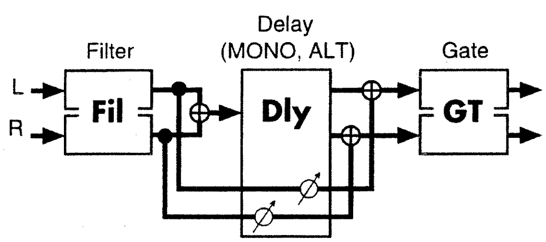
NOTE

When using the gate function to get special reverb effects, make setting the gate easier by using longer reverb times. In such instances, instead of using Low Damp or High Damp to change the tone, do this with the High Cut frequency settings or through equalization at an earlier stage. To get sharp gate reverb, make the attack and release times extremely short, and set expression time to match the rhythm with the hold time setting. To get reverse reverb, sufficiently lengthen the attack time, and keep the release time short.

02 EZ DELAY

This is a simple digital delay featuring high-quality sound. You can set length of the delay to get long echoes or fat, thick sounds. This algorithm is normally used with the send/ return method.





FILTER

These filters allow you to greatly affect the frequency characteristics of the input sound. There are four types from which to select.

Type [Type] (Filter Type)LPF, BPF, HPF, NOTCH

This sets the type of filter used.

LPF (Low pass filter)

This filter passes frequencies below the cutoff frequency.

BPF (Band pass filter)

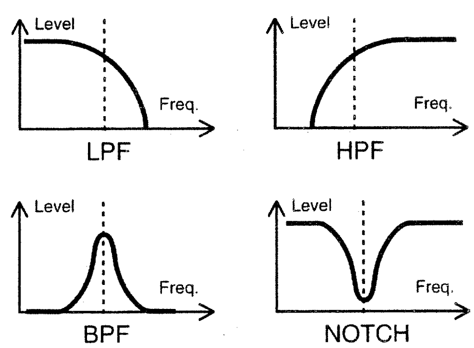
This filter passes frequencies near the cutoff frequency.

HPF (High pass filter)

This filter passes frequencies above the cutoff frequency.

NOTCH (Notch filter)

This filter passes frequencies other than those near the cutoff frequency.



Slope(oct) [Slope]-12, -24 dB

This sets the filter's slope characteristics at the cutoff frequency (-24 dB at one octave: steep; -12 dB at one octave: shallow).

CutOffFreq [Freq] (Cutoff Frequency)0 – 100

This sets the filter's cutoff frequency. The closer this is set to zero, the lower the cutoff frequency; the closer to 100, the higher the frequency.

Resonance [Reso]0 – 100

This sets the filter's resonance level. Raising the setting increases resonance near the cutoff frequency, giving the sound a particular characteristic. If the Resonance value is set too high, another sound (oscillation) begins to appear. Take care to prevent this sound from damaging your ears or your playback equipment.

Gain [Gain]0 – 24 dB

This compensates for the drop in volume in the cut frequency range in some filters. The level of correction increases as the value is increased, raising the volume.

DELAY

This digital delay can be switched between stereo, mono, and alternate settings. It features a maximum delay of 1200 msec (1.2 seconds).

Type [Type]ST, MONO, ALT

This switches stereo, monaural or alternate.

MONO (Monaural)

This is a single-input, dual-output delay. Stereo sound is mixed before being input.

ST (Stereo)

This is a dual-input, dual-output delay. The delay sound output features the same stereo placement as that of the input.

ALT (Alternate)

The left and right output of this alternate delay is the reverse of the input.

Time [Time]1 - 1200 msec

This sets the delay time, that is, the elapsed time between the source sound and the delay sound. When in mono or stereo mode, the settings value is limited to the range allowed by the left-right shift settings. In alternate mode, this is limited to 1–600 milliseconds.

L-R Shift [Shift] (L-R Time Shift) L1199 - R1199 msec

This shifts the location from where the sound appears to originate by increasing the delay sound only on the left or the right. Depending on the time setting, settings values may be limited. This is disabled in alternate mode.

L-R Order [Order]L>>R, L<<R

In alternate mode, this setting determines which side plays the delay sound before the other (with L>>R, the left side is expressed first; when set to L<<R, the right side is expressed first). This is enabled only in alternate mode.

Feedback [FB Lv] (Feedback Level)0 - 100

This sets the times for the repeated delay sound. When set to 0, each delayed sound is played only once.

FX Level [EffLv]0 - 100

This sets the volume of the delayed sound. When using the insert method, first get a rough balance between the reverb and the dry sounds, then lower the level a little.

Dry Level [DryLv]0 - 100

This sets the volume of the source sound. Set this to 0 when using the send/return method. Increase the value when using the insert method to mix the source sound into the output.

LoDampFreq [LDFrq] (Low Damp Frequency) 50 - 4000 Hz

The Low Damp effect rapidly dampens the low frequency range of the delayed sound, resulting in a cleaner delay effect. This sets the upper frequency limiting the range to be dampened.

LoDampGain [LD-G] (Low Damp Gain) -36 - 0dB

This sets the amount of the Low Damp effect.

HiDampFreq [HDFrq] (High Damp Frequency) 1.0 - 20.0 kHz

In the natural world, the high frequencies die out very quickly. By attenuating the higher frequencies first, High Damp makes the delay sound more natural. This sets the lower frequency limiting the range to be dampened.

HiDampGain [HD-G] (High Damp Gain) -36 - 0 dB

This sets the amount of the High Damp effect.

GATE

These parameters are the same as those in the GATE in Algorithm 01 (REVERB & GATE) (p.10).

Mode [Mode] (Gate Mode)GATE, DUCK

Threshold [Thre] (Threshold Level)0 – 100

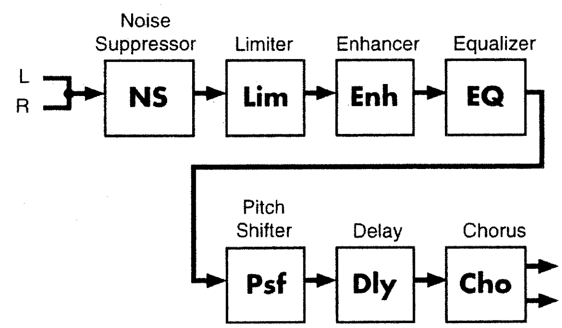
Attack [Atk] (Attack Time)0 – 100

Hold [Hold] (Hold Time)0 – 100

Release [Rele] (Release Time)0 – 100

03 VOCAL MULTI

This is a multi-effects for vocals.



NOISE SUPPRESSOR

This suppresses noise (such as background noise and hum from mics) at times when no sound is being played. The noise suppressor looks at the input level at the beginning of the effects chain, and when there is no input, suppresses any output at the end.

Threshold [Thre] (Threshold Level)0 - 100

This sets the volume level at which starts muting. Set the value higher when there is a lot of noise, and if there is less noise, decrease the value.

Release [Rele] (Release Time)0 - 100

This sets the time from when the noise suppression starts to the point where the volume reaches 0.

This effect is lost If the threshold level is set too low, while setting it too high mutes even the sounds you want to hear. Furthermore, if the release time is set too long, the noise suppression then becomes distracting; when set too short, it sounds unnatural. Adjust to obtain the most suitable settings for the input noise conditions at any given time.

LIMITER/DE-ESSER

You can use either the Limiter or De-esser functions of this effect. The limiter is an effect that compresses high-level signals, thereby preventing distortion. De-esser is an effect that cuts the sibilance in vocals, giving sounds a softer quality.

Mode [Mode]LMT, DES

This determines whether the Limiter or De-esser function is used.

LMT Level [Level] (Limiter Output Level) -60 – +12 dB

This sets the level of the signal passing through the Limiter.

Thresh [Thre] (Limiter Threshold Level) -60 – 0 dB

This adjusts the level of the signal at which the Limiter begins to function (the threshold level).

Release [Rele] (Limiter Release Time)0 – 100

This adjusts the time for the Limiter to stop functioning after the signal falls back under the threshold level.

DES Sens [Sens] (De-esser Sens)0 – 100

This adjusts the sensitivity of the de-esser effect based on the input level.

Freq [Freq] (De-esser Frequency) 1.0 – 10.0 kHz

This adjusts the frequency to which the De-esser effect is applied. The effect works best at higher frequencies than that of the settings.

ENHANCER

This effect regulates the high-end overtones, clarifying the sound and the sound contour.

Sens [Sens] (Sensitivity)0 – 100

This sets the degree to which the Enhancer effect is applied.

Frequency [Freq]1.0 – 10.0 kHz

This sets the lower limit of the frequencies to which the enhancement effect is added.

Mix Level [MixLv]0 – 100

This sets the amount of the overtones produced by the Enhancer that is mixed in with the source sound.

Out Level [OutLv]0 – 100

This sets the output volume.

EQ (3-Band Equalizer).

These parameters function in the same way as those in the EQ in Algorithm 01 (REVERB & GATE) (p.10).

Low Type [LType]SHELV, PEAK
Low Freq [LFreq]20 – 2000 Hz
Low Gain [LGain]-12 – +12 dB
Low Q [LQ] 0.3 – 10.0
Mid Freq [MFreq]200 – 8000 Hz
Mid Gain [MGain]-12 – +12 dB
Mid Q [MQ] 0.3 – 10.0
High Type [HType]SHELV, PEAK
High Freq [HFreq]1.4 – 20.0 kHz
High Gain [HGain]-12 – +12 dB
High Q [HQ] 0.3 – 10.0
Out Level [OutLv]-60 – +12 dB

PITCH SHIFTER

This effect changes the pitch of the source sound.

Pitch [Pitch]-12 – +12

This adjusts the pitch in semitone (half-step) increments.

Fine [Fine]-100 – +100

This finely adjusts the pitch shift.

FX Level [EffLv]0 – 100

This sets the volume of the pitch-shifted sound.

Dry Level [DryLv]0 – 100

This sets the volume of the direct sound.

DELAY

These parameters function in the same way as those in the DELAY in Algorithm 02 (EZ DELAY) (p.12).

Type [Type] MONO, ALT
Time [Time] 1 – 1200 msec
Feedback [FB Lv] (Feedback Level)0 – 100
FX Level [EffLv]0 – 100
Dry Level [DryLv]0 – 100

CHORUS

This effect adds breadth to the sound, making it "fatter."

Mod.LR Phs [Phase] (Modulation L-R Phase) NORM, INV

This is ordinarily set to NORM. When set to INV (INVERT), the modulation (rising and falling sound) in the right channel is inverted against the left channel. This gives an effect in which the modulation in the left and right channels is reversed.

Depth [Depth]0 – 100

This sets the chorus modulation depth.

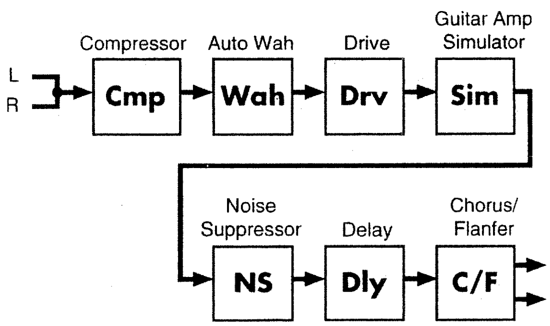
Rate [Rate]0 - 100
This sets the chorus modulation cycle time.

FX Level [EffLv]0 - 100
This adjusts the chorus volume level.

Dry Level [DryLv]0 - 100
This adjusts the volume level of the direct sound.

04 GUITAR MULTI 1

This is a multi-effect for guitars.



COMPRESSOR

This effect compresses the level of the signal by reducing the level of strong input signals and boosting low-level signals.

Attack [Atk] (Attack Time)0 - 100
This adjusts the attack strength when the sound is input.

Level [Level] (Output Level)0 - 100
This adjusts the Compressor volume level.

Sustain [Sus] (Sustain Time)0 - 100
This adjusts the length of time that the compressor continues to raise and hold the level of weak input.

Tone [Tone]0 - 100
This adjusts the compressor tone.

AUTO/TOUCH WAH

Wah is an effect created by the periodic change in a filter's frequency characteristics, giving a particular kind of tone change. You can get the wah effect by changing the volume of the input sound or by using cyclical time-based changes.

Type [Type] (Filter Type)LPF, BPF
This selects the type of filter used to make the wah.
This selects either the BPF (band pass filter) or LPF (low pass filter).
When set to BPF, the wah effect occurs within a narrow frequency range; setting this to LPF produces the wah effect over a wide range of frequencies.

Frequency [Freq] (Cutoff Freqency)0 - 100
This sets the reference frequency for the wah effect (the frequency at which the wah starts).

Peak [Peak]0 - 100
This sets the amount of wah effect near the reference frequency. The range narrows as the value increases; lower the value to get the wah effect over a wider range.

Polarity [Pol.]UP, DOWN
When the wah effect is added through changes in the source sound volume, this setting is for selecting whether the effect is to be added to the high frequencies (UP) or lower frequencies (DOWN).

Trig.Sens [Sens]0 - 100
Sets the sensitivity level when wah is added through changes in the source sound volume. The wah effect is added at lower volumes as the volume increaces.

LFO Depth [Depth]0 - 100
This sets the depth of the wah sound when the effect changes cyclically. Set this to 0 when changes in the effect are not based on time cycles.

LFO Rate [Rate]0 - 100
This adjusts the cycle time when the wah effect changes cyclically.

DRIVER

This effect adds distortion, "spreading" the sound.

Type [Type]OD, DS, METAL
This selects the effect type.

METAL
This distorts the sound most.

DS
This is what most consider the typical distortion effect.

OD
This provides the mildest distortion of the three settings.

Gain [Gain]0 - 100
This sets the amount of distortion.

Out Level [OutLv]0 - 100
This sets the volume of the effect sound.

OD/DS Tone [Tone]0 - 100
This adjusts the tone character. Setting becomes valid when TYPE is DS or OD.

METAL HiG [HGain] (High Gain)0 - 100
This sets the gain of the high range. Setting becomes valid when TYPE is METAL.

METAL MidG [MGain] (Middle Gain)0 – 100
This sets the gain of the midrange. Setting becomes valid when TYPE is METAL.

METAL LowG [LGain] (Low Gain)0 – 100
This sets the gain of the low range. Setting becomes valid when TYPE is METAL.

AMP. SIMULATOR

This simulates the sound of a guitar amplifier.

Amp.Type [Type] (Amplifier Type)
SMALL, BuiltIN, 2STACK, 3STACK
This selects the guitar amp type.

SMALL: Small amp
BuiltIN: Built-in type amp
2STACK: Stack of two large amps
3STACK: Stack of three large amps

NOISE SUPPRESSOR

These parameters function the same way as those in the NOISE SUPPRESSOR in Algorithm 03 (VOCAL MULTI) (p.13).

Threshold [Thre] (Threshold Level)0 – 100
Release [Rele] (Release Time)0 – 100

DELAY

These parameters function in the same way as those in the DELAY in Algorithm 02 (EZ DELAY) (p.12).

Type [Type] ST, MONO, ALT
Time [Time] 1 – 1200 msec
Feedback [FB Lv] (Feedback Level)0 – 100
FX Level [EffLv]0 – 100
Dry Level [DryLv]0 – 100

CHORUS/FLANGER

This provides you with chorus or flanger effects to suit your needs. Chorus is an effect that adds breadth and fullness to the sound. The flanger gives you effect that is like a jet sound rising and falling.

Mode [Mode]CHORUS, FLANGER
This selects either the chorus or the flanger.

Mod.LR Phs [Phase]
(Modulation L-R Phase)NORM, INV
This sets the phase when the chorus or flanger sound is mixed in with the source sound in the left and right channels. When this is set to NORM, the channels are in phase; when set to INV (inverted), the phases of left and right channels are inverted relative to each other.

Depth [Depth]0 – 100
This sets the chorus or flanger modulation depth.

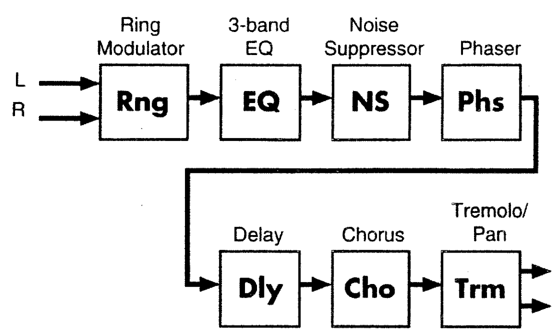
Rate [Rate]0 – 100
This sets the chorus or flanger modulation cycle time.

FL Manual [Manu] (Flanger Manual)0 – 100
This sets the center frequency at which the chorus or flanging effect is applied.

FL Reso [Reso] (Flanger Resonance)0 – 100
The more this value is increased, the stronger this distinctive effect becomes. If the Resonance value is set too high, another sound (oscillation) begins to appear.

05 KEYBOARD MULTI

This is a multi-effect designed for keyboard.



RING MODULATOR

This creates a bell-like sound by ring-modulating the guitar sound with the signal from the internal oscillator. The sound will be unmusical and lack distinctive pitches.

Osc.Freq [OscF.]0 – 100
This adjusts the frequency of the internal oscillator.

FX Level [EffLv]0 – 100
This adjusts the volume of the effect sound.

Dry Level [DryLv]0 – 100
This adjusts the volume of the direct sound.

EQ (3-Band Equalizer).

These parameters are the same as those in the EQ in Algorithm 01 (REVERB & GATE) (p.10).

Low Type [LType]SHELV, PEAK
Low Freq [LFreq]20 – 2000 Hz
Low Gain [LGain]-12 – +12 dB
Low Q [LQ] 0.3 – 10.0
Mid Freq [MFreq]200 – 8000 Hz

Mid Gain [MGain]-12 – +12 dB

Mid Q [MQ] 0.3 – 10.0

High Type [HType]SHELV, PEAK

High Freq [HFreq]1.4 – 20.0 kHz

High Gain [HGain]-12 – +12 dB

High Q [HQ] 0.3 – 10.0

Out Level [OutLv]-60 – +12 dB

NOISE SUPPRESSOR

These parameters function the same way as those in the NOISE SUPPRESSOR in Algorithm 03 (VOCAL MULTI) (p.13).

Threshold [Thre] (Threshold Level)0 – 100

Release [Rele] (Release Time)0 – 100

PHASER

By adding varied-phase portions to the direct sound, the phaser effect gives a whooshing, swirling character to the sound.

CenterFreq [Freq] (Center Frequency)0 - 100

Adjusts the center frequency of the phaser effect.

Resonance [Reso]0 - 100

Determines the amount of resonance (feedback). Increasing the value will emphasize the effect, creating a more unusual sound. Setting it to a minus value will create resonance having a reversed phase.

Depth [Depth]0 - 100

Determines the depth of the Phaser effect.

Rate [Rate]0 - 100

This sets the rate of the Phaser effect.

Separation [Sepr] (Stereo Separation)0 - 100

Adjusts the diffusion. The diffusion increases as the value increases.

DELAY

These parameters function in the same way as those in the DELAY in Algorithm 02 (EZ DELAY) (p.12).

Time [Time] 1 – 1200 msec

Feedback [FB Lv] (Feedback Level)0 – 100

FX Level [EffLv]0 – 100

Dry Level [DryLv]0 – 100

CHORUS

A sound with a subtly shifted pitch is added to the direct sound, making the final output sound thicker and broader.

Mod.LR Phs [Phase]NORM, INV

This sets the phase when the chorus or flanger sound is mixed in with the source sound in the left and right channels. When this is set to NORM, the channels are in phase; when set to INV (inverted), the phases of left and right channels are inverted relative to each other.

Depth [Depth]0 - 100

Adjusts the depth of the Chorus effect. To use it for doubling, set the value to "0."

Rate [Rate]0 - 100

Adjusts the rate of the Chorus effect.

FX Level [EffLv]0 - 100

This adjusts the volume of the effect sound.

Dry Level [DryLv]0 - 100

This adjusts the volume of the direct sound.

Pre Delay [PreDL]0 - 50 msec

Adjusts the time needed for the effect sound to be output after the direct sound has been output. By setting a longer Pre Delay time, you can obtain an effect that sounds like more than one sound is being played at the same time (doubling effect).

LoCutFreq [LoCut] (Low Cut Frequency) THRU, 50 - 800 Hz

The low cut filter cuts the frequencies below the specified frequency. This setting adjusts the frequency at which the low cut filter will begin to take effect. When "THRU" is selected, the low cut filter will have no effect.

HiCutFreq [HiCut] (High Cut Frequency) 0.5 - 12.5 kHz, THRU

The high cut filter cuts the frequencies above the specified frequency. This setting adjusts the frequency at which the high cut filter will begin to take effect. When "THRU" is selected, the high cut filter will have no effect.

TREMOLO/PAN

Tremolo is an effect that creates a cyclic change in volume. Pan cyclically moves the stereo position between left and right (when stereo output is used).

Mode [Mode] TremTRI, TremSQR, PanTRI, PanSQR

Selection for tremolo or pan. And selection for the waveform that the effect will use.

TremTRI

The volume will change cyclically. Smooth change will be produced.

TremSQR

The volume will change cyclically. Abrupt change will be produced.

PanTRI

The sound will be moved cyclically between left and right. Smooth change will be produced.

PanSQR

The sound will be moved cyclically between left and right. Abrupt change will be produced.

Depth [Depth]0 - 100

Adjusts the depth of the effect.

Rate [Rate]0 - 100

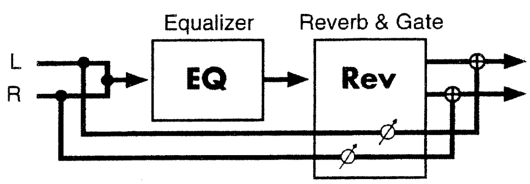
Adjusts the frequency (speed) of the change.

L-R Balance [Bal]L63 - R63

Adjusts the stereo position of the sound.

06 REVERB 2

A simulation of the reverberation of a room or hall.



EQ (3-Band Equalizer).

These parameters are the same as those in the EQ in Algorithm 01 (REVERB & GATE) (p.10).

Low Type [LType]SHELV, PEAK

Low Freq [LFreq]20 – 2000 Hz

Low Gain [LGain]-12 – +12 dB

Low Q [LQ] 0.3 – 10.0

Mid Freq [MFreq]200 – 8000 Hz

Mid Gain [MGain]-12 – +12 dB

Mid Q [MQ] 0.3 – 10.0

High Type [HType]SHELV, PEAK

High Freq [HFreq]1.4 – 20.0 kHz

High Gain [HGain]-12 – +12 dB

High Q [HQ] 0.3 – 10.0

Out Level [OutLv]-60 – +12 dB

REVERB

Reverberation (or reverb) is the effect caused by sound waves decaying in an acoustic space, or a digital simulation thereof. This decay occurs because sound waves bounce off many walls, ceilings, objects, etc. in a very complex way. These reflections, coupled with absorption by various objects, dissipate the acoustic energy over a certain period of time (called the decay time). The ear perceives this phenomenon as a continuous wash of sound.

Rev Type [Type] (Reverb Type)ROOM, HALL

This selects the Reverb Type. Various different simulations of space are offered.

ROOM

Simulates the reverberation in a small room.

HALL

Simulates the reverberation in a concert hall.

RoomSize [Size] (Room Size)1 - 10

This parameter adjusts the size of the room which is simulated.

Rev Time [Time] (Reverb Time)0.1 - 32.0 sec

This parameter adjusts the duration (time) of the reverb.

FX Level [EffLv]0 - 100

This sets the volume of the reverb sound. When using the insert method, first get a rough balance between the reverb and the dry sounds, then lower the level a little.

Dry Level [DryLv]0 - 100

This sets the volume of the source sound. Set this to 0 when using the send/return method. Increase the value when using the insert method to mix the source sound into the output.

Pre Delay [PreDL]0 - 200 msec

This parameter adjusts the time interval between the direct sound and the beginning of the reverb sound.

Density [Dens.]0 - 100

Adjust the density of the sound (Early Reflections) that arrives at the listener after bouncing off the walls once or a few times.

Early Ref. [E.Ref] (Early Reflection)0 - 100

This parameter adjusts the volume level of the initial reflected sound.

LoDampFreq [LDFrq] (Low Damp Frequency) 50 - 4000 Hz

This parameter adjusts the frequency at which the low-frequencies are damped. The reverb sound in the band below this frequency is damped.

LoDampGain [LD-G] (Low Damp Gain) -36 - 0 dB

This parameter adjusts the amount of damping for Low Damp. No low-frequency damping occurs when set to "0."

HiDampFreq [HDFrq] (High Damp Frequency) 1.0 - 20.0 kHz

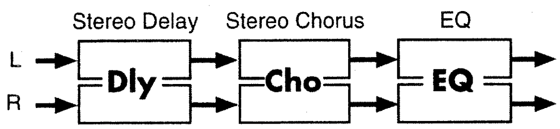
This parameter adjusts the standard frequency at which the high-frequencies are damped. The reverb sound in the band above the standard frequency is damped.

HiDampGain [HD-G] (High Damp Gain)
-36 – 0 dB
This parameter adjusts the amount of damping for High Damp. No high-frequency damping occurs when set to "0."

HiCutFreq [HiCut] (High Cut Frequency)
0.2 – 20.0 kHz
This parameter adjusts the frequency at which a low-pass filter starts to be applied. The effect is applied to the reverb sound.

07 STEREO DELAY CHORUS

This algorithm connects a stereo delay and a stereo chorus in series, allowing you to add depth and spaciousness to the sound while preserving the positioning of the stereo input signal.



DELAY

This parameter creates a distinctive effect (such as a thicker sound) by applying a delayed sound to the direct sound. By using Tempo Delay, you can easily set the delay time to match the tempo of the song.

Time L [TimeL]1 – 1200 msec
Time R [TimeR]1 – 1200 msec
This parameter adjusts the delay time (i.e., the interval for which sound is delayed).

Feedback L [FB L]0 – 100
Feedback R [FB R]0 – 100
This parameter adjusts the amount of feedback. Changing the amount of feedback causes the number of time the delayed sound is repeated to change as well.

FX Level [EffLv]0 – 100
This adjusts the volume of the delay sound.

Dry Level [DryLv]0 – 100
Adjusts the volume of the direct sound.

HiCutFreq [HiCut] (High Cut Frequency)
0.2 – 20.0 kHz, THRU
The High Cut Filter cuts the frequency contents that are higher than the set frequency. This parameter adjusts the frequency where the high cut filter starts working. When it is set to "THRU," the high cut filter does not work at all.

HiDampGain [HD-G] (High Damp Gain)
-36 – 0 dB
This parameter adjusts the amount of damping for High Damp. No high-frequency damping occurs when set to "0."

CHORUS

These parameters are the same as those in the CHORUS in Algorithm 05 (KEYBOARD MULTI) (p.16).

Mod.LR Phs [Phase]NORM, INV
Depth [Depth]0 – 100
Rate [Rate] 0 – 100
FX Level [EffLv]0 – 100
Dry Level [DryLv]0 – 100
Pre Delay [PreDL]0 – 50msec
LoCutFreq [LoCut] (Low Cut Frequency)THRU, 50 – 800 Hz
HiCutFreq [HiCut] (High Cut Frequency)0.5 – 12.5 kHz, THRU

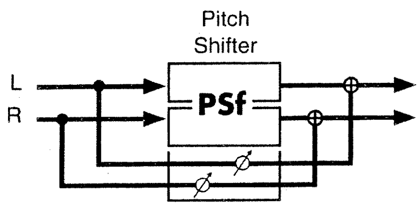
EQ (3-Band Equalizer)

These parameters are the same as those in the EQ in Algorithm 01 (REVERB & GATE) (p.10).

Low Type [LType]SHELV, PEAK
Low Freq [LFreq]20 – 2000 Hz
Low Gain [LGain]-12 – +12 dB
Low Q [LQ] 0.3 – 10.0
Mid Freq [MFreq]200 – 8000 Hz
Mid Gain [MGain]-12 – +12 dB
Mid Q [MQ] 0.3 – 10.0
High Type [HType]SHELV, PEAK
High Freq [HFreq]1.4 – 20.0 kHz
High Gain [HGain]-12 – +12 dB
High Q [HQ] 0.3 – 10.0
Out Level [OutLv]-60 – +12 dB

08 STEREO PITCH SHIFTER

This algorithm features two pitch shifters arranged in parallel, making it stereo compatible. It can shift the pitch of the input signal up to one octave up or down. This algorithm can be used with either the insert method or the send/return method.



PITCH SHIFTER

This effect changes the pitch of the source sound. The degree of pitch shift can be set separately for each channel.

StereoLink [Link]OFF, ON
This selects whether the pitch shift in left and right channels are to be linked or set independently. When set to "ON," the right channel pitch shifter settings conform to those set for the left channel.

Grade [Grade] (Sound Grade)1, 2, 3, 4, 5
This sets the quality of the effect sound. The higher the value is set, the more natural-sounding the effect is; however, this increases the delay from the source sound as well.
Depending on the setting, you may be able to hear some disruption of the sound in drums and other parts, so select the setting after listening to the sound at different settings.

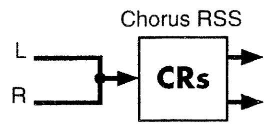
Lch Pitch [PichL]-12 – +12
Lch Fine [FineL] -100 – +100
Rch Pitch [PichR] -12 – +12
Rch Fine [FineR]-100 – +100
These set the degree of left and right pitch shift. You can adjust the pitch shift in semitones with "Pitch" and in cents (1/100 of a semitone) with "Fine" for minute adjustment of the pitch shift. When Stereo Link is on, changes to the right channel settings are disregarded.

FX Level [EffLv]0 – 100
This sets the volume of the effect.

Dry Level [DryLv]0 – 100
This sets the volume of the source sound. When simply changing the pitch of the source sound, set the dry level to 0 and use with the insert method.

09 CHORUS RSS

This algorithm is a chorus with RSS connected to the output. The sound of the left channel is placed 90 degrees left, and the sound of the right channel is placed 90 degrees right.



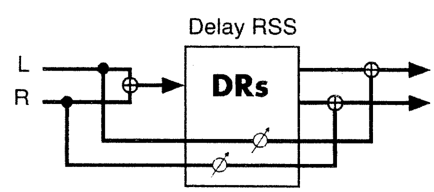
CHORUS RSS

These parameters function in the same way as those in the CHORUS in Algorithm 05 (KEYBOARD MULTI) (p.16).

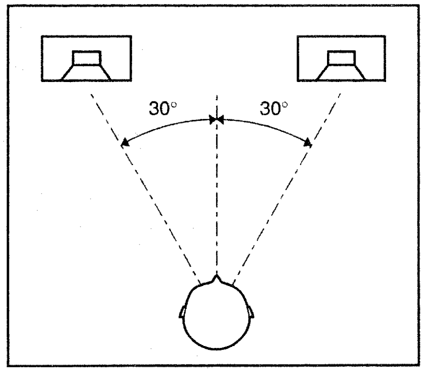
Mod.LR Phs [Phase]NORM, INV
Depth [Depth]0 – 100
Rate [Rate] 0 – 100
FX Level [EffLv]0 – 100
Dry Level [DryLv]0 – 100
Pre Delay [PreDL]0 – 50 msec
LoCutFreq [LoCut] (Low Cut Frequency)THRU, 50 – 800 Hz
HiCutFreq [HiCut] (High Cut Frequency)0.5 – 12.5 kHz, THRU

10 DELAY RSS

This is a single-in/dual-out delay with RSS effects added to the output. When heard through stereo speakers, a space of 90 degrees between the left and right sides (of your head) opens up and a wider, three-dimensional delay sound can be heard within that space. This is usually added with the send/return method.



Notes on Using RSS
To exhibit the RSS effect to the fullest extent, take note of the following points.
- RSS works best in rooms where there is little reverberation.
- One-way speakers are most appropriate. However, coaxial or virtual coaxial speakers may also be used.
- Keep speakers as far away from side walls as possible.
- Do not position the left and right speakers with too much distance between them.
- Listen from the optimal position, as shown below.

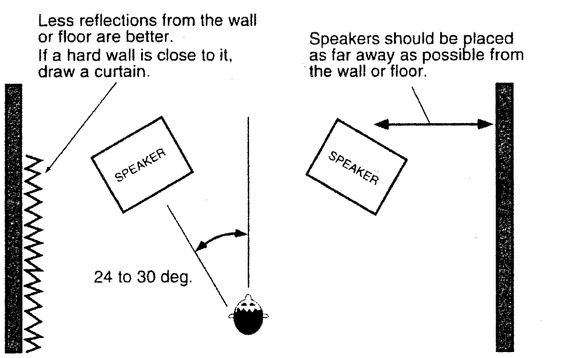


Notice on Package with RSS

RSS is an effect that gives the sound device three-dimensional sound with an ordinary stereo system. Monitoring environment is critical in exhibiting the RSS effect to the fullest extent. We recommend that packaging for products containing songs that use the RSS patch carry the following description at the time of sale.

For Stereo Speakers

This sound is made to be played specifically through speakers.
The proper effect cannot be obtained if listened to through headphor



DELAY RSS

This single-input delay adds the RSS effect for widened spatial characteristics.

Time [Time]1 – 1200 msec

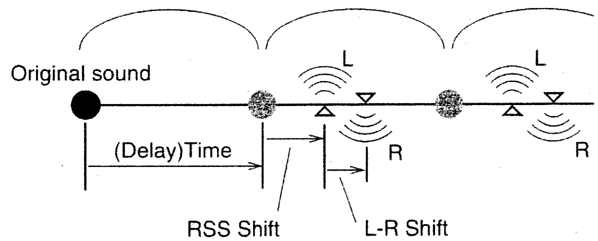
This sets the elapsed time between the source sound and the delay sound (the delay time) in millisecond units. The settings value is limited to the range allowed by the left-right shift and RSS shift settings.

RSS Shift [RsSft] (RSS Time Shift)
-1199 – 1199 msec

The further increases the delay time only of sounds processed through RSS before the sounds are played. The settings value is limited to the range allowed by the delay time and left-right shift settings.

L-R Shift [LRSft] (L-R Time Shift)
L1199 – R1199 msec

This shifts the location from where the sound appears to originate by increasing the delay sound only on the left or the right. The settings value is limited to the range allowed by the delay time and left-right shift settings.



Feedback [FB Lv] (Feedback Level)0 – 100

This sets the delay sound repeat time. When set to 0, the delayed sound is played only once.

FX Level [EffLv]0 – 100

MonoDly [Mono] (Monaural Delay Level)
0 – 100

RSS Dly [RSS] (RSS Delay Level) 0 – 100

This sets the volume of the delay sound. Set levels for the monaural delay and RSS delay sounds individually, and then adjust the total level of the overall effect. You can better interpret the RSS effect by setting the monaural delay level to 0. However, with the L-R shift set to 0 (no shift), the RSS effect may be difficult to hear.

Dry Level [DryLv] 0 – 100

This sets the volume of the source sound. Set this to 0 when using the send/return method. Increase the value when using the insert method to mix the source sound into the output.

LoDampFreq [LDFrq] (Low Damp Frequency)
50 – 4000 Hz

The Low Damp effect rapidly dampens the low frequency range of the delayed sound, resulting in a cleaner delay effect. This sets the upper frequency limiting the range to be dampened.

LoDampGain [LD-G] (Low Damp Gain)
-36 – 0 dB

This sets the amount of the Low Damp effect.

HiDampFreq [HDFrq] (High Damp Frequency) 1.0 - 20.0 kHz

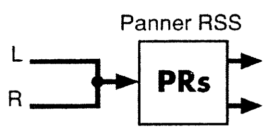
In the natural world, the high frequencies die out very quickly. By attenuating the higher frequencies first, High Damp makes the delay sound more natural. This sets the lower frequency limiting the range to be dampened.

HiDampGain [HD-G] (High Damp Gain) -36 - 0 dB

This sets the amount of the High Damp effect.

11 PANNER RSS

RSS (Panner) can make the sound seem to revolve around the listener.



PANNER RSS

Speed [Speed] (Roratiion Speed)0 - 100

This parameter adjusts the speed with which the position of the sound moves.

Direction [Dir] (Rotation Direction)CW, CCW

This parameter selects the sound's direction of rotation.

CW(Clockwise)

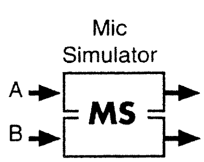
Rotates the sound clockwise.

CCW(Counterclockwise)

Rotates the sound counterclockwise.

12 MIC SIMULATOR

This effect takes sounds recorded using standard dynamic mics, pin mics, or line signals, and converts them so that they sound as if they were recorded using an expensive studio-quality condenser mic. This also lets you add proximity effect, distance, and other effects.



LINK

This is the link switch for Channels A and B.

Link Sw [Link] (Link Switch)OFF, ON

When set to Off, each of the two channels works independently as a mono channel equalizer. When set to On, both equalizer channels work simultaneously on Channel A. (The Channel B settings are disregarded.)

MIC SIMULATOR A/B

This effect converts the characteristics of inexpensive, all-purpose mics to those of expensive, studio-quality mics (microphone'microphone conversion). It makes signals that have already been recorded in your Project sound as if the changes in sound quality were made through mic selection and placement. This also adds characteristics of microphones to instrument sounds recorded through line input (line'microphone conversion).

Mic Conv. [Conv.] (Mic Converter Switch)OFF, ON

This switches the Mic Converter on and off. When turned off, the TypeIn, TypeOut, and Phase settings are disabled.

TypeIn [M.In] (Input Mic Type) DR-20, SmallD, HeadD, MiniC, FLAT

This selects the type of mic to be used for recording.

DR-20

Roland DR-20 (dynamic mic manufactured by Roland)

SmallD

Small dynamic mic used for miking instruments, vocals, and the like

HeadD

Headset-type dynamic mic

MiniC

Mini condenser mic

FLAT

Line input

TypeOut [M.Out] (Output Mic Type) SmallD, VocalD, LargeD, SmallC, LargeC, Vnt.C, FLAT

This selects the type of mic simulated.

SmallD

Dynamic mic for general use with instruments and vocals. Perfoect for guitar amps and snare drums.

VocalD

Dynamic mic especially known for use with vocals. Features exceptional midrange presence. For vocals.

LargeD

Dynamic mic with extended low range. For bass drums, toms, and similar applications.

SmallC

Small condenser mic for use with instruments. Features a particularly fine high range. For use with metal percussion instruments and acoustic guitars.

LargeC

Flat-response condenser mic. For vocals, narration, live instruments, and the like.

Vnt.C

Vintage condenser mic. For vocals, instruments, and the like.

FLAT

Mic with flat frequency response characteristics. Use this when you want the sound of a mic used for miking larger groups.

NOTE

When a condenser-type mic is selected in TypeOut, low-range noise transmitted through the mic stand may be accentuated due to the mic's low range characteristics. In such instances, either cut out any unnecessary low end with bass cut filter, or equip the mic stand with an isolation mount (a mic holder with rubber or other shock absorbing material).

Phase [Phase]NORM, INV

This selects the mic phase.

NORM: In phase to the input.

INV: Invented phase to the input.

BassCut [BsCut] (Bass Cut Filter Switch)OFF, ON

This filter cuts out popping and other such noises as well as unneeded low end sounds. Switching this on creates a simulated bass cut filter. When turned off, the Freq setting is disabled.

Freq [Freq] (Frequency)THRU, 20 - 2000 Hz

This adjusts the bass cut filter's cutoff frequency.

Position [Pos] (Mic Position Switch)OFF, ON

Microphones tend to accentuate the low end the closer they are placed to the source sound. This is known as the proximity effect. Switching on this effect simulates frequency characteristics and timing differences that change with distance. When turned off, the ProxFx, Distance settings are disabled.

Prox.Fx [Prox] (Proximity Effects)-12 - +12

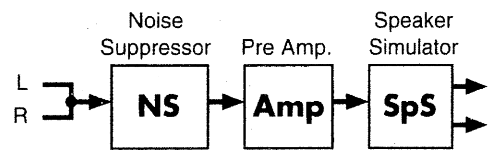
Microphones tend to accentuate the low end the closer they are placed to the source sound. This effect simulates those qualities, and compensates for the low end characteristics that change with distance. Positive settings bring the mic closer to the source, and negative settings put the mic at a greater distance.

Distance [Dis] (Mic Distance)0 - 3000 cm

This simulates the time difference that changes with distance from the source.

13 GUITAR AMP. SIMULATOR

This algorithm simulates a guitar amp.



NOISE SUPPRESSOR

These parameters function the same way as those in the NOISE SUPPRESSOR in Algorithm 03 (VOCAL MULTI) (p.13).

Threshold [Thre] (Threshold Level)0 – 100

Release [Rele] (Release Time)0 – 100

PRE AMP. SIMULATOR

This effect simulates the pre-amp section of a guitar amplifier.

Amp.Type [Type] (Pre Amp Type)J C-120, CleanTW, MatchDR, BG LEAD, 1959-1, 1959-2, 1959-12, SLDN, 5150, MetalLD, OD-1, OD-2T, DS, FUZZ

Select the type of guitar amp.

JC-120

The sound of a Roland JC-120.

CleanTW

The sound of a standard built-in type vacuum tube amp.

MatchDR

The sound of a recent vacuum tube amp widely used in blues, rock, and fusion.

BG LEAD

The sound of a vacuum tube amp representative of the late 70's and the 80's.

1959-1

The sound of the large vacuum tube amp stack that was indispensable to the British hard rock of the 70's, with input I connected.

1959-2

The same amp as 1959-1, but with input II connected.

1959-12

The same amp as 1959-1, but with inputs I and II connected in parallel.

SLDN

The sound of a vacuum tube amp usable in a wide variety of styles.

5150

The sound of a large vacuum tube amp suitable for heavy metal.

MetalLD

A metal lead sound with a distinctive mid-range.

OD-1

The sound of the BOSS OD-1 compact effector.

OD-2T

The sound of the BOSS OD-2 compact effector with the Turbo switch on.

DS

Distortion sound.

FUZZ

Fuzz sound.

Gain [Gain] (Pre Amp Gain)LOW, MID, HIGH

Switch the degree of pre-amp distortion between three levels (Low/Middle/High).

Volume [Vol] (Pre Amp. Volume)0 - 100

Adjusts the volume and degree of distortion of the amp.

MasterVol [Mst] (Pre Amp. Master Volume) 0 - 100

Adjust the volume of the entire pre-amp.

Bright [Br.Sw] (Brightness)OFF, ON

Turning this "On" will produce a sharper and brighter sound. This parameter can be set if the Type is set to "JC-120," "Clean Twin," or "BG Lead."

Bass [Bass]0 - 100

Adjust the tone of the low range.

Middle [Mid]0 - 100

Adjust the tone of the mid range. If "Match Drive" is selected for the Type parameter, this parameter cannot be set.

Treble [Tre]0 - 100

Adjust the tone of the high range.

Presence [Pres]0 - 100 (-100 - 0)

Adjust the tone of the ultra-high range. Normally the range will be 0–100, but when "Match Drive" is selected, the range will be -100–0.

SPEAKER SIMULATOR

This effect simulates a speaker system.

Type [Type] (Speaker type)
SMALL, MIDDLE, JC-120, Bltln 1, Bltln 2, Bltln 3, Bltln 4, BG STK1, BG STK2, MS STK1, MS STK2, MetlSTK

Select the type of speaker. The specifications of each type are as follows. The speaker column indicates the diameter of each speaker unit (in inches) and the number of units.

Type	Cabinet	Speaker	Mic
SMALL	a	10	D
MIDDLE	b	12 x 1	D
JC-120	b	12 x 2	D
BltIn 1	b	12 x 2	D
BltIn 2	b	12 x 2	C
BltIn 3	b	12 x 2	C
BltIn 4	b	12 x 2	C
BG STK1	c	12 x 2	C
BG STK2	d	12 x 2	C
MS STK1	d	12 x 4	C
MS STK2	d	12 x 4	C
MetlSTK	e	12 x 4	C

a: Small open-back enclosure
b: open back enclosure
c: sealed enclosure
d: large sealed enclosure
e: large double stack
C: condenser mic
D: dynamic mic

MicSetting [MicS]1,2,3

Specify the location of the mic that is recording the sound of the speaker. This can be adjusted in three steps, with the mic becoming more distant in the order of 1, 2, and 3.

Level [MicLv] (Mic Level)0 - 100

Adjust the volume of the mic sound.

Dry Level [DryLv]0 - 100

Adjust the volume of the direct sound.

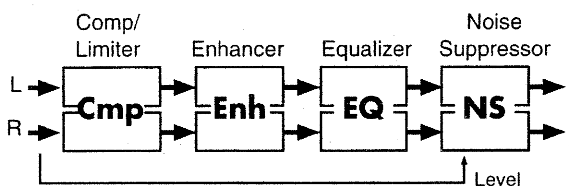
Recommended combinations of pre-amp and speaker

Pre-amp type	Speaker type
BG LEAD	BG STK1, BG STK2, MIDDLE
1959-2	BG STK1, BG STK2, MetlSTK
1959-12	BG STK1, BG STK2, MetlSTK
SLDN	BG STK1, BG STK2, MetlSTK
5150	BG STK1, BG STK2, MetlSTK
MetalLD	BG STK1, BG STK2, MetlSTK

OD-2T	BltIn 1–4
DS	BltIn 1–4
FUZZ	BltIn 1–4

14 STEREO DYNAMICS PROCESSOR

This features a comp/limiter, enhancer, 3-band equalizer, and noise suppressor, all connected in series. This is convenient as an overall effect applied during mixdown, or as a way to fix input sounds when sampling. Use the insert method with this algorithm.



COMPRESSOR/LIMITER

This effect includes a compressor, which controls inconsistencies in sound levels by suppressing high sound levels while lifting weaker signals, in addition to a limiter that prevents the signal from reaching exceedingly high levels.

Threshold [Thre] (Threshold Level)-60 – 0 dB
This sets the volume level at which the compression begins.

Attack [Atk] (Attack Time)0 – 100
This sets the time after the threshold level is crossed for compression to begin.

Release [Rele] (Release Time)0 – 100
This sets the time for compression to stop after the sound falls back under the threshold level.

Ratio [Ratio] (Compression Ratio)
1.5:1, 2:1, 4:1, 100:1
This sets the compression ratio of the source sound to the output sound.

Out Level [OutLv]-60 – +12 dB
This sets the output volume.

NOTE

When used as a limiter, set the Ratio to 100:1 with a short attack and release times. If the volume exceeds the threshold, the sound is suppressed the instant the excess input is detected.

ENHANCER

These parameters are the same as those in the EQ in Algorithm 01 (REVERB & GATE) (p.10).

Sens [Sens] (Sensitivity)0 – 100
Frequency [Freq]1.0 – 10.0 kHz
Mix Level [MixLv]0 – 100
Out Level [OutLv]0 – 100

EQ (3-Band Equalizer)

These parameters are the same as those in the EQ in Algorithm 03 (REVERB & GATE) (p.10).

Low Type [LType]SHELV, PEAK
Low Freq [LFreq]20 – 2000 Hz
Low Gain [LGain]-12 – +12 dB
Low Q [LQ] 0.3 – 10.0
Mid Freq [MFreq]200 – 8000 Hz
Mid Gain [MGain]-12 – +12 dB
Mid Q [MQ] 0.3 – 10.0
High Type [HType]SHELV, PEAK
High Freq [HFreq]1.4 – 20.0 kHz
High Gain [HGain]-12 – +12 dB
High Q [HQ] 0.3 – 10.0
Out Level [OutLv]-60 – +12 dB

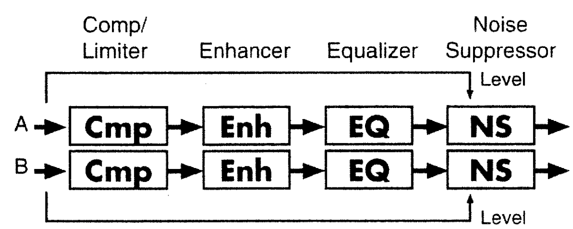
NOISE SUPPRESSOR

These parameters function the same way as those in the NOISE SUPPRESSOR in Algorithm 03 (VOCAL MULTI) (p.13).

Threshold [Thre] (Threshold Level)0 – 100
Release [Rele] (Release Time)0 – 100

15 DYNAMICS x 2

This features a comp/limiter, enhancer, 3-band equalizer, and noise suppressor, all connected in series.



DYNAMICS PROCESSOR A/B

These parameters are the same as those in Algorithm 14 (STEREO DYNAMICS PROCCESSOR) (p.25).

Threshold [Thre] (Threshold Level)-60 – 0dB

Attack [Atk] (Attack Time)0 – 100

Release [Rele] (Release Time)0 – 100

Ratio [Ratio] (Compression Ratio)1.5:1, 2:1, 4:1, 100:1

Out Level [OutLv]-60 – +12dB

Sens [Sens] (Sensitivity)0 – 100

Frequency [Freq]1.0 – 10.0 kHz

Mix Level [MixLv]0 – 100

Out Level [OutLv]0 – 100

Low Type [LType]SHELV, PEAK

Low Freq [LFreq]20 – 2000 Hz

Low Gain [LGain]-12 – +12 dB

Low Q [LQ] 0.3 – 10.0

Mid Freq [MFreq]200 – 8000 Hz

Mid Gain [MGain]-12 – +12 dB

Mid Q [MQ] 0.3 – 10.0

High Type [HType]SHELV, PEAK

High Freq [HFreq]1.4 – 20.0 kHz

High Gain [HGain]-12 – +12 dB

High Q [HQ] 0.3 – 10.0

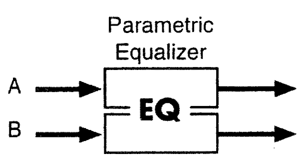
Out Level [OutLv]-60 – +12 dB

Threshold [Thre] (Threshold Level)0 – 100

Release [Rele] (Release Time)0 – 100

16 PARAMETRIC EQ

This is four-band parametric equalizer can be used as either two monaural equalizers or a single stereo equalizer.



LINK

These parameters are the same as those in the LINK in Algorithm 12 (Mic Simulator) (p.22).

Link Sw [Link]OFF, ON

PARAMETRIC EQ A/B

This is a four-band parametric equalizer.

In Level [In Lv] (Input Level)-60 – +12 dB

This adjusts the level before the signal passes through the equalizer.

Low Type [LType]SHELV, PEAK

This switches the Low EQ curve characteristics (peaking-type/shelving-type).

Low Freq [LFreq] (Low Frequency) 20 - 2000 Hz

This adjusts the Low EQ center frequency.

Low Gain [LGain]-12 - 12 dB

This sets the gain (boost or cut) of the equalizer's low range.

Low Q [LQ]0.3 - 10.0

This sets the "Q", or frequency range over which the sounds passing through the Low EQ are boost or cut. The higher the value set, the narrower the range.

LoMid Freq [LMFrq] (Low-Middle Frequency) 200 - 8000 Hz

This adjusts the Low-Mid EQ center frequency.

LoMid Gain [LM-G] (Low-Middle Gain) -12 - 12 dB

This sets the gain (boost or cut) of the equalizer's Low-Mid range.

LoMid Q [LMQ] (Low Middle Q)0.3 - 10.0

This sets the "Q", or frequency range over which the sounds passing through the Low-Mid EQ are boost or cut. The higher the value set, the narrower the range.

HiMid Freq [HMFrq] (High-Middle Frequency) 200 - 8000 Hz

This adjusts the High-Mid EQ center frequency.

HiMid Gain [HM-G] (High-Middle Gain) -12 - +12 dB

This sets the gain (boost or cut) of the equalizer's High-Mid range.

HiMid Q [HMQ] (Hi-Middle Q)0.3 - 10.0

This sets the "Q", or frequency range over which the sounds passing through the High-Mid EQ are boost or cut. The higher the value set, the narrower the range.

High Type [HType]SHELV, PEAK

This switches the High EQ curve characteristics (peaking-type/shelving-type).

High Freq [HFreq] (High Frequency) 1.4 - 20.0 kHz

This adjusts the High EQ center frequency.

High Gain [HGain]-12 - +12 dB

This sets the gain (boost or cut) of the equalizer's high range.

High Q [HQ]0.3 - 10.0

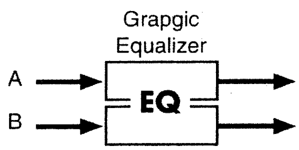
This sets the "Q", or frequency range over which the sounds passing through the High EQ are boost or cut. The higher the value set, the narrower the range.

Out Level [OutLv]-60 – +12 dB

This sets the volume after the signal passes through the equalizer.

17 GRAPHIC EQ

This simulates a 10-band graphic equalizer. It can be used as either two monaural equalizers or a single stereo equalizer.



LINK

These parameters are the same as those in Lnk in Algorithm 12 (MIC SIMULATOR) (p.22).

Link Sw [Link]OFF, ON

GRAPHIC EQ A/B

This simulates a 10-band graphic equalizer.

In Level [In Lv]-60 – +12 dB

This adjusts the level before the signal passes through the equalizer.

31.2 Hz [31Hz] (Gain)-12 – +12 dB

62.5 Hz [62Hz] (Gain)-12 – +12 dB

125 Hz [125Hz] (Gain)-12 – +12 dB

250 Hz [250Hz] (Gain)-12 – +12 dB

500 Hz [500Hz] (Gain)-12 – +12 dB

1kHz [1kHz] (Gain)-12 – +12 dB

2kHz [2kHz] (Gain)-12 – +12 dB

4kHz [4kHz] (Gain)-12 – +12 dB

8kHz [8kHz] (Gain)-12 – +12 dB

16kHz [16kHz] (Gain)-12 – +12 dB

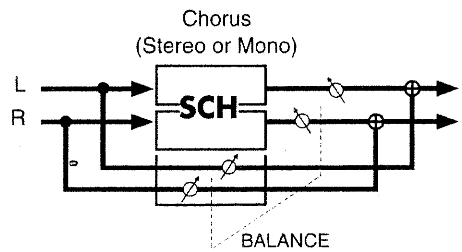
This sets the gain (boost or cut) for each of the equalizer's frequencies.

Out Level [OutLv]-60 – +12 dB

This sets the volume after the signal passes through the equalizer.

18 SPACE CHORUS

This algorithm is a reproduction of Roland's SDD-320 spatial expression effect. This effect add breadth to the stereo output. Although the insert method was used to add the SDD-320's effect, you can set this algorithm so that it can be used with the send/return method as well.



SPACE CHORUS

This is an effect that gives the sound greater fullness and breadth.

Input [Input] (Input Type)MONO, ST

This setting determines whether the stereo sound input is converted to monaural output (MONO) or not (ST) (this was accomplished on the SDD-320 by connecting to different jacks).

ModeButton [Mode] (Chorus Mode Button) 1, 2, 3, 4, 1+4, 2+4, 3+4

The SDD-320 featured four mode buttons which were pressed to change the way the effect worked. This setting simulates the buttons that were pressed (setting this to "1+4" simulates the effect achieved by pressing the 1 and 4 buttons simultaneously).

Dry/FX Bal [Bal] (Dry/FX Balance)0 – 100

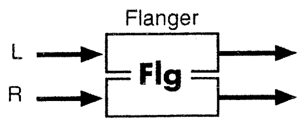
This adjusts the volume balance of the source sound and the effect. With a setting of 50, the volume of the source and the SDD-320 are equally balanced. Set to 0 for source only; at 100, only the sound from the SDD-320 is output. Set this to 100 when using the send/return method.

NOTE

Setting different mode buttons subtly changes the effect. Try each mode and select the one that best suits your aims. Roland's SDD-320, released in 1979 and in production for about eight years, is an analog effect that adds spatiousness to the sound. The panel features just five buttons (OFF and four MODE buttons); the different buttons are pressed to select the effect. Although a type of chorus effect, it features a natural sound, without a lot of oscillation. Even today, this model is a big favorite of many remix artists and musicians.

19 STEREO FLANGER

This algorithm features a pair of the same flanger circuits as that in the BOSS line compact flangers connected in parallel for stereo input. This algorithm is added using the insert method.



FLANGER

This adds a particular metallic-sounding modulation (rise and fall in pitch) to the source sound.

Model Type [Model]NORM, Hi-B

This selects the model of simulated flanger.

NORM

(Normal type <BOSS BF-2>)

HI-B

(High-Band type <BOSS HF-2>) This setting raises the flanger sound one octave above that at the NORM setting.

Manual [Manu]0 - 100

This sets the center frequency for the effect. This changes the pitch of the flanger's metallic sound.

Depth [Depth]0 - 100

This sets the depth of the flanger's modulating sound.

Rate [Rate]0 - 100

This sets the rate of the swelling of the flanger sound.

Resonance [Reso]0 - 100

This sets the intensity of the flanger's effect. Take care to prevent this sound from damaging your ears or your playback equipment.

LFO Phase [Phase]0 - 180 deg

This adjusts left and right phase shift in the oscillator that produces the wavering effect. This changes the timing of the rise and fall of the modulation in the left and right channels. At 0deg (0 degrees), the left and right pitches rise and fall together. At 180 degrees, they are completely opposite.

Cross FB [X-FB] (Cross Feedback)-100 - +100

This setting takes the input sounds from the right and left channels and returns them through the opposite channel's flanger, resulting in an even stronger flanging effect. A positive setting causes the inputs to be returned in phase; a negative setting produces outputs with inverted phase.
Setting the Cross Feedback value too high may result in extreme oscillation. Take care to prevent this sound from damaging your ears or your playback equipment.

Cross Mix [X-Mix]-100 - +100

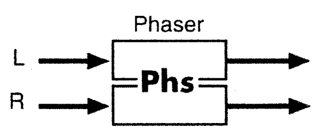
This setting takes the flanging sound from each of the right and left channels and mixes it with the flanging sound of the opposite channel.
A positive setting causes the inputs to be returned in phase; a negative setting produces outputs with inverted phase.

NOTE

Cross Feedback and Cross Mix are effects that you cannot get even by actually connecting two flangers in parallel. These parameters have been added in this algorithm are intended for use in stereo. As a "hidden" technique, set a negative value for the Cross Mix effect (invert the phases) to get a stereo flanging effect that features a particular floating sensation.

20 VINTAGE PHASE

This algorithm features two analog-type phasers arranged in parallel, making the effect good for use in stereo. The sound is added to the source sound as it cycles in and out of phase, creating the distinctive phaser modulation. This algorithm is used with insert method. With the send/return method, the effect is further mixed in with the source sound, which may weaken the effect.

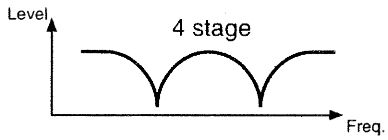


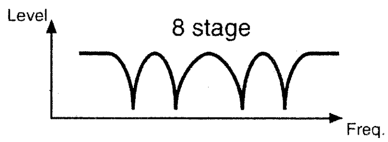
PHASER

This effect features two linked monaural phasers arranged in parallel.

ShiftMode [Shift] (Phase Shift Mode) 4STAGE, 8STAGE

This sets the number of stages in the phase shift circuit (four or eight). Setting this to eight stages increases the number of points with cancelled frequencies, which sharpens the effect.





CenterFreq [Freq] (Center frequency)0 - 100
This sets the center frequency of the phaser effect applied. The phaser effect frequency range rises as the value is increased.

Resonance [Reso]0 - 100
The more this value is increased, the stronger this distinctive effect becomes. However, setting the Resonance value too high can result in extreme oscillation. Take care to prevent this sound from damaging your ears or your playback equipment.

LFO1 Depth [L1Dep]0 - 100
LFO2 Depth [L2Dep]0 - 100
These set the depth of the swelling sound.

LFO1 Rate [L1Rat]0 - 100
LFO2 Rate [L2Rat]0 - 100
These set the modulation rate.

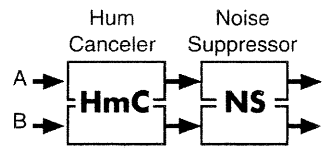
LFO1 Phase [L1Phs]NORM, INV
LFO2 Phase [L2Phs]NORM, INV
These set the phase of the modulation left and right. When set to Normal (NORM), the phase is unchanged; when inverted (INV), the phase is inverted.

NOTE

This algorithm reproduces the sound of the 2U rack-mounted phasers of the early 1980s. Two monaural single-input, single-output phasers are arranged in parallel, allowing the creation of complex modulation patterns. The LFO1 and LFO2 have different modulation rates. LFO1 creates an extremely slow modulation; that of LFO2 is faster. You can set the phase of each one independently. With a large swelling sound from LFO1 and a very short, phase inverted wavering from in LFO2, you can create a sound producing a sensation of great breadth.

21 HUM CANCELER

This removes unpleasant hum from the sound. Noise suppression is added to the output.



HUM CANCELER

Frequency [Freq]20.0 - 800.0 Hz
This sets the frequency at which the hum is removed. Hum is removed at the selected frequency as well as multiples of that frequency. Set this to the frequency of your power source.

Width [Width] (Band Width)10 - 40 %
This sets the bandwidth of the filter removing the hum.

Depth [Depth]0 - 100
This sets the depth of the filter removing the hum.

Threshold [Thre] (Threshold Level)0 - 100
This sets the level at which the Hum Canceller becomes effective. When the signal falls below the specified level, only the hum is removed the signal. At the maximum value, the hum is removed at all times, regardless of the signal level.

Lo-F Limit [LoLim] (Low Frequency Limit)
THRU, 20 - 2000 Hz
This sets the minimum frequency for the Hum Canceller function. When "THRU" is selected, all frequencies that can be played back through the A-6 are processed with the Hum Canceller.

Hi-F Limit [HiLim] (High Frequency Limit)
1.0 - 20.0 kHz, THRU
This sets the maximum frequency for the Hum Canceller function. When "THRU" is selected, all frequencies that can be played back through the A-6 are processed with the Hum Canceller.

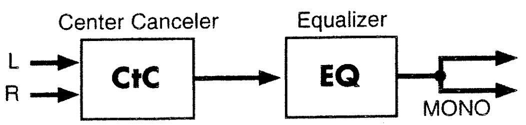
NOISE SUPPRESSOR

These parameters function the same way as those in the NOISE SUPPRESSOR in Algorithm 03 (VOCAL MULTI) (p.13).

Threshold [Thre] (Threshold Level)0 – 100
Release [Rele] (Release Time)0 – 100

22 CENTER CANCELER

Center Canceller is an effect that cuts out the sounds positioned in the center of the stereo field. In addition to this, it features a 3-band parametric equalizer connected in series. Use the insert method with this algorithm.



CENTER CANCELER

This cuts sounds (such as vocals) placed in the center of the stereo field.

Position [Pos] (Cancel Position)-50 – +50

This is for finer adjustment of the cut position. This can be adjusted so that the sound is cut to a great extent.

Lo-F Limit [LoLim] (Low Frequency Limit) THRU, 20 – 2000 Hz

Hi-F Limit [HiLim] (High Frequency Limit) 1.0 – 20.0 kHz, THRU

These set the upper (Hi-F) and lower (Lo-F) limits of the frequency range to be cut. When "THRU" is selected, there is no limit on the frequencies to be cut.

NOTE

With this effect, output is converted to mono. Although you can get a similar effect by using the Anti-Phase function in Algorithm 23 (Isolator + Filter), this algorithm differs in that you can specify the upper and lower frequency limits of the effect. This is especially effective when cutting vocals, for example.

NOTE

This has no effect when the input sound is monaural. Additionally, even with stereo input, the amount of cut may differ depending on the particular recording.

EQ (3-Band Equalizer)

These parameters are the same as those in the ENHANCER in Algorithm 03 (VOCAL MULTI) (p.13).

Low Type [LType]SHELV, PEAK

Low Freq [LFreq]20 – 2000 Hz

Low Gain [LGain]-12 – +12 dB

Low Q [LQ] 0.3 – 10.0

Mid Freq [MFreq]200 – 8000 Hz

Mid Gain [MGain]-12 – +12 dB

Mid Q [MQ] 0.3 – 10.0

High Type [HType]SHELV, PEAK

High Freq [HFreq]1.4 – 20.0 kHz

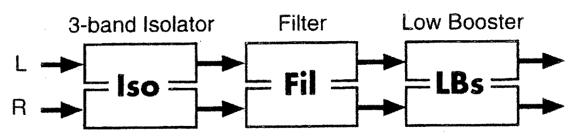
High Gain [HGain]-12 – +12 dB

High Q [HQ] 0.3 – 10.0

Out Level [OutLv]-60 – +12 dB

23 ISOLATOR & FILTER

This features a 3-band isolator, filter, and low booster that are connected in series in stereo.



ISOLATOR(3-Band Isolator)

This effect separates the input sound into three frequency ranges— High, Mid, and Low—and boosts or cuts each range.

Level High [Hi Lv]-60 – +4 dB

Level Mid [MidLv] -60 – +4dB

Level Low [LowLv]-60 – +4 dB

Each frequency range—High, Mid, or Low—can be boost or cut. At -60 dB, the sound becomes inaudible. A level of 0 dB is equivalent to the input level of the sound.

AntiPhs Md [PhsMd] (Anti-Phase Middle Switch) OFF, ON

AntiPhs Md Level [PhsML] (Anti-Phase Middel Level)0 – 100

AntiPhs Lo [PhsLo] (Anti-Phase Low Switch) OFF, ON

AntiPhs Lo Level [PhsLL] (Anti-Phase Low Level) 0 – 100

This turns the Anti-Phase function on and off and sets level settings for the low and mid frequency ranges. When turned on, the phase from the opposite stereo channel is inverted and added to the signal. Depending on the level settings, you can achieve an effect that sounds as if only a particular part is being boosted. (This is effective only in stereo.)

NOTE

Functions featured in machines consider standard equipment for remix artists and pro DJs have been carefully analyzed and reproduced. Ordinary equalizers allow some sound to persist even when the gain is turned down all the way. In contrast, the Isolator completely cuts off the sound. By switching the effect on and off, and by changing levels in real time, you can have the sound of *specific parts appear and disappear.*

FILTER

These parameters are the same as those in the ENHANCER in Algorithm 02 (EZ DELAY) (p.12).

Type [Type] (Filter Type)LPF, BPF, HPF, NOTCH

Slope(oct) [Slope]-12, -24 dB

CutOffFreq [Freq] (Cutoff Frequency)0 – 100

Resonance [Reso]0 – 100

Gain [Gain] 0 – 24 dB

LOW BOOST

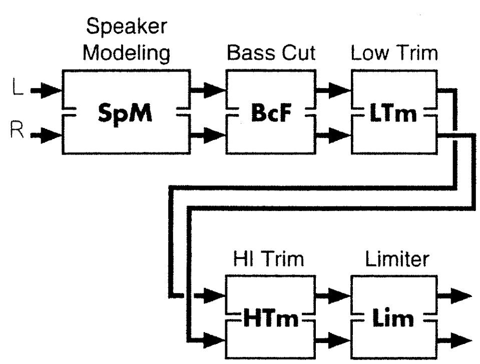
This adds emphasis to the low end to create a fuller bass sound.

BoostLevel [Boost] 0 - 100

Increasing this value gives you a heavier low end (this effect may be hard to distinguish with certain Isolator and filter settings).

24 SPEAKER MODELING

You can model the acoustical characteristics of a variety of speakers, ranging from high-level professional monitor speakers used in studios worldwide, to the speakers of small televisions or portable radios.
Speaker modeling has been calibrated so that the optimal effect will be obtained when Roland DS-90 powered monitors are connected digitally. If you are using other speakers, you may not be able to obtain the desired effect.



SPEAKER MODELING

Model [Model] (Modeling Speaker)
THRU, FLAT, Pwd.BLK, Pwd.E-B, Pwd.MAC, SmlCUBE, Wh.CONE, WhTISUE, RADIO, SmallTV, BoomBox, BoomLoB, Pwd.SR, StackSR

Select the speaker whose characteristics will be simulated (modeled).

THRU

Modeling will not be applied, and the sound will be output without change. Use this as a comparison with the speaker-modeled sounds.

FLAT

Modeling is used to compensate the DS-90, to produce an even flatter sound with a wider range.

Pwd.BLK

A widely used model of powered monitors (two-way type, with a woofer diameter of 170 mm (6-1/2 inches)).

Pwd.E-B

Powered monitors characterized by a bright tone.

Pwd.MAC

Powered monitors characterized by an extended low-frequency response.

SmlCUBE

Small full-range speakers widely used in recording studios.

Wh.CONE

Sealed enclosure two-way speakers known for their white woofers and widely used in recording studios.

WhTISUE

A more mild sound, with tissue paper affixed over the tweeters of the above "White Cone" speakers.

RADIO

Small pocket-type radio.

SmallTV

Speakers built into a 14 inch size television.

BoomBox

Radio cassette recorder.

BoomLoB

Radio cassette recorder with the Low Boost switched on.

Pwd.SR

American powerd speaker widely used in concert, etc..

StackSR

A sound of the above "Pwd.SR" speakers with sub woofer.

OutSP [OutSP] (Output Speaker) DS-90, MS-50, SST-151, SST-251, 151+351, 251+351

Select the type of speakers that was actually used. When either "151+351" or "251+351" is selected for "OUT SP," use parallel connections for chaining two speakers.

DS-90

Roland Powered Monitor DS-90

MS-50

Roland Monitor Speaker MS-50

SST-151

Roland Speaker System SST-151

SST-251

Roland Speaker System SST-251

151+351

Roland Sub Woofer SSW-351 with SST-151

251+351

Roland Sub Woofer SSW-351 with SST-251

Phase [Phase]NORM, INV

Specifies the phase of the speakers.

NRM: Same phase as the input.

INV: Opposite phase of the input.

BASS CUT FILTER

This removes unwanted low-frequency components, such as pop noise.

Frequency [Freq]THRU, 20 - 2000 Hz

Adjusts the cutoff frequency of the bass-cut filter.

LOW FREQUENCY TRIM

Gain [Gain]-12 - 12 dB

For low frequency trimmer, adjusts the gain (amount of boost/cut).

Frequency [Freq]20 - 2000 Hz

Set the center frequency of low frequency trimmer.

HIGH FREQUENCY TRIM

Gain [Gain]-12 - 12 dB

For high frequency trimmer, adjusts the gain (amount of boost/cut).

Frequency [Freq]1.0 - 20.0 kHz

Set the center frequency of high frequency trimmer.

LIMITER

This effect limits excessive output levels.

Threshold [Thr]-60 - 0 dB

Specifies the level at which the limiter will begin to operate.

Release [Rele]0 - 100

Adjusts the time from when the input level falls below the threshold level until the limiter ceases to operate.

Out Level [OutLv]-60 - 24 dB

Adjusts the volume level of the sound that has passed through the limiter.

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